

audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Market Update

Installation, Distributed Audio, and Multiroom

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Practical Test & Measurement

The Proof Is in the Testing

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and Christian Bellmann (Klippel)

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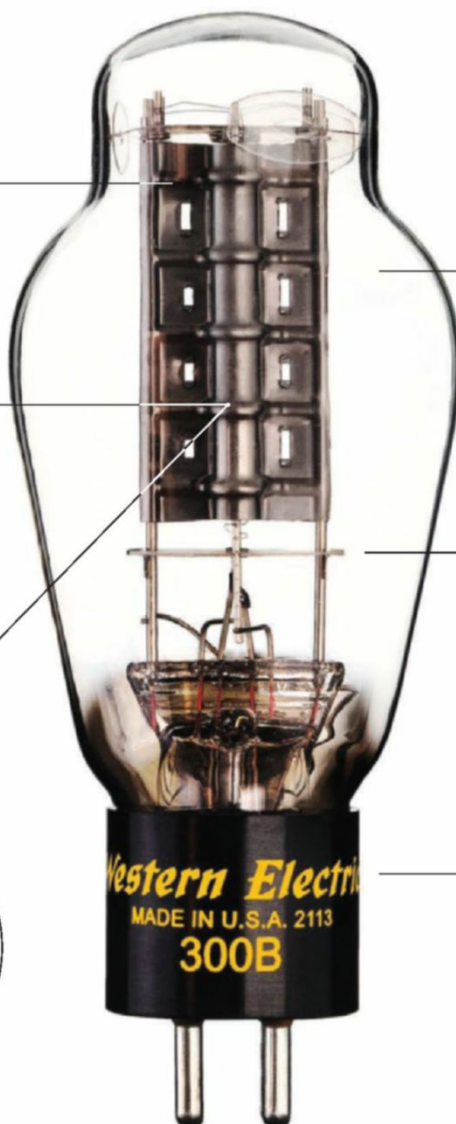
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December 2022

ISSN 1548-0628

www.audioXpress.com

audioXpress (US ISSN 1548-0628) is published monthly, at \$50 per year for the US, at \$65 per year for Canada, and at \$75 per year Foreign/ROW, by KCK Media Corp., P.O. Box 417, Chase City, VA 23924, US Periodical Postage paid at Chase City, VA, and additional offices.

Head Office:

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audioXpress

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Chase City, VA 23924, US

Phone: 434-533-0246

E-mail: custserv@audioXpress.com

Internet: www.audioXpress.com

Postmaster: Send address changes to:

audioXpress

P.O. Box 417

Chase City, VA 23924, US

Advertising:

Strategic Media Marketing, Inc.

2 Main Street

Gloucester, MA 01930, US

Phone: 978-281-7708

Fax: 978-281-7706

E-mail: audioXpress@smmarketing.us

Advertising rates and terms available on request.

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Printed in the US



Wireless and Wired

Although unintended, this issue of *audioXpress* offers a very interesting comparison between the thriving installation market and the complexity of consumer mainstream technologies, while also revealing the convergence trends. From wired to wireless, the two Market Updates featured in this issue detail technology evolving and adjusting to different requirements at its own pace.

In the process that will eventually lead us to a world of Wireless Audio, many efforts and even companies that pioneered the early streaming audio and multiroom systems have been left behind. And sometimes evolution serves up strange twists. Due to the mighty power of the mobile industry and the spectacular rise of smartphones, Bluetooth technology ended up being perceived by consumers almost as synonymous with wireless audio—even if wasn't designed for that and barely handles it. The result today is that any superior audio system working over Wi-Fi or even using a dedicated/proprietary transmission system, eventually needs to be able to receive a signal source streamed over Bluetooth due to that technology's ubiquity for mobile devices and low-power wearables.

While the early goal of many wireless audio systems was to allow "Wired-Quality Sound Without the Wires," the reality of the Wireless Networking technologies that might eventually become standard will be far more complex and will enable even more possibilities than we could ever have imagined.

But there's also still the need for high-quality, high-resolution Wireless Audio Transmission that "simply works." And that's the reason why we see a growing number of radio frequency transmission efforts using dedicated hardware and proprietary digital encoding and protocols, mostly just for simple point to point or to multipoint applications. We can see the signs everywhere, with manufacturers shipping dedicated transmitters to plug directly into analog jacks or USB ports, simply to enable a reliable and "better than Bluetooth" wireless audio transmission. The gaming industry is also another example of discontent, with wireless systems that require "dongles" and dedicated drivers and software to enable lower latency.

It is clear that we still need evolution in wireless audio to address simple Wireless Audio Transmission and Wireless Multichannel applications. At the consumer level, wireless audio is the only way to enable immersive audio in a practical and economic way for mainstream applications.

Meanwhile, the Connectivity Standards Alliance, supported by more than 550 technology companies, announced the release of the Matter 1.0 specification, which will finally pave the way for the next generation of Smart Home connected products that work across brands and platforms. This is certain to have a significant impact over all sectors of the audio industry, including residential integration.

However, consumers continue to buy connected products. In its latest consumer survey of 10,000 US households, research firm Parks Associates reveals that 38% own at least one smart home device. Twenty-seven percent of US Internet households report buying a smart home device in the past 12 months, and 44% of households report intentions to purchase a smart home device in the next 12 months.

For perspective, Nielsen estimates that TVs are in 96.2% of US homes (2020-2021), and 52% of all TV sets are now reported to be smart TVs, and are one of main content hubs in the smart home. This means that more than half of those homes are able to stream content directly from the TVs to speakers, soundbars, and other systems in the same or other rooms in the house. But it doesn't mean that they do. And the reason is mainstream consumers are not going to run wires to anything that's not under or very close to the TV. And wireless audio, particularly wireless home theater, is not yet well understood.

Connected homes, smart TVs, computers, tablets, and smartphones are massively used as access points for streaming content services, represent a massive opportunity for the audio industry.

Impactful audio experiences are only possible from high-quality speakers. As long as those are connected, they remain the preferred option for quality reproduction of music, movies, and gaming. The missing link, literally, is the convenience of wireless audio transmission for all those applications.

J. Martins
Editor-in-Chief

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By Adrián Lara Moreno and Christian Bellmann

The Klippel Near Field Scanner (NFS) System was used for the development of Holoplot's X1 Matrix Array compensation algorithms, which was the topic of a September 2022 *audioXpress* article. This joint article by Holoplot and Klippel offers a deep dive into the tuning and performance adjustments of the innovative technology from the Berlin, Germany-based company.

44 An Approach to Designing RIAA Networks

By John Ellis

This technical article addresses an issue that often keeps designers busy when calculating the RIAA parts' values for a phono amp. The reason is that the surrounding circuit parts always somehow influence the RIAA correction. So it is often a matter of trial and error. With the author's method, it is easier to quickly calculate the correct values.

50 The LNA-100

A Low Noise Amplifier with Filters and a RMS Voltmeter

By George Ntanavaras

This project describes the design and construction of the LNA-100 device, which comprises a low noise, high gain amplifier with three different gain settings, six filters for bandwidth restriction, and a True RMS voltmeter that gives a DC voltage at its output.

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By Chuck Hansen

In the second part of this article series, Chuck Hansen details the history of transformer cores and shares information about their various construction methods.

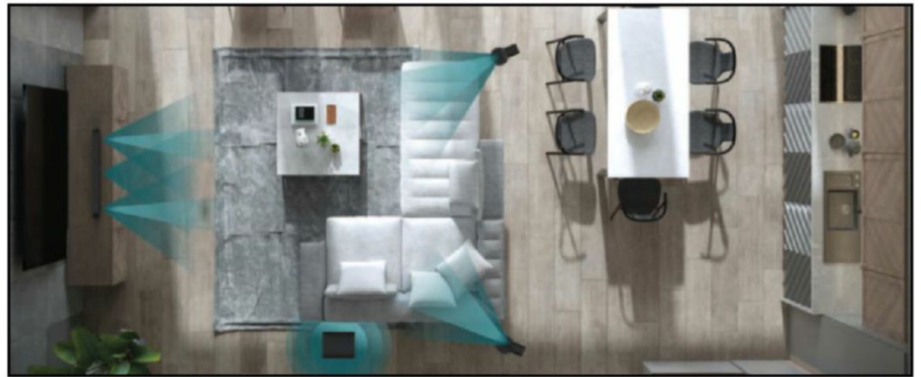
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HOLLOW-STATE ELECTRONICS

64 Unusual Tubes

By Richard Honeycutt

Most aficionados of hollow-state electronics are familiar with the tubes commonly used in classic or contemporary audio amplifiers. In this article, we explore some of the less common tube types used for audio applications.



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MARKET UPDATE

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By J. Martins

Following the previous Market Update on residential integration, *audioXpress* explores the latest audio installation trends and solutions. Complementing the residential perspective, the article also addresses the latest audio multiroom approaches and technologies, an area that is clearly on the verge of serious evolution in 2023.

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MARKET UPDATE

30 Improving Wireless Audio

Part 2 — Evolution and Exciting Perspectives

By J. Martins

Even though key technologies remain in development, wireless audio evolution beyond Bluetooth offers some very promising perspectives to the industry. This update looks at the efforts and the most promising platforms and technologies for wireless audio.

40 Using Ultra-Wideband Technology

Wireless Delivery of Audio and Multimedia

By Frederic Nabki, Nicolas Paillusseau, and Ronald Rinkenbach

For tomorrow's audio and AR/VR devices, Ultra-Wideband (UWB) technology provides high throughput, low latency, and low power connectivity for body and personal area networks—even in congested environments. UWB complements Bluetooth and Wi-Fi with high-performance and mobile-friendly short-range wireless connectivity.

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Installation, Distributed Audio, and Multiroom

By
J. Martins
(Editor-in-Chief)



Following the previous Market Update on residential integration, *audioXpress* explores the latest audio installation trends and solutions. Complementing the residential perspective, the article also addresses the latest audio multiroom approaches and technologies, an area that is clearly on the verge of serious evolution in 2023.

The last time we had a serious look at the state of installation audio was October 2019. Commercial and residential audio was thriving and trade shows were growing exponentially as a new breed of IT-oriented integration companies were trying to combine audio products and expertise into their systems approach. Not surprisingly, just before what was to be the last large trade show before the pandemic (Integrated Systems Europe in February 2020, which was also the last edition in Amsterdam) we titled our Market Update “Integrated, Networked, and Secure.” That clearly reflected the priorities for technology evolution and systems integration.

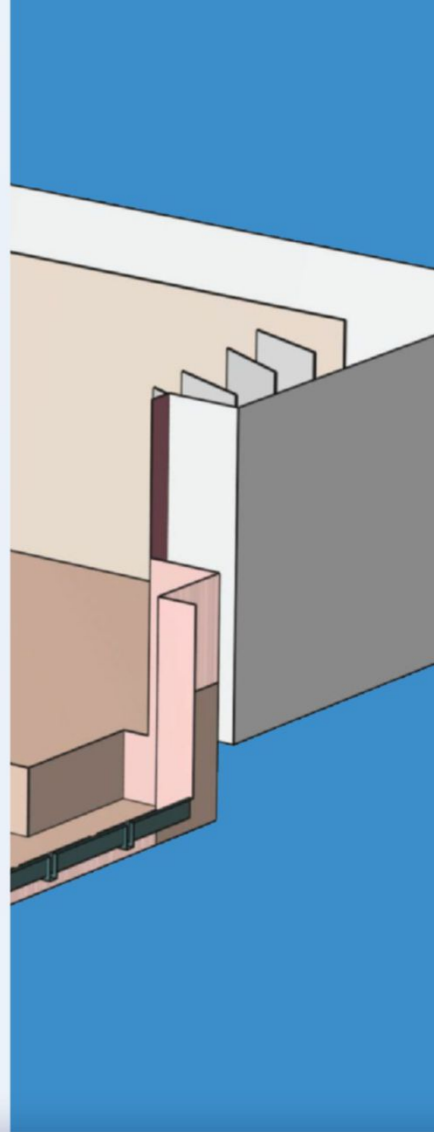
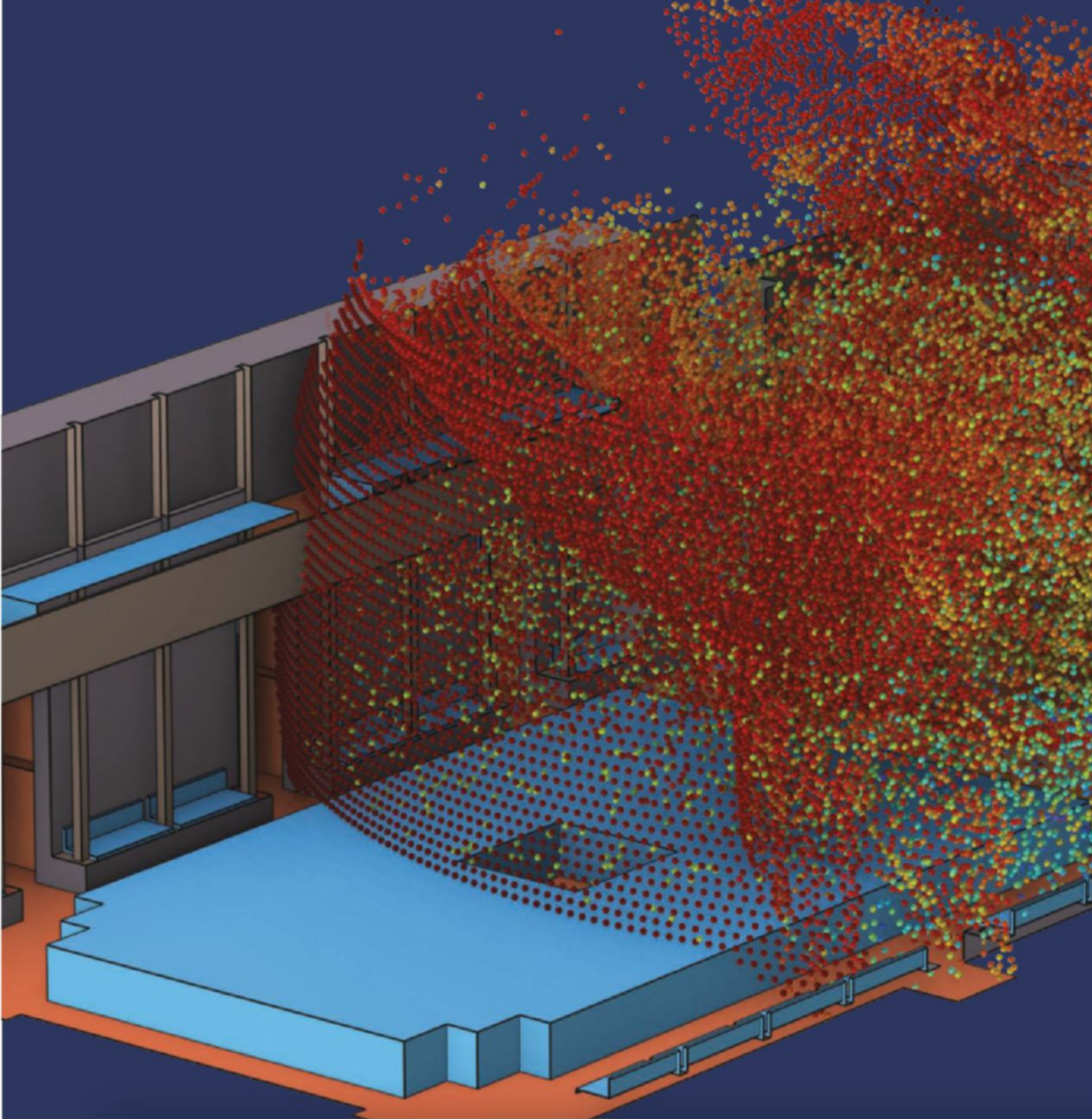
As everyone in the industry adapted to the “bumping fists” and occasional “elbow rubbing” alternatives imposed by “no handshake” policies of a global pandemic, things have not evolved as one would expect in this field. Not that the pandemic affected systems integration as badly as it did live performance and entertainment. But what it did cause was a complete lack of industry forums to confirm trends and innovations, although companies in all market segments continued—and some even accelerated—their R&D efforts. Many companies even used the opportunity to completely rethink their market strategies, reflecting the opportunities that were opening up.

A good example, although still with unforeseen consequences, was the surprise split of QSC Audio into two separate divisions: Q-SYS and QSC Pro Audio. Anyone who attended InfoComm 2022

could notice that something was changing when, in place of QSC, the booth was occupied by a company called Q-SYS. Small signs in that booth still showed the QSC logo and the slogan “This Is Q-SYS”. Soon after, QSC announced the new QSC Pro Audio division with the appointment of a new leadership team of “veteran staff” (recruited from JBL, where else?).

QSC/Q-SYS remains a single company for now, and still shares the same board, with Joe Pham as Chairman and CEO, and Jatan Shah as President and Chief Operating Officer. QSC (formerly QSC Audio) and Q-SYS still share the same headquarters location in Costa Mesa, CA. It’s just that the company is now split into two separate divisions, each one with its own brand and a different mission. The main motivation for this strategic change is that QSC is no longer just “an audio company” and that it seeks growth and opportunity in the broader systems integration space.

For nearly 54 years, QSC was a familiar brand for musicians, mobile entertainers, and sound reinforcement professionals, first with a range of pro audio amplifiers and next with a very successful range of speakers and cinema systems. Meanwhile, the Q-SYS range of networked products was born from a foundation on QSC’s own Q-LAN technology developed since 2006 by the Boulder, CO, team lead by Rich Zwiebel and John Britton, who had left Cirrus Logic and accepted management positions at QSC Audio—both of which remained working at QSC.

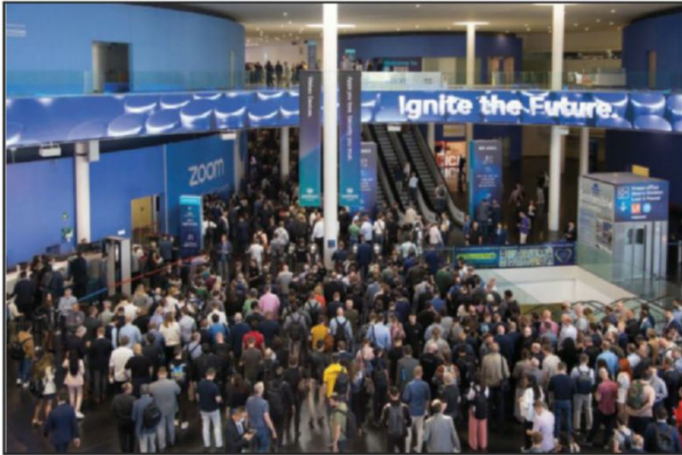


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A return to trade shows in 2022. Integrated Systems Europe (ISE) opening for the first time at the Barcelona Fira exhibition center, with 834 exhibitors in 48,000 square meters of show floor across six Technology Zones, visited by a total of 43,691 unique attendees from 151 countries.



Opening of InfoComm 2022 in June, at the Las Vegas Convention Center. Again a well-attended show, but masks were back on. This show was basically a smaller extension for North America of the previous ISE 2022 show in Barcelona. Attendance was decent during the first two days (decreasing sharply on the third day), product launches were limited—but then again, there were two years of things to show and hear and discover due to the pandemic.



A smaller but busy show, CEDIA Expo 2022 in Dallas, TX, where exhibitors finally revealed new products. There were 317 exhibitors and approximately 12,000 visitors, over 40% of which were first-time attendees.

Q-LAN and Q-Sys quickly expanded to cover all the applications that QSC embraced, from portable PA systems and DSP amplifiers, to its prestigious Cinema systems. More important, Q-SYS enabled QSC to grow into the installation business, from commercial audio to meeting room solutions. Particularly since the acquisition of Attero Tech in 2019, QSC was able to grow its audio, video, and control networked ecosystem, while also boosting QSC's development capabilities with Attero Tech's Fort Wayne, IN-based engineering staff.

Today, the Q-SYS business division offers a complete cloud-manageable, software-driven audio, video, and control platform supported by a vast partner ecosystem. Its range of products includes Q-SYS Core Processors, NV Series and SPA-Q Series/CX-Q Series amplifiers, AcousticDesign Series loudspeakers, Q-SYS software licenses, Cinema products, and more.

Never Mind the Standards. Where's the Chips?

In 2020, the installation sector was excited and supportive of some of the open standard efforts and specifications promoted by industry alliances, and the convergence of industries. Also of note, commercial and residential installation sectors were at the forefront of exciting innovations in terms of networking, signal processing, and control, benefiting from the convergence with cutting-edge professional audio and consumer technologies.

Unfortunately, those efforts have been mostly kept quietly in the background, waiting for a better moment to generate impact in the industry. And although 2022 has seen the return of industry events, with two strong editions of Integrated Systems Europe (ISE) in May, in Barcelona, Spain, and InfoComm, in June, in Las Vegas, NV, those shows hardly allowed tracing major shifts in existing trends. ISE was a "return to business" event, with most companies focusing on showing all the products that were launched in the previous two years. InfoComm 2022, a month later, was more about corporate changes, and systems, and less about products. Only at CEDIA Expo 2022 in Dallas, TX, did we see the residential integration sector showing more products, but again, this was a much smaller show compared to ISE, which combines both markets and organizations.

And after all, even if there were new products unveiled at these shows, no one really could confirm when they would be available. The professional audio and installation industries, undergoing a transition to IT systems and networking, naturally suffered the impact of semiconductor and electronic component shortages. And in the last year, price increases and logistical challenges have made a strong dent on the recovery strategies, meaning that everyone at ISE, InfoComm, and CEDIA Expo was basically "selling" for the end of the year and early 2023.

Under the circumstances, it doesn't really matter if systems are designed using "proprietary" closed solutions or based on open-standards, and the industry has focused mainly on interoperability efforts and making sure installed systems met the specified requirements with available technology. Also, neither side has proven its case regarding the advantages of one perspective or another. While manufacturers that have relied upon market-standard solutions, such as Dante audio networking, were severely affected by the supply chain disruptions, Audinate, the Australian company that develops and specifies the technology, managed to navigate

those hurdles much better than the consortiums responsible for other industry standard technologies. Also, many of the efforts that depended upon “voluntary” cooperation of working groups were simply frozen under the unpredictability of the circumstances and the need to focus on more pressing matters.

Both the “proprietary” technologies and the open source or standard-based efforts were equally affected by the vanishing stocks of chips and unexpected end-of-life announcements for products that no one really knew were no longer being manufactured, and were simply subject to availability from existing stocks, as it was the case with some FPGAs, microcontrollers, and DSPs. Apparently, many semiconductor companies were manufacturing and “dumping” the whole production to global retailers and omni-channel distributors, without caring much about controlling inventory. When those organizations sold out the stock all at once and started asking the chip manufacturers for more, it was discovered that those semiconductor companies had no intention of resuming production of items they sometimes were not even able to manufacture.

Semiconductor analysts report that the global supply chain challenge is largely a demand issue more than a supply issue. We have now learned that the global supply chain is also a very disconnected and opaque structure, where supply is not really able to predict demand. And this is exacerbated in lower-volume industries, such as professional audio and video, because chip allocations are being diverted to large volume clients/industries, such as consumer electronics, automotive, and military. The supply chain challenges facing professional audio and AV manufacturers took on additional complexity and intensity in the second half of 2021 and into 2022, when the supply of particular chips used widely in the industry suddenly and nearly completely became unavailable.

The Audinate example ended up being a positive one for its ability to address the most important, complex, and dynamic commercial issues impacting the global audio and video industry. During InfoComm 2022, Audinate confirmed that it had instituted strategies that were implemented in advance of the chip shortages and have served to significantly mitigate the impact of a very challenging supply environment. Those included leveraging strategic partnerships with leading chip suppliers including Xilinx, NXP, and Silicon Labs/Skyworks, and continuously lobby for increased allocation of chips on behalf of the more than 500 Dante manufacturer licensees. This strategy has been successful in procuring inventory for several key parts for Audinate and its licensees. It has also allowed the company to ramp up manufacturing of several Dante products in March 2022.

At the same time, during InfoComm 2022, Audinate announced the Brooklyn 3, the next generation, drop-in replacement for the popular Brooklyn 2 module, as well as Fremont 3, the replacement for the Dante AV “Fremont” module, both available and in production in 2022. Audinate is also advanced in evaluating alternative options for its Broadway and Ultimo products.

In a forward-looking strategy, Audinate already had strong adoption of its software offerings with Dante Embedded Platform and Dante IP Core, and that adoption has only accelerated in recent months as manufacturers look to reduce the number and type of chips they need to procure and diversify their designs with different chip manufacturers. Dante Embedded Platform is a Linux software solution

running on off-the-shelf ARM and x86 processors from a variety of chip manufacturers, while the IP Core enables manufacturers to run the Dante image on a variety of pre-qualified FPGAs.

Connected Amplifiers

Given the evolution toward networked and distributed systems and even the transition to cloud connectivity, audio amplifiers are now an even more vital part in any audio installation. With pandemic pressures over the industry, the segment suffered a few drastic changes, from major acquisitions to new strategic horizontal expansions.

One of those horizontal moves, and quite surprising as such, was from Danish amplifier powerhouse Pascal Audio, which is an established OEM supplier to the professional audio industry. Pascal had already given signs that the company was not indifferent to the positioning of some of its competitors, which already offered complete solutions for the installation market. And the technology evolution and need for full software, cloud, and services integration paved the way to a new strategic move.

First, Pascal started to offer a wide range of solutions for commercial install products, with Pascal IP-branded solutions that ranged from simple power amplifiers to advanced DSP/matrix amplifiers with AoIP and network control solutions. Its ZoneControl



A batch of Dante Brooklyn 3 audio-over-IP networking modules, which are a pin-compatible replacement for one of Audinate’s most popular products, had to be redesigned due to supply chain issues, and to ensure continuous availability while delivering support for current and future Dante features.



Pascal Audio now offers a range of installation amplifiers under the Blaze Audio by Pascal brand. The PowerZone Connect range delivers up to 3000W total system power, offer multiple channel and power options, with software-controlled DSP, routing, and easy to configure sound zones.



Linea Research, a manufacturer of high-power Class-D amplifiers and digital signal processors, was acquired by Focusrite, causing ripple effects across the audio industry. A supplier of OEM amplifiers to audio companies from all over the world, its own branded products can be controlled by a range of third-party software, including QSC Q-SYS (pictured).

software provided installers with advanced features, allowing amplifier configuration, even wirelessly from any device without the need to download an app, as the software runs on any Internet browser. And when needed, companies could always acquire OEM amplifier modules for integration in their own boxes, which many large system integrators also do.

But at ISE 2022, Pascal announced an even bolder move, with the creation of Blaze Audio—a brand of Pascal A/S—specializing in professional solutions for the commercial audio market. At the core of the strategy, Pascal extended its existing range of installation amplifiers in the IP series to create the Blaze PowerZone Connect range of DSP amplifiers.

The newly released DSP amplifiers deliver up to 3000W total system power, offer multiple channel and power options, with software-controlled DSP, routing, and easy-to-configure sound zones. In this Market Update we are not going to get into deep details of these amplifiers, which were discussed in our dedicated article in the July issue of *audioXpress*. But suffice it to say that these Blaze PowerZone amplifiers are now being offered in direct competition not only to Powersoft and QSC products, but also as direct alternatives to the rising success of LEA Professional, the company that was reborn from the ashes of Harman/Crown to



British holding company Focusrite launched Optimal Audio to offer a complete portfolio of advanced commercial audio products, designed for a fully connected and remotely controlled systems approach. Optimal Audio offers three SmartAmps with DSP, multichannel direct drive constant voltage, optimized presets and with half-rack and full-rack options available.

create a new-generation amplifier platform, designed for cloud connectivity. In doing so, Blaze is also competing in the same systems market where many large and small audio companies that relied upon Pascal amp modules are doing business.

This is a bit strange because one of the fundamental features of audio amplifiers these days is the direct interface with solutions from other brands, and not only in terms of remote control. Audio amplifiers increasingly combine DSP and signal matrix/routing functions, and are normally already offered with a variety of tuning options, including dedicated speaker presets, being able to monitor and even optimize the whole system response, by managing power loads, channel bridging, and more. Maybe it could make good business sense in the long term for Pascal. It certainly is another disruption in a market that is quickly evolving.

And speaking of acquisitions, as mentioned in our Audio Amplifier report in July 2022, Focusrite plc, the British music and audio conglomerate company that acquired Martin Audio in 2019, has since acquired Linea Research Holdings, Ltd., a UK-based designer and manufacturer of professional audio equipment. Again, an acquisition that accelerated changes in the audio industry, and particularly in the installation segment.

Linea Research is known as a manufacturer of high-power Class-D amplifiers and digital signal processors, with a strong knowledge of audio networking and software products, including remote monitoring. Headquartered in Letchworth Garden City, UK, Linea Research was formed in 2003 by a team of experienced professional audio specialists that previously led the product development efforts at Harman's BSS Audio, and were responsible for creating world renowned products, such as the Soundweb and Omnidrive processor range, and the Varicurve multiband parametric equalizer.

In its fiscal year ended February 2022, Linea's unaudited revenue was £5.2 million, with its business resulting from both its own branded products, and also OEM products that it supplies to audio companies from all over the world, including Martin Audio.

Focusrite's subsidiary Martin Audio is a significant customer of Linea—Linea technology is behind the successful iKON amplifier series that powers Martin Audio's line arrays seen on live tours and at prestigious festivals. The acquisition strengthened the critical source of supply of amplifier modules for Martin Audio and enables greater integration of loudspeaker and amplifier technology in Martin Audio products.

Although Focusrite confirmed its intention to continue to develop the sales of Linea products through third-party distribution and to OEM loudspeaker customers, which included many famous professional loudspeaker companies in the UK, Germany, Italy, Spain, and Brazil, or Danley Sound in the US, as an example, many of these brands immediately initiated their own search for alternatives. When confronted with the supply chain disruptions and these corporate changes, the transition to developing systems in-house also started to make more sense to many companies.

It is important to highlight, every aspect of a Linea amplifier, in terms of hardware, software, and DSP algorithm creation, is produced by Linea's team, which have decades of combined experience in design and manufacturing. The integration of switched-mode power supplies (SMPS), very high power sonically transparent

Class-D amplifiers, networked digital signal processing, and the associated control software are all non-trivial and are all areas in which Linea Research is well recognized. These are the demanding technologies increasingly needed in modern audio systems, and Linea is able to offer them all. Apart from its own software, Linea Research amplifiers can be controlled by a range of third-party software, including QSC Q-SYS. Integration and interoperability efforts are naturally one of the victims of these strategic acquisitions that continuously mold the industry.

As the new owner of Linea Research, it is also important to note that the British holding company Focusrite plc has launched a new seventh brand, Optimal Audio, in 2021. Unlike the ongoing acquisitions, Optimal Audio was created from scratch using the group's resources, in a strategy to offer a complete portfolio of advanced commercial audio products.

As with the Pascal Audio decision to create Blaze Audio, Optimal Audio leverages the opportunities of fully connected and remotely controlled systems approach, with ease of installation in mind. Reflecting developments taking place over four years, involving extensive R&D across product and software, Focusrite (and Martin Audio, which is naturally very much involved) understood that commercial and installation audio required "a streamlined product offering working seamlessly together." And the clear focus is in systems integration that deliver not only high quality sound but a solution that is easy to install and can be operated by anyone, not just engineers.

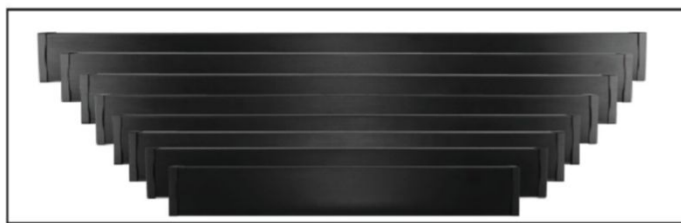
Creating a product ecosystem, Optimal Audio now offers a one-stop solution of control, amplification, and loudspeakers for small to medium-sized commercial installations, with a focus on supporting multi-zone venues. At the forefront of the eco-system are four- and eight-zone, powered and non-powered Zone controllers that have DSP at their heart and simple web control and apps. Optimal Audio decided to go a step further and even created dedicated wall-mounted controllers called ZonePads. And naturally, Optimal Audio offers SmartAmps with DSP, multichannel direct-drive constant voltage, and optimized presets and with half-rack and full-rack options available.

Origin Acoustics Expanded Scope

And because we are talking about both commercial and residential installation, and we highlighted the "ecosystem" approach that is followed by Optimal Audio (and also by many other manufacturers), we should also discuss the extraordinary expansion of Origin Acoustics.

During CEDIA 2022 (September 29 to October 1) in Dallas, TX, Origin Acoustics was one of the busiest exhibitors. Following the acquisition of Ambisonic Systems in August 2021, Origin Acoustics has now introduced an expanded range of loudspeakers, including architectural and landscape solutions with pro audio level specifications. It also relaunched renowned subwoofer brand Velodyne Acoustics, and expanded its partnership with Bang & Olufsen, among many other announcements.

As a quickly expanding manufacturer of architectural speakers, Origin Acoustics has demonstrated its evolution since the company's inception in 2014. While the brand's heritage dates back to the first architectural speaker manufacturer, the Origin brand has only been



Origin Acoustics now offers a new line of eight multichannel soundbar models with aluminum cabinets, using a patented InstaWidth design to match most TVs on the market.

around for eight years and is now one of the top brands used by today's custom installers.

The custom integration industry has seen many companies get acquired by larger brands, meanwhile Origin has increased its product and solutions portfolio by putting together a curated approach, which includes building strategic partnerships with some of the world's biggest brands and category leaders. Announced back in 2017, the first brand partnership was with Bang & Olufsen, the Danish manufacturer recognized globally in high design high-performance audio for nearly 100 years, with worldwide sales approaching \$1 billion dollars.

The partnership started with the development of a co-branded line of architectural speakers designed and built by Origin Acoustics, which was then sold by both Origin and Bang & Olufsen worldwide through each company's respective sales channels. Just announced prior to CEDIA, Origin Acoustics' relationship with Bang & Olufsen has evolved by becoming the North American Brand Manager and Master Distributor. Origin is now responsible for expanding the Bang & Olufsen's Custom Technology Installers (CTI) program, by bringing on new resellers to grow the Bang & Olufsen presence in the US.

Over the past eight years, Origin has continued its curation approach by partnering with companies such as Audio Control (amplifiers), Artnovion (acoustic panels), Amina Sound (flat panel speakers), Crestron (control systems), and most recently Austere (cables) and Flush Mount (designer switches and outlets). In addition, it acquired Ambisonic, a high-performance line of outdoor and commercial speakers, from Jeffrey James Coombs, best known as the founder of James Audio. Each partnership has its own special



Origin Acoustics impressive demonstrations of its latest Ambisonic Systems products—pro audio level and outdoor landscape solutions - an expanded range of installation loudspeakers.

value, but what they all have in common is a focus on performance, design, and being some of the industry's finest brands.

This curation approach of partnering with companies is different compared to the traditional acquisition model we have seen in the industry. When companies are acquired, often the people who built those brands leave, and the soul of the company goes with them. With Origin's strategic partnerships, the businesses continue to operate as they have, but now with a network of partners that bring added value to each other.

But that's only part of the evolution. With Origin Acoustics' rapid growth, its product line has evolved immensely, with an increased focus on design. Recognizing that the interior designer, architect, and builder are all playing a major role in influencing today's home buyer, Origin Acoustics developed what it calls design-centric technology solutions. Products following this design philosophy include the Bollard outdoor speakers and a new line of



Australian brand Blustream recently introduced the new Dante NPA70 compact two-channel amp with PoE++. This networked audio zone amplifier delivers advanced audio integration, featuring DSP, 31-band EQ, audio delay for lip sync correction and extensive control options. Blustream exhibited for the first time at CEDIA Expo 2022 with its range of signal distribution solutions for both residential and commercial applications on display.



APEX IP10 installation amplifiers can be monitored and controlled remotely from any device over the cloud.



APEX CloudPower four-channel Class-D amplifiers offer 350W to 3000W per channel, with fully integrated DSP and remote control/monitoring in the cloud.

eight multichannel soundbar models with aluminum cabinets, using a patented InstaWidth design to match most TVs on the market.

Core and Cloud Amplifiers

Reacting to this changing scenario, consolidation, and growing competition, manufacturers of original audio equipment for the installation market are all trying to expand their systems approach or expanding horizontally their product positioning.

Belgium manufacturer APEX, which until now mainly focuses on finished products and system solutions, is an interesting example. APEX decided to expand its reach by offering OEM amplifier modules, leveraging its unique GlidePath direct drive architecture, a proprietary Class-D technology where the DSP system is an integral part of the amplifier system, with a fully DC-coupled audio signal path.

Based in Beringen, Belgium, APEX has earned a reputation for developing high-quality, innovative speaker processing and network system management products for the AV and pro audio markets, both under its own name and in support of partner brands. Founded by managing director Paul Van Hees in 1991, in recent years APEX has seen strong growth, fueled by investment in R&D expertise and a strong culture of innovation.

The company's latest products include a pair of higher-powered models in its CloudPower four-channel installation amplifier range, which was presented at ISE 2022, together with a feature rich firmware upgrade for the series. The 4x 1500W CP1504 and 4x 3000W CP3004 can be monitored and controlled remotely from any device over the APEX Cloud, empowering integrators to make adjustments without the need for a site visit and creating opportunities for value-added remote monitoring services.

The latest APEX firmware release for CloudPower amplifiers, includes additions to the IntelliCloud remote WebUI with a mixer facility, and a built-in signal generator to assist with system setup and troubleshooting, among other new features. CloudPower amplifiers can be controlled over a network using the UDP protocol, allowing greater integration with third-party control systems, such as AMX and Crestron.

Following the same systems integration path, Powersoft has been perfecting its range of Dynamic Music Distribution (DMD) solutions, which allow system integrators the ability to scale inputs and zones conveniently and efficiently with dynamic routing capabilities embedded on its amplifier platforms. This makes it simple to dynamically route music and other signal sources along with control signals between different zones and across multiple amplifiers—all independently of source location and without the need for a centralized DSP. At ISE 2022, the Italian manufacturer highlighted the combined capabilities of Powersoft's Mezzo amplifiers, ArmoniaPlus software, WM Touch control panels, and SYS Control App in several interactive walls that represented different DMD application examples.

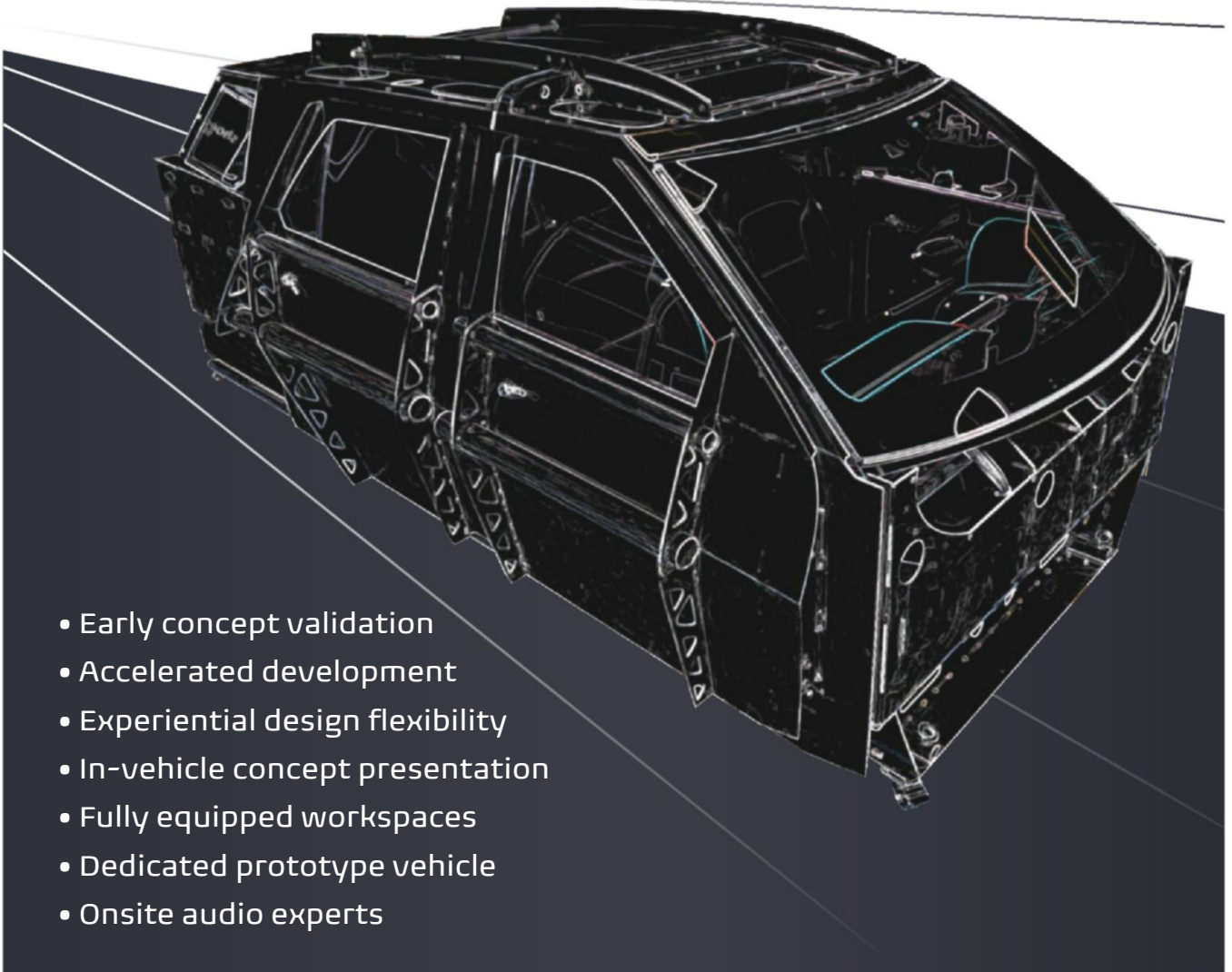
And its best-selling range of installation amplifiers continues to evolve, with Powersoft further enhancing its product range with AES67-compatible versions of its popular Duecanali and Quattrocanali amplifier platforms, which were previously available with Dante. The new models, called Duecanali DSP+ and Quattrocanali DSP+, are able to receive audio-over-IP (AoIP) streams from AES67-capable networked sources for increased interoperability across different systems.

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While AES67 is already the AoIP standard of choice for networked audio in broadcast systems, the technology has started to become widely supported by all manufacturers in professional audio and custom integration. Systems that use Ravenna or Dante, as well as networked solutions from QSC or Crestron, all support AES67 interoperability for audio distribution over IP infrastructures. All Powersoft amplifiers featuring a Dante card, including Duecanali DSP+D and Quattrocanali DSP+D models, can also be configured to work with AES67 audio streams by enabling AES67 mode in ArmoniaPlus or Dante Controller.

AES67 compliance for high-performance audio-over-IP networking is also supported by a free update to ArmoniaPlus, Powersoft's powerful systems design, management, control, and monitoring software.



Powersoft Duecanali and Quattrocanali DSP+ installation amplifiers are now able to receive audio-over-IP (AoIP) streams from Dante and AES67-capable networked sources for increased interoperability across different systems.



LEA Professional disrupted the installation market with its Connect Series networked audio amplifiers featuring advanced DSP and cloud and Wi-Fi connectivity. The concept accelerated the integration of distributed audio technologies. In 2023, the company will enter the residential integration space.



Theory Audio Design is now shipping its DLC-250.4 distribution amplifier and loudspeaker controller, ideal for distributed audio in any residential or commercial installation.

But the company that is now more widely recognized by its innovative approach to connected audio amplifiers is LEA Professional, which launched its cloud-connected range of products in 2019. All LEA Professional Connect Series amplifiers leverage the built-in Amazon Web Services (AWS) cloud platform that allows integrators to control and monitor critical operational data points that are integral to maintaining system health, securely and remotely. Not surprisingly, even with the global pandemic affecting the company's plans, LEA Professional quickly became a leading contender in the commercial installation space, and is preparing to launch in the residential space in 2023.

Since its launch, LEA Professional has consistently invested in improving safety and performance standards across its Connect Series range, meeting increasingly stringent worldwide standards and ensuring its amplifiers can be used in projects of all sizes. All Connect Series amplifiers feature cloud connectivity enabling users to harness the power of the cloud for remote control, monitoring, configuration, and more from any personal device. And firmware upgrades continuously improve the features, including adding the IEEE 802.1X security protocol.

The South Bend, IN, manufacturer recently started shipping its new Connect Series 1504, a new four-channel, high-power amplifier that is able to deliver 1,500W per channel (into 2Ω, 4Ω, and 8Ω as well as 70Vrms and 100Vrms) and is available in Network Connect (1504) and Dante Connect Versions (1504D). The 1504 and 1504D models include advanced DSP with linear phase FIR filters and up to 4 seconds of delay. The Smart Power Bridge feature of this amplifier provides flexibility allowing for two Smart Power Bridge channels that can turn the amp into a 2x 3000W amplifier. With three different methods of network connection, users can connect using the amplifier's built-in Wi-Fi Access Point, connecting to an existing Wi-Fi network, or using a Local Area Network Ethernet connection.

Theory Audio Design is another example of a company that decided to expand its range of branded products to increase its resilience from the supply chain. This premium audio brand from Lake Forest, CA, was created by loudspeaker engineer and industry veteran Paul Hales, who previously had founded the highly-regarded audio manufacturer Pro Audio Technology (PRO). Theory products marry PRO's acoustic performance and professional features with sophisticated contemporary design at a much lower cost, focusing on the installation market.

Among other electronics that are strategically specified to complete a systems approach by the loudspeaker company, Theory Audio Design launched the DLC-250.4 distribution amplifier loudspeaker controller, a flexible and powerful half-rack design that fits any residential or commercial installation where distributed audio is desired. Each DLC-250.4 offers four channels of intelligent power, a full DSP matrix, and is TCP/IP controlled through a wired connection, or directly via its on-board Wi-Fi access point.

The DLC-250.4 is designed to be at the core of every Theory distributed audio system. With channel-pair power sharing of up to 250W, each DLC-250.4 offers the flexibility to power any system size, operating in both 8Ω or 70V/100V modes, separately or in combination. Via its configurable outputs, the DLC-250.4 can power up to 16 Theory 16Ω ceiling or pendant loudspeakers in Lo-Z

mode, and up to 64 Theory architectural speakers in Hi-Z mode—or an optimal hybrid of both or in any quantity in between. This feature is especially beneficial in commercial spaces (e.g., bars, restaurants, retail environments, fitness centers, and spas) where a combination of power, from quiet background to loud foreground audio, is required. Theory also is launching a Dante-enabled version of the DLC-250.4, offering further installation flexibility by adding Dante network audio and AES67 compatibility.

A similar example of this connected and flexible approach in core audio amplifiers can be found in the expanding AudioControl range, which is being adopted by an increasing number of system integrators in North America and other continents.

Already in 2022, AudioControl has introduced the Architect Model P250EQ, a high-power two-channel amplifier engineered with an emphasis on sound quality plus key features for the custom installation channel. The foundation of the Architect Model P250EQ is the company's Constant Power Mode amplifier design, with an output power of 250W per channel into 8Ω, 4Ω, and 2Ω loads. The model offers selectable EQ curves to ease installation and setup of commonly deployed architectural loudspeakers, and offers an impressive array of inputs designed to accommodate every application.

And AudioControl CM Series amplifiers have established a new standard of performance in the realm of 70V amplifiers with its Constant Power Mode architecture featuring Gallium Nitride (GaN) semiconductors that deliver more output power, greater efficiency, and sound quality—far surpassing the performance of conventional 70V solutions. CM Series amplifiers feature optional Dante digital audio networking capability along with the flexibility of both 70V and 4Ω/8Ω. These ultimate-performance amplifiers are ideally suited for commercial and residential installations, delivering channel matrixing and a powerful, intuitive DSP engine. More examples of this flexible and connected approach can be found in the most specialized companies focused on installation, all the way to the larger audio companies such as d&b audiotechnik or L-Acoustics.

We already highlighted the latest 5D installation amplifier from d&b audiotechnik in our July Market Update. But to refresh your memory, the 5D is a four-channel DSP amplifier featuring integrated audio networking and flexible output power sharing—all within an ultra-compact design. This amplifier addresses a broad range of small- to mid-scale installation projects, and perfectly complements the German company's connected systems approach.

Equally advanced and versatile, L-Acoustics new LA7.16i multichannel amplifier controller is able to predict power and load requirements from all channels, surmounting the limitations of current high power amplifier configurations used in most projects. Unique in the market, the LA7.16i offers an unprecedented 16×16 architecture—each of the 16 output channels delivers up to 1300W at 8Ω or 1100W at 4Ω—in a 2U chassis with seamless networking redundancy.

With the new L-SMART amplification control, L-Acoustics also introduced a suite of advanced power management technologies that allow designers to mix and match any L-Acoustics loudspeaker onto one single LA7.16i amplifier. Hardware sensors within the amplifier supply real-time information, which is analyzed by the DSP to dynamically match loudspeaker system needs. The power supply unit (PSU) can then provide extremely high short-term



AudioControl introduced the Architect Model P250EQ, a high-power two-channel amplifier for the custom installation channel.

peak power or lower power over longer hold times. This energy is delivered intelligently to the advanced Class-D output stages, assuring optimum system performance at all times. And obviously the LA7.16i supports network connectivity, with the dedicated Milan-AVB technology for connection with other systems from L-Acoustics, and AES67 support for integration with the rest of the audio industry.

Crestron Whole-Home Audio

If there was a company that always led the way in systems integration it was Crestron Electronics. The Rockleigh, NJ, organization essentially carved the path for automation solutions, designing software and hardware for commercial and residential installation solutions. As the technology evolved to meet that same advanced integration vision, Crestron expanded its systems approach to embrace more networked solutions, with equipment from more manufacturers and more interoperability.

Recently, Crestron announced the DM NAX distributed AoIP amplifier, an all-in-one networked solution that provides all the connectivity and audio functionality integrators need for residential applications, with interoperability for both Crestron Home and Crestron



L-Acoustics new LA7.16i multichannel amplifier is described as an amplified controller. L-Smart is actually an acronym for L-Acoustics System Modeling Adaptive Resource Technology, a controlling system that monitors audio signals and manages all outputs in order to stay within the available power from the supply, whenever the system demands more power than can be provided.



The Crestron DM NAX distributed AoIP (AES67) streaming amplifier range includes the DM-NAX-4ZSP, DM-NAX-8ZSA for four and eight zones, respectively, and the DM-NAX-16AIN AoIP encoder that provides 16 local stereo audio inputs to a DM NAX Crestron multiroom audio distribution network.



Lithe Audio's new IO1 Wi-Fi speaker, the 70W Wi-Fi Multi-Room Micro Subwoofer and the Pro Series Wi-Fi Multi-Room Ceiling Speaker for residential installations. To expand its All-in-One (AiO) range of in-ceiling and in-wall speakers, Lithe Audio designed the versatile IO1 active speaker, offering a variety of mounting configurations, wired and wireless connectivity options, and high-quality components in one weather-resistant, IP66-rated speaker with an aluminum grille that requires only a power source for installation.



The Pro Series from Lithe Audio, with its PoE++ and WiSA bolt-on modules was presented at CEDIA Expo 2022 for the first time and immediately received multiple awards.

custom-programmed projects. We discussed the Crestron Home concept in our previous article focused on residential integration.

Crestron pioneered home automation and never stopped reinventing the smart home. With the new DM NAX, integrators can deliver complete control of the audio environment to users, allowing them to toggle between TV audio, streaming music, podcasts, alarms, doorbells, and even paging and intercom usage to any speaker in the house. A system that takes whole-home audio to a new level, it provides effortless audio distribution and is easy for dealers to configure.

Dealers who have worked with Crestron's DM NVX products will find familiarity in DM NAX, which will integrate simply with legacy Crestron systems. Allowing for audio from any input to be sent to any output, the new DM NAX provides the ability to scale to 256 stereo zones. And all DM NAX products are interoperable with AES67 and Dante audio networking, use hardwired connections, and require no custom network architecture or hardware. This allows integrators to use DM NAX to specify an entire house's worth of audio systems with a single device.

Beyond native integration of music streaming services, DM NAX allows for selectable speaker profiles throughout the house once paired. This feature will automatically optimize output on a speaker-by-speaker basis as integrators test settings in each environment throughout the space. Other features include the choice of up to 40 built-in chimes for alerts such as doorbells and door openings, among many other smart-home integrations that will be uploaded with future firmware updates.

Crestron also expands this audio-over-IP distribution platform with a robust audio portfolio that includes Crestron Speakers made by Origin Acoustics as well as an all-weather model of Crestron's Saros Surface Mount Speakers, all built natively for the Crestron Home OS and for easy integration with custom systems. The addition of these Crestron-branded Ultimate Series Speakers includes an active subwoofer, in-ceiling speakers, and a brand-new dual tweeter version of the Crestron Ultimate IC8 speakers.

Multiroom Audio Speakers

While some companies are focusing on the core components, which are networked amplifiers with built-in matrixing and processing, other audio companies are basically integrating all the distributed audio features directly inside all-in-one speakers.

Lithe Audio, the UK-based manufacturer of wireless audio systems for residential and commercial installations, has pioneered this approach directly into in-ceiling, in-wall speakers, by creating a modular platform and combining the most advanced wired and wireless connectivity options.

Recently, Lithe Audio launched its complete lineup of all-in-one ceiling, surface, and garden speakers in the US market. Featuring an integrated amplifier, connectivity, and high-quality speaker components in one device, each system requires only a power source for installation, making it simple for integrators to deliver audio streaming in single rooms, distributed systems, and outdoors while enabling the creation of wireless surround-sound home theater systems.

After having successfully launched its first Bluetooth ceiling speaker in the European market in 2017—which continues to be

the company's best-selling product given consumers familiarity with Bluetooth—Lithe Audio expanded its approach with a range of all-in-one multiroom audio systems. The company's in-ceiling speakers feature woven glass fiber woofer cones, titanium tweeters, and Class-D amplifiers pre-tuned for in-ceiling use, with Alexa and Google support for voice control. The company's Bluetooth 5 Ceiling Speaker offers a single-room solution for streaming music in kitchens, bathrooms, and hotel rooms, while its Wi-Fi Multiroom Ceiling Speaker adds built-in support for AirPlay 2 and Chromecast for the creation of distributed audio systems in the home. The speaker is available with drivers for integration into Control4 and Fibaro control systems, with support for additional solutions already planned.

At the top of Lithe Audio's ceiling speaker line is the Pro Series Wi-Fi Multi-Room Ceiling Speaker for residential installations, which offers all the same features as the Wi-Fi speaker while adding support for full high-resolution audio at 24-bit/192kHz, an optical input, and two Ethernet ports—one for a hardwired network connection and one for daisy-chaining another speaker. For additional functionality, the Pro Series is available with PoE++ and WiSA bolt-on modules. The PoE++ bolt-on adds PoE capabilities that deliver up to 70W to each speaker, while the WiSA bolt-on allows integrators to easily incorporate the speaker into wireless Dolby 5.1, Dolby 7.1, and Dolby Atmos surround-sound home theater systems. To add extra bass, the 70W Wi-Fi Multi-Room Micro Subwoofer—small enough to fit under a bed or behind a couch—can be wirelessly connected to Lithe Audio's Wi-Fi ceiling speakers.

For areas that require high-end audio quality but can't accommodate a traditional ceiling speaker, Lithe Audio's new IO1 speaker offers the same technology as the company's Wi-Fi Multi-Room Ceiling Speaker but can be installed in a number of configurations to meet the needs of any indoor or outdoor space—including vertically or horizontally on walls, flushmount, in corners, mounted on a yard stake, and more.

By only requiring power, the company's wireless audio systems make installing a speaker no more difficult than mounting a ceiling light, while their embedded technologies allow for the simple creation of distributed audio systems, surround-sound theaters, and more.

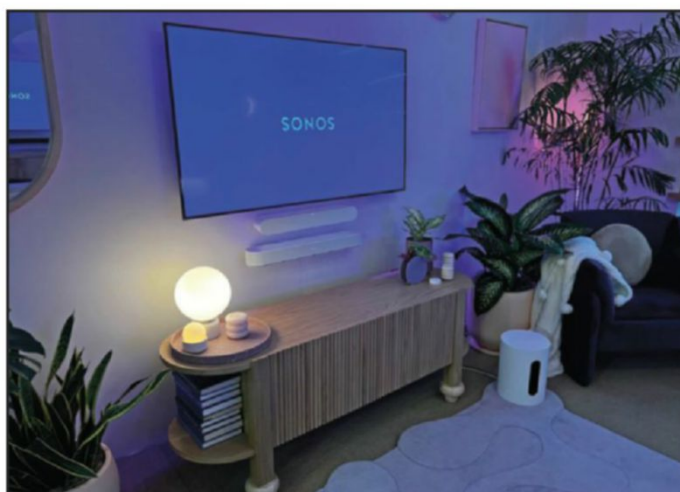
Distributed Audio

When we discuss whole-home, multiroom, or distributed audio systems, we normally are talking about solutions that are designed for fixed installation and that typically will use some form of wired connectivity and networked access. We discussed Lithe Audio as an example that achieves the same goals using technologies based on wireless standards, normally over Wi-Fi or in the same band, including the U-NII 5GHz spectrum, as is the case with WiSA HT technology. Other than that, there are proprietary protocols that include Apple AirPlay, or Google Chromecast, which have limited functionalities when it comes to relaying audio on a wireless network. And there is a reason.

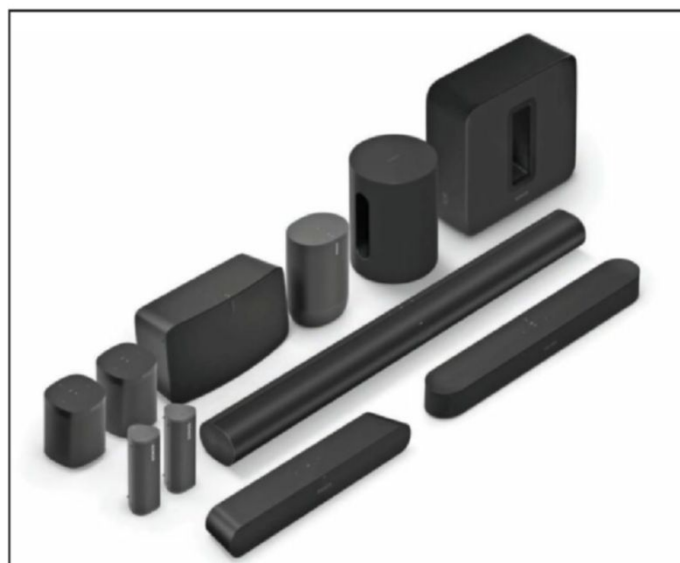
To stream audio over wireless LAN technology, all companies face the original technology pioneered by Sonos since 2005, which owns a solid portfolio of patents when it comes to streaming audio over a wireless LAN. We discuss that topic in more detail in our second article dedicated to wireless audio. Sonos defends its patent portfolio



The Dynaudio Focus series of fully active wireless stereo speakers are designed too be all-in-one connected music streaming systems.



Sonos' demonstration book at CEDIA Expo 2022, displaying the full range of wired and wireless streaming solutions, including the latest generation of portable (battery-operated) speakers.



The Sonos family of products in 2022, including the latest Sonos Sub Mini wireless subwoofer.

vigorously, and successfully, and most companies active in wireless multiroom have to license Sonos technologies, one way or another.

It's important to mention that fact to understand why Sonos systems are so frequently found even in commercial audio installations, and why Sonos continues to be a major player in the installation sector and a regular exhibitor at trade shows, such as InfoComm or CEDIA Expo.

As an example of the relevance of Sonos in this space, even the prestigious British manufacturer Meridian Audio has certified its high-performance audio solutions in the "Works with Sonos" program, offering an integration for its premium audio equipment to seamlessly connect with the Sonos sound system. This new integration makes it easier for Sonos users to enjoy Meridian's high-performance audio using the familiar Sonos interface, while streaming from more than 100 music services available and enjoying the ability for Meridian speakers to instantly wake and change inputs by simply pressing play in the Sonos app.

The "Works with Sonos" program certifies products that seamlessly connect with the Sonos home sound system. Now with added capabilities to automatically wake and change inputs, Meridian products can become part of an existing Sonos system or be the start of a new one. The integration is available by simply pairing a Sonos Port or Connect with a Meridian 218 or 251 controller, and any loudspeakers that are connected to the controller.



The complete family of Bluesound BluOS products, including the latest Node wireless hi-res multi-room music streamer, PowerNode wireless hi-res multi-room streaming amplifier, PowerNode Edge "just-add-speakers" streaming amplifier, and Pulse soundbar.



The Bluesound Professional B170S networked streaming stereo amplifier is a complete solution for a commercial music installation. The B170S combines a high-quality stereo, low impedance, and constant voltage amplifier with a BluOS-enabled network music player.

Bluesound and BluOS

One of the organizations that attempted to challenge Sonos' IP in the courts was Lenbrook, the owner and manufacturer of prestigious brands such as NAD Electronics, PSB Speakers, and the Bluesound wireless high-resolution multi-room audio systems, which have meanwhile expanded into commercial audio with Bluesound Professional. Bluesound and NAD products all use the BluOS hi-res distributed audio platform, which was also adopted by other leading premium audio brands, such as DALI and Monitor Audio.

On July 30, 2020, Sonos and Lenbrook Industries announced they had reached a settlement in Sonos' patent infringement lawsuit against Lenbrook Industries and, as part of this settlement, Lenbrook pays for a license to Sonos' patents for all BluOS-enabled devices. In contrast, Sonos continues to fight against Google in court to protect its wireless audio patents, forcing Google for now to remove all wireless multiroom features from its range of smart speakers.

Anyway, before and after this agreement, Lenbrook has been extremely successful with its Bluesound, Bluesound Professional, and BluOS ecosystem, which continues to expand every year and is gaining more partnerships and support in the residential and commercial installation industries. In its approach to market, Bluesound combines the best of wired and wireless infrastructure technology, depending on the use case. Bluesound Professional designed a systems approach to installation audio using a fusion of hardware and software that is purpose-built for high performance networked audio. Ideal for retail shops, bars, restaurants, hotels, gyms, and other commercial applications. The multi-zone audio management software platform, Powered by BluOS is native in a comprehensive range of controllers, streamers, amplifiers, and speakers.

Originally unveiled in March 2022, the B170S Networked Streaming Stereo Amplifier by Bluesound Professional is now available, introducing a high-quality, low impedance, and constant voltage amplifier with a BluOS-enabled network music player built-in. This makes the B170S module an all-in-one streaming solution for any commercial music installation.

The low impedance output is perfect for applications where intelligibility and sophistication are important. Alternatively, multi-speaker installations (e.g., corridors or high-traffic areas) can be catered to with the constant voltage option that allows for daisy-chained speakers, reducing cable lengths and complexity. With keyhole slots allowing for flexible mounting options, the B170S is well-suited to hide under shelves, tables, and walls or be rack mounted into a 3U space with the racking accessory.

Expanding on Bluesound Professional's multi-zone commercial music solutions, streaming services (e.g., SoundMachine, Custom Channels, SiriusXM for Business, QSIC, Tunify, and others) are natively integrated into the BluOS-enabled network music player in the B170S and can be controlled with the BluOS Controller app on PC, Mac, iOS, and Android. With a 6W standby mode, the B170S features secure network preventing unauthorized control, and supports multiple third-party control options compatible with PC, Mac, iOS, and Android.

Note that such a system can receive wireless streaming sources and can be wirelessly controlled by the user's devices, but it doesn't necessarily stream audio wireless from point to point, since it

normally is connected to wired—networked—speaker configurations. One example is the recent BSP200 PoE Network Streaming Speaker that Bluesound Professional launched for distributed audio systems in commercial spaces. Because the Bluesound BSP200 is powered by Power over Ethernet (PoE+) allowing a one-wire connection to the speaker for both power and audio, it doesn't need wireless streaming. Only the control is wireless.

The player in the BSP200 speaker is based on the BluOS platform and is capable of playing and distributing content from local network storage and streaming services. In addition to stand-alone use, a pair of BSP200s can be used in a stereo configuration. The BSP200 can also be used in a larger BluOS system, allowing for grouping into user-definable groups of whatever size is required with other BSP200 units and/or any other connected Bluesound Professional players on the network in an installation. Built-in DSP allows for equalization as well as other audio processing. Access to content is via Ethernet for glitch-free streaming. A USB input further adds to the capability of the system by providing replay of files stored on connected drives. The player supports playback of Hi-Res Audio content up to 24-bit/192kHz as well as MQA content decoding.

When it comes to residential audio and whole-home audio, the original Bluesound brand is the one that makes use of wireless audio streaming, given the more limited scope of the application and the convenience factor for the installation. But Bluesound also advertises the fact that its systems all support high-resolution audio wireless multiroom as a differentiator. The BluOS ecosystem is the only platform in the world that can support up to 64 players, all streaming in 24-bit hi-res audio. To support that vision, the company offers a comprehensive range of solutions, which includes the latest HUB, a network accessory that allows connecting a variety of analog and digital audio sources to the Bluesound ecosystem for multiroom streaming.

The Bluesound HUB offers a wide range of inputs on the back panel, including HDMI eARC, SPDIF coaxial and optical, stereo analog, and a moving magnet phono stage, unlocking the possibility to stream vinyl wirelessly. Each HUB manages one networked audio source, and up to four HUBs can be added to a single network. Using the BluOS Controller, users can manage those audio sources and quickly configure a whole multiroom system from a smartphone, tablet or computer.

Inside, the HUB is powered by a powerful ARM Cortex A53 processor and its module offers dual-band Wi-Fi and Gigabit Ethernet connectivity to cover the best possible integration. It is powered from any external USB-C power supply, and can easily be placed on technical cabinets when required.

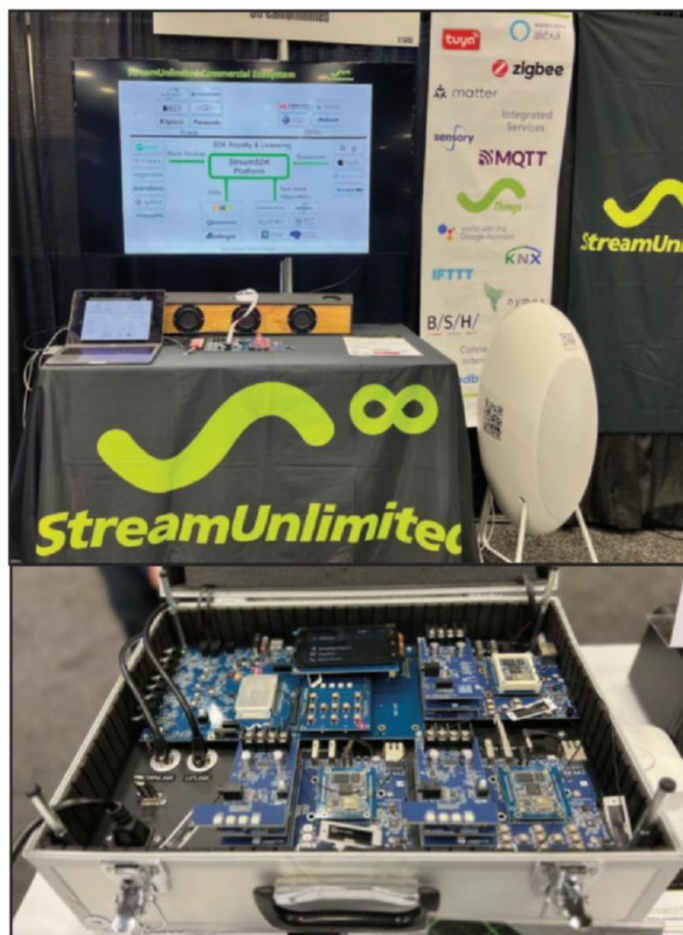
Other recent products in this BluOS ecosystem are the Bluesound Node Wireless Hi-Res Multi-Room Music Streamer and PowerNode Wireless Hi-Res Multi-Room Streaming Amplifier. Updated versions of the original products, these include new 32-bit/384kHz DAC stages, more powerful quad-core processors, and touch panel controls with presets and proximity sensors, while the PowerNode now offers substantially more power.

The Node supports 24-bit/192kHz audio processing, MQA decoding, and is equipped with an array of digital and analog inputs and outputs, supporting wired and wireless headphones,

speakers and even subwoofer outputs with crossover optimization. Users can bring music into their Node using Dual-band Wi-Fi or Gigabit Ethernet, Apple AirPlay 2, aptX HD Bluetooth, or USB Type A connectors. The Node also features an HDMI eARC input on the back panel and is an ideal multi-room hub.

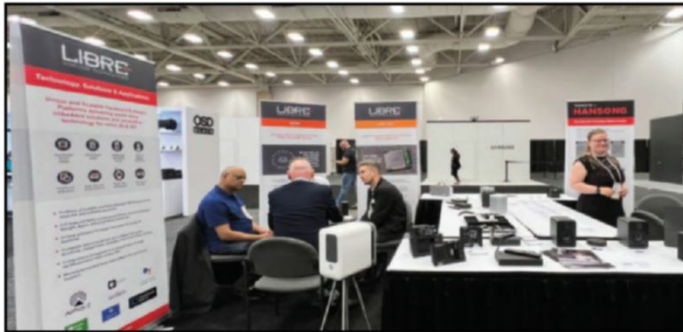
The Bluesound PowerNode amplifier combines all the features of the updated Node, to be used as a hi-fi amp, reinvented for the streaming age. The updated design also combines multi-room music capabilities to create an intuitive “just add speakers” hi-res sound system with 80W per channel. It can power a traditional two-channel or 2.1 wired or wireless systems, and can even be combined to a pair of Pulse FLEX 2i speakers to provide rear channel additions for a wireless Dolby surround sound home theater experience. And combined with other Bluesound players around the home, it can become the center of a multi-room audio solution.

More recently Bluesound announced the Povernode Edge to expand its hi-res wireless multiroom line of audio products, with similar features than the Povernode, with a conservatively rated 2x 40W. This is a new single-zone, wireless music streaming amplifier that offers a perfect entry point into “just add speakers” hi-fi systems, combining streaming sources, control, and amplification



Development boards and modules from StreamUnlimited, covering all the most desirable technologies for wireless audio streaming, including AirPlay 2, Chromecast, Spotify Direct, AMHD, music streaming implementation, and also voice control.

all in one affordable device. It makes it easy to connect a pair of passive speakers to create or extend any home audio system and even allows connecting the Bluesound Pulse SUB+ wireless subwoofer for a powerful but compact 2.1 stereo system. Alternatively, the Powernode Edge can power the rear channels of a Bluesound wireless surround home theater sound setup.



The joint LibreWireless and Hansong booth at CEDIA Expo 2022 displayed the full range of unique solutions for the residential integration market, including the latest Libre LS10 platform for streaming audio and wireless connectivity.

Resources

APEX | www.apex-audio.be
 AudioControl | www.audiocontrol.com
 Bang & Olufsen | www.bang-olufsen.com
 Blaze Audio | www.blaze-audio.com
 Bluesound Professional | www.bluesoundprofessional.com
 Creston | www.creston.com
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 QSC | www.qsc.com
 Sonos | www.sonos.com
 SteamUnlimited | www.StreamUnlimited.com
 Theory Audio Design | www.theoryaudiodesign.com
 Vanco International | www.vanco1.com
 WiSA Technologies | www.wisatechnologies.com

The Bluesound Powernode Edge was designed to be a key component in multiroom installations and expand the possibilities of BluOS as a versatile whole-home, hi-res audio platform. For installers and system integrators, the Powernode Edge can be integrated with the most popular smart home controllers (e.g., Crestron, Control4, URC, RTI, ELAN, and others), while hands-free control is possible with voice assistants such as Apple's Siri, Amazon's Alexa, and Google Assistant.

Engineering Support for Manufacturers

To support manufacturers with this type of whole-home audio platforms, multiroom systems, and all wireless streaming technologies, companies such as StreamUnlimited and LibreWireless, offer a complete portfolio of modules and integration options.

StreamUnlimited, located in Vienna, Bratislava, Lași, Shenzhen, and Santa Clara, CA, originated as a team of 25 engineers and in 16 years has built its own IP while more than quadrupling its size. The company supports a global cross-section of high-end audio, CEDIA-channel, premium and mainstream consumer electronics manufacturers. At the core of its offering is StreamSDK, a flexible software solution, portable to various classes of silicon, which enables the application of streaming audio functionalities. Its software stack supports the latest features, including Apple AirPlay2, Chromecast built-in, MICOSS Amazon Alexa Voice Service (AVS) MRM (Multi-Room Music), Tidal Connect, expanded voice assistant support, and new immersive audio integrations.

Alongside streaming and Internet music services, StreamUnlimited's StreamSDK helps manufacturers support high-resolution audio, multiple connectivity options, Android/iOS applications, and integration with smart home ecosystems. In combination with StreamUnlimited's engineering expertise and a family of ready-to-use hardware modules, StreamSDK also benefits from actual implementation projects and its features continue to grow.

Another important resource for many manufacturers is Libre Wireless, based in India, which recently became part of Hansong, one of the largest ODM and OEM manufacturers of audio, AV, and installation products, with a vast experience in supporting wireless technologies. Hansong Technology has facilities in Asia and all over the world, with direct investments in other companies.

At the recent CEDIA Expo 2022, Libre Wireless demonstrated its new LS10 platform, which now supports even more streaming music services and features. From GoogleCast, AirPlay2, Roon, Amazon Alexa, and more integrations, the platform now benefits from a Gracenote implementation that enables music fingerprinting so users can retrieve artist, album, album cover art, and track name on display-based products. LS10 is available for testing by manufacturers with the Libre evaluation kit (EVK).

If Wi-Fi 6 is a requirement, Libre Wireless offers its new LS11 module, based on the Libre LS10 platform. The LS11 option offers all the functionality and features of the platform, providing the foundation for a host of current and emerging uses from Wi-Fi 6 applications requiring high bandwidth and low latency. Together with Hansong, which has vast experience in WiSA integration and wireless streaming, Libre Wireless now can offer an expanded range of engineering and manufacturing options for brands.



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The Proof Is in the Testing

By
Adrián Lara Moreno
(Holoplot) and
Christian Bellmann
(Klippel)

Figure 1: Illuminarium Atlanta is at the forefront of fully immersive experiences, placing audiences “inside” the narrative with techniques used in traditional motion picture production and virtual reality that facilitate the experience of real-world, filmed content and authentic, re-created worlds in an immersive environment.

The Klippel Near Field Scanner (NFS) System was used for the development of Holoplot’s X1 Matrix Array compensation algorithms, which was the topic of an article in the September 2022 issue of *audioXpress*. This joint article by Holoplot and Klippel offers a deep dive into the one of the many benefits of the innovative technology from the Berlin, Germany-based company.

Holoplot is a Berlin, Germany-based pro audio company and manufacturer of innovative 3D professional audio technology, the X1 Matrix Array. X1 is made up of two audio modules—Modul 96 (MD96), a two-way module integrating 96 drivers in a two-layered matrix design, and Modul 80-S (MD80-S), a three-way module with 80 drivers in its first two matrix layers and a sensor-controlled subwoofer driver in its third layer. This matrix arrangement of loudspeaker drivers of different sizes—in a unique multi-layered configuration—enables control over sound propagation across a large frequency range, while providing the SPL and quality desired for high-performance sound reproduction.

Opposed to conventional loudspeakers relying mostly on hardware design, X1 is software-driven, combining Wave Field Synthesis and 3D Audio-Beamforming to achieve authentic sound localization and complete control over the beam shape. It’s the first sound system to decouple sound from its source, enabling the positioning of virtual loudspeakers within a space, creating sonic freedom for the user and exceptional experiences for the audience. X1 is able to generate multiple

simultaneously running sound fields, each with its own content, level, shape, and position—all generated from as little as one module.

To develop such complex solutions, a balance between simulation and measurement during the development process is key to ensure continuous verification and improvement of the models and algorithms as well as the enhancement of sound quality. This process requires an extremely advanced measurement solution, which is why Holoplot turned to Klippel for its expertise.

With the Near Field Scanner (NFS) System, Klippel—an innovative leader in providing unique test equipment for electro-acoustic transducers and audio systems—has created a new approach for directivity testing in loudspeakers. This measurement system has been instrumental in the ongoing development of Holoplot’s X1 Matrix Array.

The NFS System is considered a state-of-the-art technology to capture the full acoustic characteristics of transducers used in every loudspeaker design.

From early on, the development of X1 and the underlying Holoplot technology has been served by the data acquired using Klippel’s NFS

Carousel. On the design side of the Matrix Array, the electroacoustic design, modeling, and validation were a complex procedure due to the multi-layered design of the array, especially the MD80-S one with the subwoofer driver.

Acoustically speaking, the integration and interference between each speaker layer in the Matrix Array are extremely important to capture in order to achieve the best simulations and performance of the driving algorithms. The underlying optimization algorithms built into the Holoplot technology require the highest precision, three-dimensional information from each transducer-type built into each of the X1 modules.

The benefits that Klippel and the NFS offered Holoplot are best explained by the collaboration and testing around the Illuminarium Atlanta deployment (**Figure 1**). Illuminarium is at the forefront of fully immersive experiences, placing audiences “inside” the narrative with techniques used in traditional motion picture production and virtual reality that facilitate the experience of real-world, filmed content (e.g., a safari) and authentic, re-created worlds in an immersive environment.

The Challenge

X1 contributes to this experience by delivering unprecedented authenticity of sound reproduction, perfectly matched with the video projection. While Holoplot’s Wave Field Synthesis and 3D Audio-Beamforming technology enables the creation of authentic audio objects, the immersive nature of Illuminarium Atlanta’s Wild: A Safari Experience presented an additional challenge to solve—finding a way to completely hide the X1 modules installed throughout the venue behind a seamless projection surface, without compromising on the system’s peerless sound quality and control.

Concealing any sound system behind an acoustically semi-transparent panel leads to absorbed or reflected acoustic energy, resulting in gain changes in the audio spectrum. The Transmission Function (TF) is the frequency-dependent reduction or amplification in the sound level of a sound source as it passes through the panel used to conceal the sound system.

Traditionally, the compensation of the concealed speaker’s TF is achieved by equalizing the average TF across multiple angles or simply taking the TF on-axis and applying the inverse curve as the profile of an equalization stage. A preliminary assessment of the TF in the anechoic chamber led to the conclusion that the evaluated panel introduced very different gain variations to varying angles for the same frequency.



Figure 2: The X1 Matrix Array is shown in situ at Illuminarium Atlanta, placed behind a specially designed MDI panel.

As a result, the spectral balance across the audience area would be considerably different at different angles and distances to the concealed audio module, reducing spectral homogeneity. A TF compensation as described would not be sufficient, but rather an angle-dependent, Spatial Transmission Function compensation would be required.

The Solution

Holoplot Wave Field Synthesis and 3D Audio-Beamforming technology rely on high-resolution sensitivity and 3D directivity balloons from the transducers built into the audio module. Holoplot obtained the necessary data from Klippel’s NFS service during the design and development stage of X1. Using 3D Audio-Beamforming algorithms, Holoplot can define custom-shaped wavefronts using level and phase manipulation that fully adapt



Figure 3: NFS measurement setup is shown for a full-space 4π scan of the X1 audio module.

to the audience area. Furthermore, the resulting wavefronts are optimized for spatial and spectral homogeneity by taking a reference target curve.

Once the Spatial Transmission Function compensation became a challenge to solve, an opportunity arose; a solution to improve the spectral balance in 3D space of concealed audio modules already seemed to be part of Holoplot's core technology capabilities.

If Holoplot's algorithms were aware of the Spatial Transmission Function introduced by the acoustic panel in front, the optimization and equalization engine would compensate for the effect of the panel in any direction, and not only on-axis, performing similarly as if the panel would not be present. Changes in the transducer's radiation balloon caused by panel resonances, reflections, or acoustic absorption at specific angles would be known in advance and partially counteracted to achieve the desired spectral profile across the audience area.

This approach presented a requirement to be solved by Klippel—capturing the transducer's directivity balloon when deployed behind the carbon fiber panel as used in the actual application at Illuminarium (**Figure 2**).

The Near Field Scanner

Klippel's NFS Carousel uses a holographic measurement approach to determine the directivity of a loudspeaker. This method uses special solutions of the wave equation (spherical harmonics, Hankel function) to determine the 3D sound pressure output of the audio device. Compared to traditional measurement techniques this provides more comprehensive and accurate measurement data while minimizing costs (e.g., for an expensive measurement room) and measurement time.

The device under test (DUT) remains at a fixed position in the center of the scanner. This simplifies the handling of heavy devices and ensures constant room excitation and therefore constant

room reflections during the scanning process. The robot arm moves a microphone around the device under test and captures the sound pressure in the near field.

Due to the scanning along a double layer, the Direct Sound Separation (Klippel's patented technology) uses the additional phase information to detect the direction of the sound wave and can remove all room reflections from the direct sound of the loudspeaker. Thus, the measurement system provides accurate free field data in any environment (e.g., a workshop or office).

The Measurements

To answer the question of the interaction and influence of an acoustically semi-transparent panel on the audio quality of X1 a few different testing scenarios were run.

At first, a standard full-space 4n measurement was performed with the DUT placed in the center of the NFS Carousel. **Figure 3** shows the X1 Modul 80-S mounted on the scanner.

Second, the X1 audio module was placed behind the semi-transparent panel, mounted in a baffle. Usually, half-space scans in 2π are performed in a similar setup with a baffle mounted in the center of the NFS Carousel. However, the overall size and weight of the construction and the special positioning conditions of the X1 audio module made this setup difficult. For this reason, the baffle construction was built up next to hardware on the floor of the workshop and the microphone arm of the scanner was turned to the outside by additional poles to put the microphone into position (**Figure 4a**). **Figure 4b** visualizes the movement range of the microphone in this setup (the blue surface).

The sound field is scanned on a hemisphere in front of the baffle. By using symmetry conditions of the basic functions, the same spherical wave expansion can be adapted to such a half space measurement in 2π .

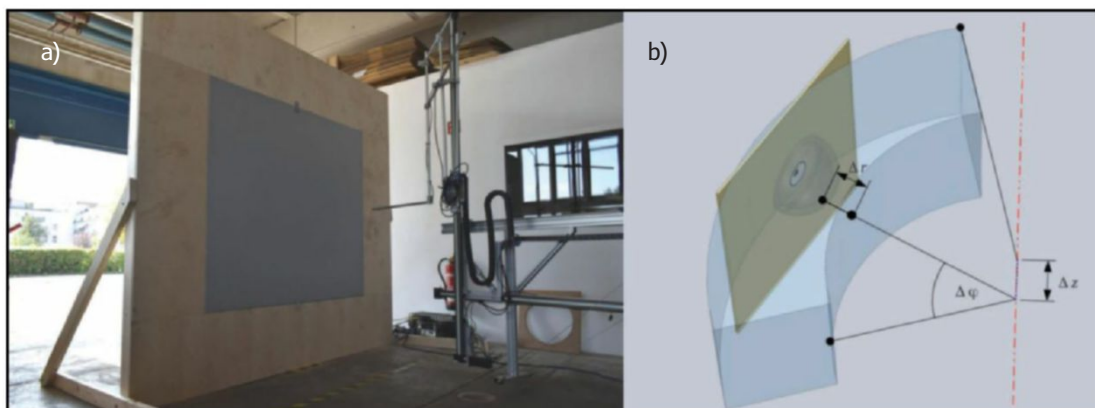


Figure 4. NFS measurement setup for half-space 2π scan in front of the acoustically semi-transparent panel (a). This graph (b) visualizes the movement range of the microphone in this setup (blue surface).

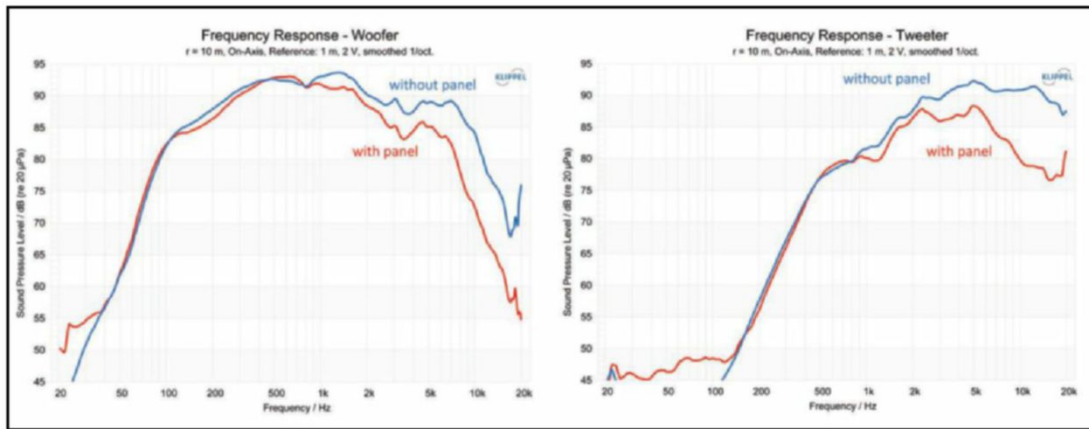


Figure 5: Frequency responses of a woofer (a) and a tweeter (b) are shown with and without acoustically semi-transparent panel—signal processing not included.

To achieve perfect half space conditions, the NFS measurement principle only requires a baffle that is slightly larger than the measurement surface, because effects of the baffle (e.g., diffraction) can be removed by Klippel's Direct Sound Separation technology.

For the measurement, the Klippel Analyzer (KA3) was directly connected to the speaker terminals of the individual transducers. Such a setup is useful to get a clean set of acoustic transfer functions without any influence of a DSP or smart amplifier. In total, multiple transducers were scanned as well as tweeters and woofers, which can be comfortably done using the Klippel Multiplexer hardware to switch between the drivers and shortcut inactive channels automatically.

The Results

An exemplary pair of woofer and tweeter was used to evaluate the effect of the acoustic panel used to conceal the audio module. It's important to note that, as the nearfield measurements do not include any signal processing, the spectral performance out of the transducer's working range is shown as well. The frequency responses of each transducer with and without the acoustic panel are shown in **Figure 5**.

Comparing the acoustical output of both measurements shows a transmission loss in the on-axis frequency response. Using a traditional approach, these frequency responses would be a base to calculate the transmission gain and compensate for this loss of energy with a DSP. However, considering also the off-axis responses, the acoustic semi-transparent panel causes other disturbances. There are additional resonances that affect the radiation pattern at certain frequencies, especially in the range between 2kHz and 5kHz. Above 7kHz the measurements show higher transmission losses on-axis than off-axis, resulting in a lower directivity index and a slightly wider coverage angle when the panel is attached.

The Spatial Transmission Function of the acoustic panel shown in **Figure 6** pictures the angle dependency of gain changes across the spectrum.

The Spatial Transmission Function is the absolute spectral gain difference between the bare transducer and the same transducer behind the acoustic panel after applying frequency smoothing of 1 octave and spatial smoothing of 15°. The spatial smoothing was applied so that isolated artifacts introduced by the panel used for measurements would not be introduced in the generic compensation to other panels with different properties: bracing differences, panel stiffness, and manufacturing variations or slight panel positioning differences.

To illustrate the benefit of the Holoplot 3D Audio-Beamforming approach compared to conventional audio solutions, the 3kHz frequency has been used as an example in Figure 6. The level difference between the 0° (on-axis) and 45° is approximately 2dB; hence, any global spectral correction at 2kHz would work effectively for one angle but over, or under-compensating at other angles.

Spatial Transmission Function differences are challenging to solve with a single, global equalizer. On the other hand, using Holoplot's 3D spectral

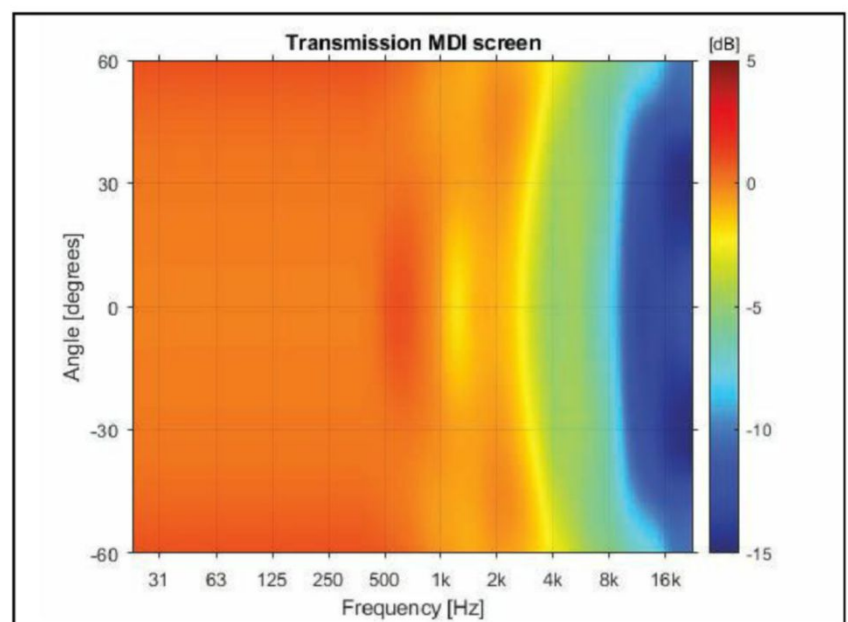


Figure 6: This is the Spatial Transmission Function of the MDI Strong panel.

compensation as part of the optimization engine, the transducers used to render the beam, can be individually spatially equalized, resulting in optimal spectral balance as the listener moves across the audience area.

Once the X1 transducer balloon data is corrected using the Spatial Transmission Function of the panel and included in Holoplot's algorithms as an audio module variant, concealed X1 audio modules can be optimized, simulated, and benchmarked.

Figure 7 shows exemplary transfer functions of an optimized beam with a 120° opening angle under

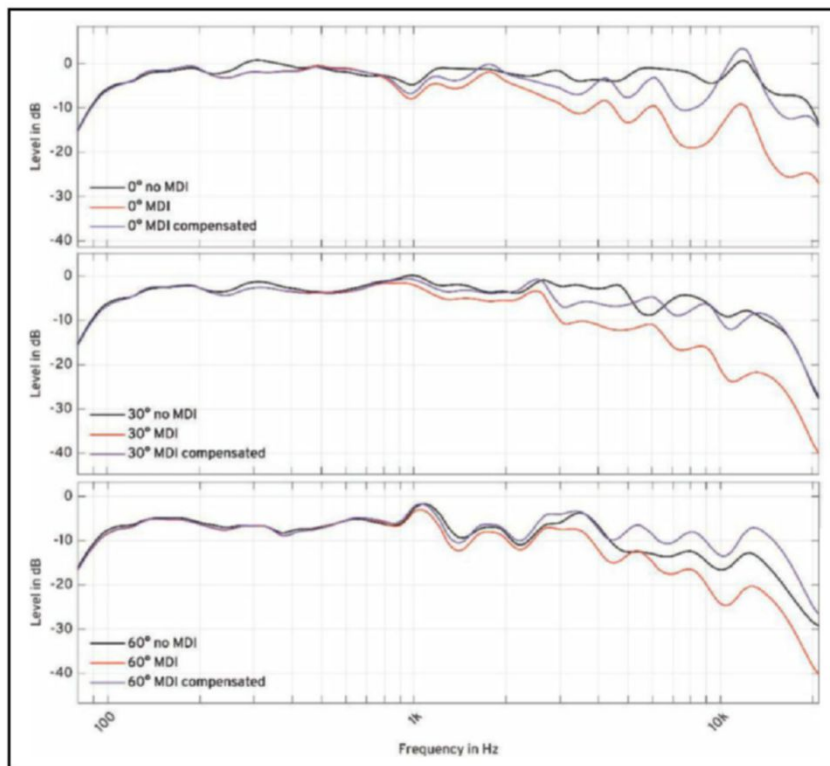


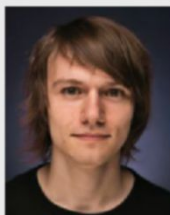
Figure 7: The transfer function of 120° optimized beam is shown at 0°, 30°, and 60° angles.

About the Authors

Adrián Lara Moreno is head of Research at Holoplot. Adrián joined the company in 2011 as a DSP Engineer, researching Matrix Array concepts together with Helmut Oellers, the founding father of Holoplot technology. Adrián played a pivotal role in creating Orion, Holoplot's first loudspeaker and proof-of-concept product. Adrián's progression to Head of Research has been instrumental in delivering X1 as well as driving Holoplot's further product development.



Christian Bellmann studied mechatronics at Dresden University of Technology with a focus on control engineering, power electronics, and electrical drives. In 2013 he received a Diploma degree for his thesis "Separation of direct sound and room reflections using holographic methods," which was supervised by Dresden Technical University and Klippel GmbH in Dresden. After graduation, he joined Klippel, where he is currently engaged in the research and development of loudspeaker measurement systems with a focus on acoustical holography.



different scenarios for different angles (0°, 30°, and 60°) at 1/3 octave resolution: a plain Holoplot MD96 audio module (black), the same module and beam configuration concealed with the MDI panel in front (red), and finally the Holoplot MD96 audio module concealed with the MDI panel and spatially compensated using Holoplot algorithms.

Figure 7 clearly illustrates the different spectral variations at different angles, which can only be solved with individual equalization. Holoplot's spatial compensation for the acoustic panel was implemented to restore overall spectral balance to the desired frequency response. Isolated local artifacts or spectral coloration resulting from panel resonances or reflections were not part of the correction, as compensating for these has proven ineffective.

Conclusion

Audio systems can be concealed in a variety of ways. Materials, such as cloth or perforated screens that are acoustically transparent, let sound pass through with minimal acoustic performance loss and can be effective in specific settings.

None of these solutions were feasible for the Illuminarium project as they cause visible distortions in the projection mapping of video content. To overcome this, Holoplot partnered with Canadian company MDI Strong, which developed a high-resolution video projection solution using a micro-perforated carbon fiber panel. This proved highly effective in providing a seamless projection surface, but introduced some disadvantages for the sound system concealed behind it, namely angle-dependent transfer function variations.

Klippel's NFS System has proven an effective and robust way to capture any loudspeaker's directivity balloon, including those concealed behind a panel. Capturing the three-dimensional behavior of the drivers mounted in the X1 audio modules and when deployed behind the MDI panel gave Holoplot the data required to implement the spatial spectral balance correction.

The spectral balance correction counteracts the level differences between different angles in 3D space for the same frequency across the audio spectrum. This newly developed feature for the Illuminarium project increased the spectral homogeneity of the deployed audio beams considerably and presents a clear advantage over traditional compensation methods used otherwise.

The result of this study is a concealing method for Holoplot X1 Matrix Arrays that delivers a clear, full-range sound with improved spectral homogeneity and a completely unobstructed visual projection surface.



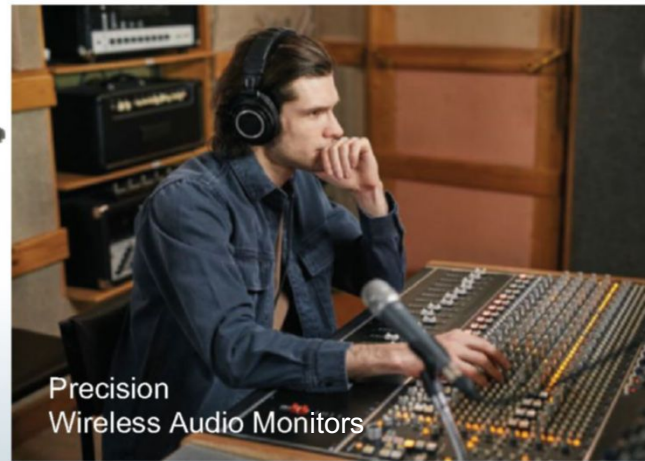
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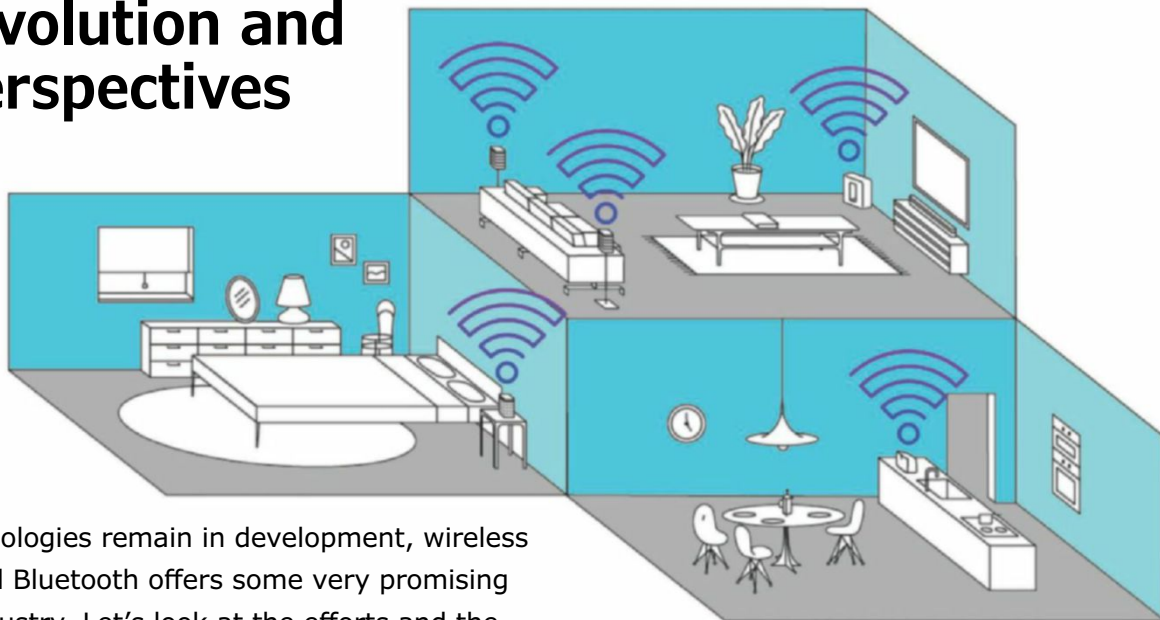
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Improving Wireless Audio

Part 2 — Evolution and Exciting Perspectives

By
J. Martins
(Editor-in-Chief)



Even though key technologies remain in development, wireless audio evolution beyond Bluetooth offers some very promising perspectives to the industry. Let's look at the efforts and the most promising platforms and technologies for wireless audio.

Wireless audio technologies remain a work-in-progress after several decades. As mentioned in the first part of this article, the lack of interoperability between proposed solutions, most using proprietary technology while sharing the available unlicensed 2.4GHz spectrum of the original Industrial, Scientific, and Medical (ISM) radio bands, have resulted in consumer confusion and lack of awareness of the full possibilities of wireless audio.

Consumers are able to understand some sort of replacement of wired connections they use today, but wireless audio in the digital age is a much more complex environment, with streaming services over the Internet as sources, signal distribution in the home using Local Area Network (LAN) standards, not to mention the lack of interoperability in multiroom synchronization.

With the overwhelming consumer awareness of Bluetooth, consumers understand the convenience of wireless, but it also created more confusion and distrust. Bluetooth is still considered too complicated and unreliable by most consumers, and only recently the key challenges for basic true wireless reproduction of stereo signals were sorted.

In this article, we are exploring alternative technologies to Bluetooth that pave the way for building future wireless audio standards. We also discuss different use cases that will never be covered by Bluetooth. And those remain largely tied to Wi-Fi standards, and radio frequency transmission in the same unlicensed bands using proprietary digital encoding and protocols.

Bluetooth Classic and the new Bluetooth LE Audio specifications are likely to remain a dominant technology for mobile applications

and noncritical applications for the foreseeable future. But the audio industry is very much aware that wireless audio needs to evolve beyond what was specified and what is possible over Bluetooth, and remains a challenge for the industry. After all, consumers are constantly being sold new experiences that challenge that technology. Take as an example immersive or spatial audio content that combines multichannel with synchronized metadata for object-based information, or gaming and entertainment applications in general that require very low latency.

Fortunately, there are new horizons in the use of other interesting technologies, such as Ultra Wideband (UWB), as well the convergence of applications with Wi-Fi and Bluetooth.

Wireless Home Audio

Starting with Wi-Fi (802.11), as discussed in the first part of this article, today we have AirPlay and Sonos as the two most recognized wireless solutions for the home audio market. We previously discussed AirPlay (1 and 2), and we will pass on discussing Google's Chromecast, which is a very specific effort to stream content to TVs or home audio systems that remains tied to hardware adapters and basically trails behind AirPlay or the SonosNet protocol, while its multiroom synchronized playback features clash with existing Sonos IP, with the two companies still involved in a legal dispute. And that's mainly why we will focus on Sonos for some additional perspective of wireless audio applications in the home.

We all got used to connecting our audio equipment with familiar cables and connectors, providing analog and digital connections.

And residential integration professionals are perfectly capable of offering all the features and convenience one might need with a structured cabled installation, also based on well-established and reliable standards, covering all use-cases, including multiroom audio with the highest quality digital high-resolution audio.

So, why is wireless audio for the home so important? The answer is audio is increasingly connected to the Internet through streaming services, the personal mobile systems we carry everywhere we go, and increasingly to the smart home. And one way or another, the smart home increasingly involves wireless interfaces and communications in order to function and evolve.

Also, because at the consumer level, wireless audio is the only possible way to move the audio industry in the direction of emerging standards such as immersive audio, in a practical and economic way for mainstream consumers.

What Exactly Is in Play?

Before Sonos, the only existing company with some degree of success for wireless audio was Apple, even if with a totally different vision. Apple gradually expanded the AirPlay ecosystem originated in 2004 from its iTunes software to a growing AirPlay ecosystem, built as a wireless connectivity protocol that started with audio streaming and evolved to include video and file transmission (including photos with its own metadata), eventually also allowing wireless screen sharing.

Sonos' approach was targeted from the start toward multipoint audio streaming, creating its own mesh network over Ethernet and Wi-Fi, and working better than those technologies could. The company designed speakers and perfected the technology as a distributed audio-centric system, allowing a central hub of connected audio sources to stream through a wired network and also wirelessly to multiple speakers around the home. And while Sonos was gradually building up its own home ecosystem, it also quickly built a strong portfolio of patents.

At the same time, multiple chip vendors and technology companies tried to follow suit, and improve on Sonos' own implementation. Companies such as Imagination Technologies with its CaskeID approach, were quickly followed by Qualcomm with its AllPlay solution, and DTS with Play-Fi, among others. As many large consumer electronics companies faced the need to match Sonos' features by licensing the technology, which Sonos kept exclusive, large corporations (e.g., Google and Amazon), also understood that it would be strategically important to offer the convenience of wireless home audio, and meet the growing demand.

Suddenly, multiple brands announced their own wireless audio streaming solutions, including multiroom features. But most licensing efforts from established technology companies have not evolved as expected, and the whole discussion about which technology brands would adopt, kind of "magically disappeared." And the main reason was that any implementation was sooner or later clashing with existing patents from existing players, but mostly from Sonos, which felt the need to take legal action as soon as it started to feel its market position menaced.

In May 2018, the Sonos lawsuit against Sound United, the parent company to Denon, Marantz, Polk Audio, Definitive Technology, HEOS,



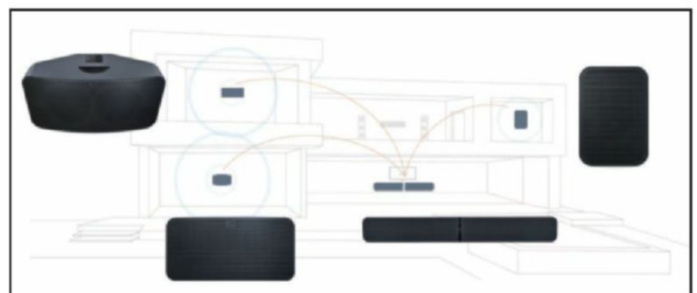
The Sonos family of multiroom audio products, connected over Wired Ethernet, wirelessly or a mixture of the two, creates a proprietary AES-encrypted peer-to-peer mesh network, known as SonosNet that supports multiroom synchronized audio. This was emulated in systems such as the Denon HEOS, leading to a patent infringement case that was settled in May 2018.

Boston Acoustics, and Classé Audio, reached a settlement, resolving all patent infringement litigation between the two companies, including a pending case inversely filed against Sonos by Denon. The terms of the agreement remained confidential and Sound United said that its Denon HEOS line of products would remain unchanged. The Sonos patent infringement case lasted for four years.

A more significant challenge to Sonos was created by Bluesound, part of Lenbrook International, the owner and manufacturer of prestigious brands, such as NAD Electronics, and PSB Speakers. Bluesound multiroom audio systems have a global presence and have expanded into (mostly wired) commercial audio applications with Bluesound Professional. Bluesound and NAD products all use the BluOS high-resolution distributed audio platform, which was also adopted by other leading premium audio brands (e.g., DALI and Monitor Audio).

On July 30, 2020, Sonos and Lenbrook Industries announced they reached a similar confidential settlement in Sonos' patent infringement lawsuit against Lenbrook Industries, and Lenbrook America. As part of this settlement, Lenbrook Industries agreed to pay royalties for a license to Sonos' patents for all BluOS-enabled devices.

"Lenbrook and the brands we own have been in the premium consumer audio space for almost 50 years. Our record of innovation and critical acclaim across all our brands speaks for itself. We have partnered with more than 20 innovative technology companies in our quest to offer the market the highest performance and most advanced audio solutions possible. We are pleased that we have



Bluesound approached the same multiroom streaming audio opportunity but focused on high-resolution audio and its own BluOS distributed audio platform, currently supporting AirPlay 2 for wireless multiroom.



We now also better understand what happened with these recent settlements in court, which Sonos has been fighting with determination. Since January 2020, Sonos has battled in court against Google, which is accused of infringing on five of its wireless audio patents. Sonos is seeking financial damages for infringement of its core wireless audio patents but, being a much smaller company, Sonos knows that Google is going to be a different challenge from previous lawsuits. Still, in January 2022, Sonos won the lawsuit and Google had to remove certain features from its devices. Predictably, Google is already countersuing Sonos for any reason they could think off, including mesh networking and echo cancellation technology.

Wireless Multichannel

In these disputes it's important to bear in mind that we are talking about essential audio streaming synchronization mechanisms for multiple device networks, crucial for multiroom audio systems, in particular for wireless. That is what Sonos systems do well.

The boundaries that all companies, including Apple and Sonos, have to work with for these applications are the Wi-Fi wireless network protocols, based on the IEEE 802.11 family of standards. It makes sense to comply to those standards, given that they create the environment—the infrastructure, if we can call it that—for those applications to run. And with the evolution of speed, bandwidth, and frequency bands being introduced by the Wi-Fi Alliance, currently promoting the latest Wi-Fi 6E standard, the successor of 802.11ac (now Wi-Fi 5) that uses 2.4GHz, 5GHz, and 6GHz, it becomes obvious that the possibilities for home audio look even more promising. What we will likely never see is a true “Wi-Fi Audio” standard in such a competitive and already disputed space.

This was made obvious when Sonos got its IP settlements in court, and when Apple announced AirPlay 2, implementing multiroom and stereo pairing for left and right speakers via Wi-Fi, with obvious latency issues. The Wi-Fi benefits are obviously the higher quality of streaming audio, and much wider wireless range, including through walls. Great for simple multiroom transmission, even with stereo signals. Not so great for surround multichannel and immersive formats with additional channels and object-based metadata.

Bluesound and its continuously expanding BluOS ecosystem builds on Ethernet and Wi-Fi to connect more than one enabled device and play music in perfect sync, or different music simultaneously, anywhere in the house, wirelessly. Using advanced technologies enabled by its own source and receiver products equipped with advanced Wi-Fi chipsets, BluOS can wirelessly transmit lossless music, with relatively lower latency. For compatibility and convenience for the user, most Bluesound or BluOS devices also support AirPlay 2 and even Bluetooth, including aptX HD, because the consumer demands the choice.

BluOS devices can also connect directly to streaming music services (e.g., Spotify Connect, Tidal Connect, Roon Ready integrations) while being remotely controlled directly from the BluOS Controller app. In this regard, Bluesound offers everything that the Sonos ecosystem offers, but with up to 24-bit/192kHz stereo streaming, a much larger set of technology options and compatibility.

Recently, Bluesound, like Sonos and all the other companies in this space, have faced the need to integrate Dolby Atmos immersive

WiSA SoundSend is an HDMI audio transmitter that enables WiSA Certified speakers to instantly connect to Smart TVs and serves as a central device for seamless management of immersive home cinema media systems. Transmitting high-quality 24-bit/96kHz wireless audio for up to eight channels, SoundSend offers advanced tuning features and decoding capabilities for Dolby Digital, Dolby Digital+, Dolby TrueHD, and Dolby Atmos.

been able to reach an agreement with Sonos to end this patent dispute,” explained Greg Stidsen, Lenbrook’s Chief Technology Officer.

So far, Sonos’ legal efforts seem to be relatively successful given the reinforced scope and width of its patents. But these settlements also have a bittersweet aspect for Sonos, since it effectively awards its competitors the possibility to use the technology—something which of course they always pursued and Sonos never allowed.

“Sonos is a company founded on innovation—it is a part of our DNA. Nearly 20 years ago, we invented multiroom wireless streaming, and since then we have developed many of the innovations that enable today’s listening experiences and will also enable tomorrow’s,” said Eddie Lazarus, Sonos Chief Legal Officer. “We welcome and encourage competition, and want to make sure that all companies entering this space recognize the strength of our IP and provide appropriate compensation.”



In 2022, WiSA Technologies confirmed that the new Monaco 5.1.2 speaker system (\$1,499) from Platin Audio would be the first fully immersive, multichannel, separates-based home theater system on the market to be WiSA HT Certified and Dolby Atmos enabled.

audio support in its ecosystem. With an update for BluOS (since 3.16) Bluesound introduced Dolby Atmos support for its Pulse Soundbar+ and BluOS-Enabled products, including all-new audio settings for home theater sound.

The Bluesound Pulse Soundbar+ is a high-quality design, configurable to a range of home entertainment applications and needs and can be combined with the Pulse SUB+ subwoofer. It reproduces all the Dolby Atmos channels and objects using virtualization profiles to create an enveloping soundstage. This is not much different than what other companies are doing with soundbars, with the challenge being connecting other speakers, restricted to two wireless rear satellite speakers. To sort the challenge, Bluesound requires combining its Pulse Soundbar+ and a Pownode streamer to connect the rear speakers. This is because wireless multichannel, even of basic surround 5.1 remains a challenge over Wi-Fi.

And this is where the Wireless Speaker and Audio (WiSA) Association comes into play, combining an essential part of the technology puzzle to enable wireless multichannel audio in the home.

WiSA Multichannel

Obviously, home theater systems are a natural application for wireless that even integrators increasingly embrace. Not many home entertainment technologies are designed and promoted with a focus on installation simplicity and ease of configuration such as WiSA.

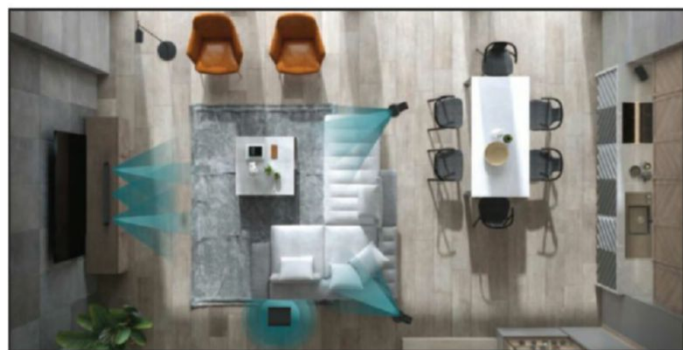
The organization originally created a strong momentum with its consumer electronics consortium with dozens of important brands involved. But in fact WiSA was developed based on technology owned by Summit Semiconductor, later called Summit Wireless Technologies, and more recently renamed simply WiSA Technologies.

The technology itself is until now the sole solution directly focused on wireless audio multichannel transmission for a home theater/cinema experience. All channels are transmitted in high-resolution, 24-bit/48kHz or 96kHz, with a unique signal latency of just 5.2ms and 1μs channel synchronization.

The reason for this superior performance is a combination of dedicated hardware and transmission mechanism working on the more reliable U-NII 5.1GHz to 5.8GHz spectrum, using dynamic frequency selection (DFS) procedures on channels providing great noise immunity for home use, away from Wi-Fi traffic. This allows a robust expansion to immersive configurations by supporting up to 12 speakers connected simultaneously (adding height speakers to the standard surround configurations).

Following the direct integration of the technology in the latest generations of LG Electronics TVs since 2019, and with growing support from basically all the large TV manufacturers—an increasingly important audio signal source in home entertainment, replacing set-top boxes—the technology became even more attractive as an interoperability standard for all leading brands and manufacturers to deliver wireless immersive sound.

Apart from LG, large TV and OEM/ODM manufacturers currently include Hisense, Skyworth Group, Compal (Toshiba), Foxconn, and TCL. Not only WiSA allows multiple components from more than 60 leading consumer electronics brands to be combined and ensure robust multichannel wireless audio, it also eliminates the complicated setup of traditional audio systems.



WiSA's Discovery wireless audio module supports up to four separate wireless audio channels, ideal for 2.1 wireless sound expansion for TVs, or stereo to 3.1 home theater configurations including soundbar wireless surrounds and subwoofer applications.

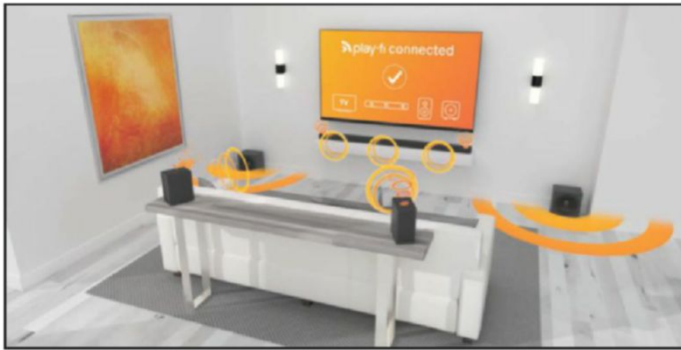
On the speaker side, WiSA Certified speakers are available from well-known brands, such as Bang & Olufsen, Harman, Klipsch, System Audio, Buchardt Audio, Piega, and many more. No other wireless audio technology currently on the market allows this level of performance and convenience for home theater environments.

But WiSA Technologies has recently embarked on an ambitious expansion of its market approach. With the vastly improving processing capabilities embedded in new smart TVs and wireless audio modules, and the evolution of market trends, WiSA Technologies has recently focused on a simpler approach for the growing soundbar market. This is essentially the result of pressures from the large TV manufacturers, OEMs and ODMs, and large audio companies to offer a more convenient wireless audio solution working on 2.4GHz for simple five channel and 3.1 configurations (soundbar, two surround speakers and subwoofer). This more affordable alternative is now designated as WiSA Discrete System Technology (or WiSA DS) and was developed in partnership with Espressif Systems and Realtek.

While the traditional WiSA solution that works in the U-NII 5GHz spectrum is now called WiSA Home Theater Technology (WiSA HT), and is mainly focused on immersive home cinema audio systems and namely Dolby Atmos support to be set up and connected quickly and easily, WiSA DS offers a simpler alternative using standard Wi-Fi bands in order to be cheaper to implement. For now, WiSA Technologies created the new DS 2.4GHz multichannel wireless audio module that transmits four full-range uncompressed 16-bit/48kHz audio channels and one high-fidelity subwoofer channel.



The original Phorus PS-1 speaker signaled the effort to develop audio streaming working over industry-standard Wi-Fi. Phorus was acquired by DTS in 2012 and resulted in what is now DTS Play-Fi.



In 2021, Xperi announced the introduction of DTS Play-Fi Home Theater, adding wireless surround sound capabilities to TVs without incurring additional hardware costs for manufacturers or consumers, using just the TV's built-in Wi-Fi.

Ideal for soundbar systems, the Discovery transceiver delivers the performance and affordability that those entry-level consumers and the manufacturers' bottom lines demand. To do this, WiSA's DS technology was designed to not switch channels once a link is established, and maintain the highest quality audio even in the most congested wireless environments.

The major trade-off is the slightly higher signal latency of 40ms (30ms transport latency), and 20μs channel synchronization.

In recent weeks, WiSA Technologies also announced that it will soon expand its approach to a software-based solution, with a low-cost licensing program, allowing WiSA's audio technology to run on readily available industry platforms, such as low-cost HDTV SoCs and Wi-Fi chips. This would have the potential to make WiSA adoption explode, but details of the implementation are still scarce and will be confirmed in 2023.

Play-Fi Native Sound

There are still other dedicated alternative technologies for wireless audio transmission, most of which are extremely robust and high-quality, including from point to multipoint. But most require dedicated hardware support on both the transmission and receiver(s) side, and most only support low latency mono and stereo. No synchronized multichannel transmission, and no network protocol to allow multiroom or multizone configurations with independent signals to be transmitted.

That's why the obvious solution is to use Wi-Fi, which is already a local area networking technology with Internet access. In the early



2022 represented a key year for wireless connectivity innovation as Wi-Fi 6, 6E, and Ultra Wideband (UWB) adoption accelerated across a growing number of end markets.

days, some companies explored the path to make wireless audio multichannel with the highest possible quality to work over Wi-Fi. But all of them struggled with the implementation or the entry barriers due to the patents already in place, as previously discussed.

One company did have that vision and invested in a technology that would deliver, resulting in what is now DTS Play-Fi. In fact, the technology was developed by a company called Phorus, which was acquired by DTS in July of 2012. DTS Play-Fi was launched that same year with the release of the Phorus PS-1 speaker and the PR-1 receiver. That company was founded by Dannie Lau, now General Manager of the Networked Audio Division at Xperi Corp. (Xperi acquired DTS in December 2016).

Of course, in the early days, DTS Play-Fi struggled with the basic streaming of compressed files over Wi-Fi, which Sonos and Apple AirPlay already were doing. But the development progressed over the years on multiple fronts, while rivals discontinued their similar efforts. Eventually, DTS Play-Fi could stream to multiple speakers over standard Wi-Fi, as Sonos could do, but without the proprietary mesh.

Play-Fi was also promoted as using shared metadata according to the Digital Living Network Alliance (DLNA) specifications, and was supposed to support lossless audio streaming from multiple devices, including Windows PCs over standard Wi-Fi. Play-Fi source devices could stream audio directly to multiple speakers, and support FLAC up to 24-bit/192kHz. Effectively, Play-Fi down-sampled the audio to CD quality (16-bit/48kHz) for wireless transmission to allow for a more robust experience in crowded Wi-Fi environments. High Resolution Audio, 24-bit/192 kHz streaming was implemented in what is called Critical Listening mode, with no down-sampling or network synchronization adjustment. Still today, DTS Play-Fi recommends the use of a wired LAN connection for Critical Listening mode, which says a lot about the technology challenges.

But thanks to DTS's and Xperi's persistence, DTS Play-Fi continued to evolve, gradually building an ecosystem of hundreds of compatible products from more than 30 brands. In market share, DTS Play-Fi struggled for some time to challenge rival Sonos on wireless audio streaming and multiroom. In 2019, DTS announced the introduction of AirPlay 2 in Play-Fi products from McIntosh, and Arcam (Harman), which also reveals a lot about the pressures that were being placed over the effort.

In 2021, DTS finally announced revitalized efforts for its Play-Fi wireless audio platform, and directly challenged the Wireless Speaker and Audio Association (WiSA) approach in multichannel applications with the introduction of DTS Play-Fi Home Theater. Xperi anticipated that televisions featuring DTS Play-Fi Home Theater running on Android TV and embedded Linux without requiring dedicated hardware would expand the existing ecosystem of Play-Fi soundbars, speakers, and amplifiers in the market.

This enables consumers to complete the "surround sound" from TV, and add up to two Play-Fi enabled satellite speakers and one subwoofer to improve the impact of the system, or go up to a 5.1 surround system, all easily configured through the app. And the TV manages all the decoding and the channel configuration. A movie with a soundtrack in Dolby Atmos or DTS:X will be automatically rendered to the number of channels and speaker layout in place.

Also, on the latest 2022 Philips TVs, the built-in speakers can be used to create a center channel, expanded with four external full-range and one subwoofer, with the lowest latency. “DTS Play-Fi automatically adapts to the number of speakers connected to the TV giving users a modular way to meet the needs of their room or a path to upgrade over time as they purchase additional components,” the company explains.

At IFA 2022, DST demonstrated a wireless 7.2.4 system (DTS struggles to call it Dolby Atmos...) system, on par with 24-bit/192kHz hi-res streaming, gapless playback. The company also promoted all new Play-Fi TVs from Philips and Vestel, which used Novatek’s SoC solutions, and the first Play-Fi soundbars from TCL and Hisense. All those new products and more will be demonstrated at CES 2023, where DTS plans to promote a big push to its wireless audio technology.

For this article, *audioXpress* had the chance to hear directly from Dannie Lau, who summarized the advantages of Play-Fi in the current market. First, he underlined the interoperability of the whole Play-Fi ecosystem, where all Play-Fi products can communicate and talk to each other and be used together in groups, across different product categories—speakers, TVs, soundbars, AVRs and streamers—and across different brands. Plus Play-Fi-enabled products can support all voice assistants and even support AirPlay or Google Cast as well, because it all works over standard Wi-Fi.

Second, it offers the support for high-res audio up to 24-bit/192kHz and support for direct streaming from lossless hi-res audio streaming from Amazon or Qobuz.

Third, DST Play-Fi Home Theater has an extremely low latency and multiroom audio synchronization below one millisecond. This enables, as an example, having multiple TVs across the house all with audio playing from external speakers with no lip sync issues. All this under standard Wi-Fi. No proprietary meshes, no proprietary radios, or dedicated chips required. The technology works over the Wi-Fi radios that manufacturers are adding to audio products and TVs today. In fact, as we’ve learned, manufacturers even have uploaded the firmware update to existing models over-the-air (OTA).

Another focus is the Play-Fi app that defines and controls all the available features, no matter the protocol, and continuous support on Windows PCs, since Microsoft doesn’t have a streaming audio technology like AirPlay. Play-Fi is the only solution of this type to incorporate Windows platforms.

In a nutshell, DTS Play-Fi is aiming to become the first truly “Wi-Fi native” wireless audio solution in the market, integrating any source of type of audio products and application. The same technology and the same app can control surround sound home theater, multiroom audio, and high quality, high-resolution stereo listening. A complete ecosystem.

And as Wi-Fi gets better, this technology will take advantage of it. As Lau stated, “I think in the early days of Play-Fi there was a reason why there weren’t other people trying to do the same thing at that time. Wi-Fi wasn’t very good yet. Our protocol was designed to be very robust in non-optimal conditions. And I think that’s benefited us, as Wi-Fi has gotten better. As we’re talking to different CE companies about doing wireless surround sound now, their first reaction was “that’s not possible for Wi-Fi. It’s not



Yes, Wi-Fi could eventually be optimized for low-power, but the problem of antenna size remains. Ubilite, a San-Diego, CA-based semiconductor company, launched its UBI206 platform and SoC, targeted specifically for ultra-low power wireless applications. The UBI206 SoC enables ultra-low power Wi-Fi (802.11b/g/n) audio streaming solutions up to 72Mbps with a 10-fold improvement in power consumption.

going to work. It’s going to have tons of dropouts, while you’re streaming YouTube, it’s going to cut out.” We’ve proven to them it does work. We do 4K streaming with Play-Fi all day and there’s no problem. So, a lot of that is a testament to the early design work we had around the protocol and the optimization around bad Wi-Fi networks. And now it’s paying its dividends. As things improve

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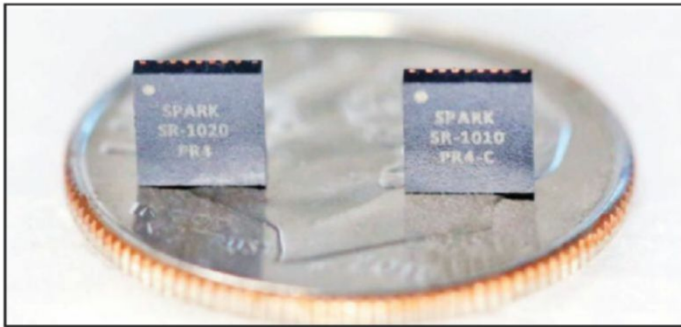
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even further with Wi-Fi 6, you'll see even more capabilities added to the system. And that's because our efficiencies will go beyond what other competitors will be able to do."

"In terms of channel count, it doesn't matter if it's Dolby Atmos or DTS:X IMAX Enhanced, we're going to support whatever type of experiences that other people are defining around higher channel counts," he also adds.

Of course, we also asked Lau about the patent issues that other companies have faced and his reply was confident. "We started way back in 2010 when most people weren't even using Wi-Fi yet. So we've developed a lot of different technologies and we patented these technologies that make our things work over Wi-Fi. We're very confident in the patents we have, and the fundamentals behind what we own, and that's something that we provide to our users because we did it first."

Going Forward

As we discuss what is available in wireless audio for the home, it becomes clear that Wi-Fi is going to be evolving and that AirPlay,



The current SPARK SR1000 UWB IC family takes full advantage of UWB technology to simultaneously deliver industry-leading energy efficiency, latency and bandwidth, with tested data rates of up to 10Mbps @1.5 nJ/bit energy efficiency, 2mW power consumption while transmitting and receiving 1Mbps (scaling to 6uW at 1kpbs), ultra-short wireless latency (50µs for 1kb), and time-of-flight positioning at 30cm accuracy.

BluOS, Sonos, the different WiSA approaches, and obviously DTS Play-Fi are all going to be strong contenders, and that we will see significant announcements on all those fronts going forward.

Meanwhile, as discussed in the first part of this article, Bluetooth LE Audio will start making an appearance in actual products—CES 2023 will be a decisive moment, when we will have the first glimpse of the new products supporting LE Audio and what those will be able to offer, including some supporting audio sharing and broadcast through Auracast. And this will determine, largely, what people will associate with wireless audio on the move, outside the home, and inside it.

Regardless of the quality issues, Bluetooth "continuity" will be an essential of the convenience that consumers will expect to have anywhere. No consumer electronics manufacturer will be able to stay away from supporting Bluetooth control and connectivity for streaming applications—as Apple painfully learned in its first HomePod.

Speaking of Apple, we can be certain that the numerous Apple audio development teams are working hard on these very same questions—how to combine the best wireless audio quality and experiences that we are able to have at home, with the convenience of mobile and the ubiquity of Bluetooth. And since Apple controls its full ecosystem of hardware and software, and holds a leading seat in virtually all major technology interest groups and consortiums, its engineers are in a unique position to anticipate what will be possible.

In fact, Apple has started paving the way with efforts that so far we do not associate with audio applications, when its products combine Near Field Communication (NFC), Ultra Wideband (UWB), GPS, Bluetooth, and cellular networks radios.

Since the launch of the Apple iPhone 11 line, which integrated its Apple U1 UWB short-range radio chip, Apple enabled a lot of interesting proximity applications with AirDrop, time-of-flight ranging and network position, and ultimately enabling users to interact with connected accessories through "Nearby Interaction" features.

And this is not done through any type of closed approach. In fact, Apple recommends that chipset vendors interested in enabling Apple UWB interoperability with its UWB solutions should become a member of the FIRA Consortium (www.firaconsortium.org), which is the organization where these specifications are being openly defined.

Apple is also a sponsor member of the NFC Forum and sits on its board, implementing NFC-based systems that are often combined with Bluetooth and UWB to extend usability.

Recently, The NFC Forum unveiled a new NFC Device Class specification that provides device manufacturers and OEMs with a miniature, low-cost, sustainable solution for wireless wearable devices. The Card Emulation (CE) Device class specification defines an emulation of a contactless card, providing another example of a framework connectivity for payment or access applications (digital keys), which can be extended in range with other radio technologies, such as UWB.

These are great examples of how the audio industry could approach an interoperable solution for high-quality audio wireless transmission in and out of home, helping to finally establish wireless audio to everyone's content.

UWB promises to be one of the key solutions to create the bridge between the range and bandwidth limitations for audio, paving the way to optimize for different applications, such as ultra-low latency, multichannel, and spatial formats.

UWB radio technology is based on the IEEE 802.15.4a and 802.15.4z standards that can enable a more precise measure of the Time of Flight of the radio signal, leading to centimeter accuracy distance/location measurement. In addition to this unique capability, UWB offers data communication capabilities up to 27Mbps, while using extremely little energy. Also relevant, it offers exciting possibilities for wireless mesh networking and MIMO/beamforming, which could be applied toward wide-area network applications, expanding beyond Wi-Fi.

Qorvo, a semiconductor company that offers MFi-certified interoperable UWB solutions that already work with Apple's Nearby Interaction accessory protocol, is excited about future possibilities, given that the technology uses a more secure physical layer, offers lower latency, supports lower power, and has a longer range/coverage than all existing wireless network technologies.

It is also important to detail that UWB technology has been successfully deployed in experimental applications for distributing large numbers of HDTV channels across digital terrestrial networks, and even recently field tested for multichannel wireless microphone systems in professional audio for live events.

In 2016, Audio-Technica created Alteros, a new subsidiary solely

focused on the development of products capitalizing on years of extensive UWB and RF technology research. Alteros was led by Jackie Green, as president and CTO, together with Brian Fair and Bob Green in product engineering roles. The expectation was that the use of UWB would enable addressing the growing UHF and VHF frequency spectrum challenges. In 2017, following seven years of dedicated research and development and field testing in actual live events, Alteros launched the Alteros GTX 6.5 GHz UWB digital wireless microphone system. Apparently, even though it worked, the benefits of the technology were not enough to challenge the established competition in this field by companies such as Sennheiser, Shure, and Sony, which had migrated successfully to digital transmission, were able to accommodate larger numbers of channels and at a lower price. In October 2019, Audio-Technica announced the closing of Alteros "due to corporate strategic decisions."

Jackie Green, who went on to found Nexonic, a consulting firm in field of RF and digital audio transmission, is now supporting UWB efforts as Technical Advisor for the UWB Alliance. And precisely in 2020, the UWB Alliance and the FiRa Consortium, the two existing supporting organizations, established a joint liaison agreement to strengthen the adoption of UWB. As a part of the new liaison, the UWB Alliance agreed to focus on the promotion and safeguarding of UWB technology and advancing updates of the UWB rules, while the FiRa Consortium would focus on UWB use cases specification, IEEE 802.15.4z interoperability and certification.

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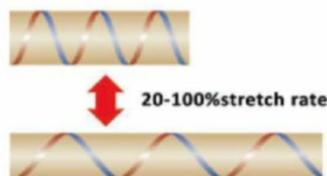
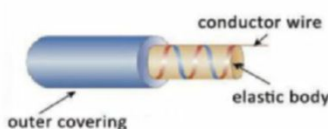
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UWB is also a key technology on digital car key applications for the automotive industry and is ready to boost adoption of smart home technology. That's the reason why all the largest connectivity chipset companies are now pursuing new UWB platforms, frequently combined in triple- or quad-connectivity modules.

For product developers looking at long-term projects for devices, such as true wireless earbuds, using UWB as a bidirectional link with a smartphone in combination with Bluetooth Low Energy, is a very attractive possibility.

SPARK Microsystems

In 2021, SPARK Microsystems, a Canadian fabless semiconductor company specializing in next-generation UWB applications for consumer and connected devices, joined the UWB Alliance and the FiRa Consortium to accelerate the development and adoption of the technology. Anticipating the low power, low latency, and 10x more data throughput than Bluetooth for short range audio applications, SPARK Microsystems is the first company to fully embrace the creation of platforms for audio applications based on UWB.

The SPARK Microsystems series of low-power UWB wireless transceiver ICs enable short-range wireless connectivity applications that were, until now, impossible due to power and latency requirements. Targeting gaming peripherals, AR/VR headsets, smart home devices, and battery-less sensors for connected devices, SPARK introduced solutions with an order of magnitude better energy efficiency, latency, and range than Bluetooth Low Energy or ZigBee, working at faster data rates. When paired with energy harvesting technologies, SPARK's UWB technology significantly extends the battery life of electronics and enables the battery-less operation of wireless devices such as sensors.

But more important, SPARK specifically promoted UWB technology for high-quality audio applications. The company conducted tests

that showed superior data throughput and enhanced directional audio, enabling situational awareness, all of which are key for UWB applications in gaming peripherals and personal, mobile audio.

SPARK already offers a unique Personal Area Network (PAN) wireless transceiver that achieves 40x more energy efficiency, 60x lower latency, and 10x more data throughput as compared to Bluetooth LE. In addition, the patented SPARK UWB radio does not interfere with other narrow band radios such as Wi-Fi, BLE, Zigbee, Z-Wave, and cellular bands.

SPARK also developed a complete evaluation solution handling 10Mbps for uncompressed, CD-quality audio, with support for multichannel streaming and a 5ms overall audio latency.

SPARK Radio can also support device-to-device, star, and mesh network configurations. These features allow for increased connectivity and reliability, as well as better coverage of large areas. And SPARK Radio can multiplex several users/devices in the same space (audio channels, controllers, keyboard, mouse, etc.).

The company offers a Spark Audio Demo Kit, powered by an ARM Cortex-M4 MCU, which supports transmission of uncompressed audio in a range of up to 50 meters, together with a software development kit (SDK) with code examples and APIs for its SR1000 Series UWB transceivers.

Several audio manufacturers are already testing these development solutions from SPARK with promising results. Over the coming years we can expect to see the results becoming visible, although the biggest sense of urgency seems to come from the true wireless and hearables applications, where there are more cooperative efforts happening. CES 2023 should allow for some of those companies to make those efforts visible and present their expected roadmaps.

As stated earlier, wireless audio is still an area of work-in-progress with very exciting possibilities. 

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Using Ultra-Wideband Technology

Wireless Delivery of Audio and Multimedia



For tomorrow's audio and AR/VR devices, Ultra-Wideband (UWB) technology provides high throughput, low latency, and low power connectivity for body and personal area networks—even in congested environments. UWB complements Bluetooth and Wi-Fi with high-performance and mobile-friendly short-range wireless connectivity.

By

**Frederic Nabki, Nicolas Paillusseau,
and Ronald Rinkenbach**

(SPARK Microsystems)

For the short-range wireless delivery of audio and multimedia, the performance capabilities of Bluetooth/Bluetooth Low Energy (BLE) and Wi-Fi are well understood to product designers and consumers alike. Bluetooth/BLE provides mobile-friendly power efficiency, but with extremely low data rates (typically < 1Mbit/sec) and high latency compared to Wi-Fi. Bluetooth's low data rate forces compression to be used that increases audio latency and also reduces audio quality.

The Wi-Fi local area network communications standard, on the other hand, is well suited in many respects for high data throughput applications and supports a wider range/distance—but at a huge power penalty and long latency that makes Wi-Fi impractical for tomorrow's mobile apps.

The advent of ultra-wideband (UWB) technology presents a new and compelling short-range wireless option that blends the best of both worlds of Bluetooth and Wi-Fi (**Figure 1**). UWB achieves ultra-low power and ultra-low latency, with up to 10× higher data throughput than BLE. It's optimized for high-quality audio and emerging AR/VR applications serviced by smart glasses and other wireless

peripheral devices requiring higher data rates and lower latency.

UWB technology can provide more immediate benefits in the traditional audio domain, where it's applicable for the next generation of wireless headsets, speakers and more. UWB's promise to deliver "wired like" audio quality without the wires—there's no compression needed—is enormously enticing for anyone who laments the loss of audio quality we experienced decades ago when we largely jettisoned wired audio connectivity in favor of mobile-friendly wireless audio devices.

UWB's value proposition extends beyond the audio domain, however, and acknowledges the coming convergence of traditional multimedia delivery with high-speed human interface device (HID) control and positioning capabilities. The augmentative audio and video overlays delivered wirelessly to tomorrow's smart glasses and audio devices, for example, will change the way we engage/interact with our environments. These devices will depend on innovations such as UWB to harness and transmit data wirelessly among our personal, body-borne electronic devices.

All three technologies—Bluetooth/BLE, Wi-Fi and UWB—will play a valuable role for tomorrow’s short-range wireless device connectivity, and their interaction and interoperability must be made seamless going forward. This requires, among other things, minimal interference between them, and here again UWB shines in its ability to enhance wireless audio and mobile AR experiences in congested environments.

Body & Personal Area Networks

UWB-enhanced devices can deliver data throughput and low latency performance on par with wired links—without the wires—to enable a new class of body area network (BAN) and personal area network (PAN) applications. We can think of BANs/PANs as the personal zones that surround each individual in a connectivity mesh incorporating their wearable wireless devices, distributing data between them in ways that fully immerse the user’s sense of sound, sight and touch—this is the vision of fully-realized AR/VR.

Bluetooth-connected wireless earbuds, paired with our smartphones, are a simple example of a modern BAN/PAN application in action. For the AR/VR-optimized devices of tomorrow, RF signals transmitted within our BANs/PANs will carry audio data as well as video and positioning data between several devices at once, all of which must be precisely orchestrated at extremely high speed in dynamic user environments.

This requires extreme low latency to enable real-time comms and positioning synchronization between devices to ensure true AR/VR-caliber responsiveness. After all, the quality of the AR/VR experience hinges completely on the quality of the communications. Signal interruptions and desynchronization distract us from the experience and interfere with our immersion. This challenge is magnified considerably in congested user environments, where signal interference can become problematic.

UWB stands alone as the short-range wireless technology that can deliver the high data throughput and ultra-low latency required for the real-time immersive AR and metaverse experiences of tomorrow. And UWB achieves this without introducing additional interference for physically adjacent Bluetooth/BLE and/or Wi-Fi devices—and/or other UWB devices.

SPARK Microsystems takes the UWB value proposition another step further by ensuring that the power consumption profile for these wireless audio and AR/VR peripherals will allow for lengthy usage times between battery charges. Again, this

means fewer interruptions for more seamless immersion in the AR and metaverse environment.

These PAN/BAN-connected devices will become as ubiquitous and essential to our daily lives as our smartphones and earbuds are today, going with us wherever we go...for as long as their battery charges will last them. AR and the metaverse will be fundamentally mobile experiences, and we will be reliant on BAN/PAN-connected devices being charged and ready for use as we go about our day. Indeed, many consider the “holy grail” of fully realized AR smart glasses, wireless audio, and hardware peripherals to be the ability to effectively address the lingering power consumption challenge.

UWB Excels in Environments Congested by Existing BT/BLE and Wi-Fi Radios

There are several qualities unique to UWB that make it an ideal fit when it comes to enhancing wireless audio and AR/VR application user experience in congested environments.

Low RF Power—UWB delivers low spectral output power by default, required per regulatory standards. Most countries have now agreed on the maximum output power for UWB, defined as -41.3dBm/MHz, to avoid disturbing adjacent wireless communications.

UWB can coexist with other transmission standards and not interfere with them. It complements Bluetooth/BLE and Wi-Fi but does not compete with them or cause additional congestion. In fact, other radio systems will see UWB as background noise.

Low RF power naturally limits the range of UWB target applications to several feet/meters, thereby creating “personal zones” and reducing interference

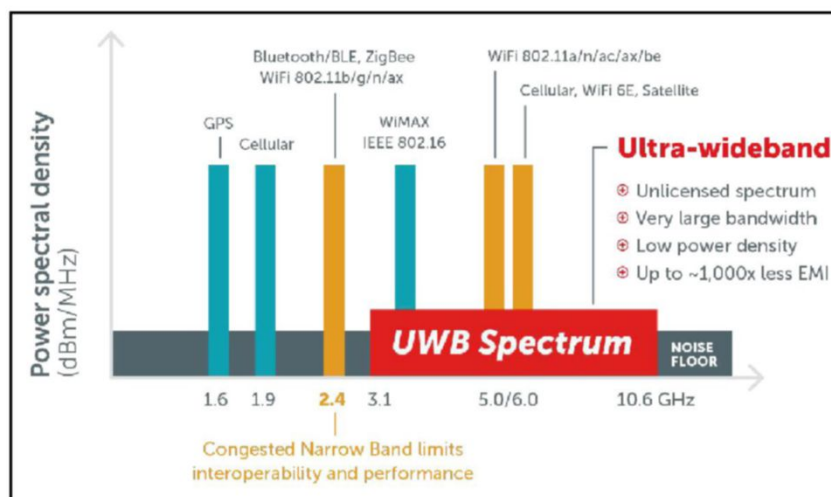


Figure 1: UWB transmits very low spectral output power and uses large portions of the RF spectrum, making it ideal for short range communications.

with nearby devices in congested environments. UWB range is limited to “near me” distances, making it ideal for indoor audio and AR/VR applications. This also serves to enhance the privacy of communication, confining it to a small “bubble.” Bluetooth and Wi-Fi’s sphere of influence on adjacent systems stretches far beyond the personal bubble required by most applications and therefore creates congestion with other users’ personal bubbles.

Low Power Consumption—The biggest difference between traditional narrowband RF systems and UWB is the underlying transmission method. UWB uses an impulse-style radio whereas the majority of narrowband radio systems use a modulated carrier frequency. Sending pulses rather than modulated sinewaves, SPARK’s implementation of UWB is considered a non-coherent impulse technology. With UWB, the underlying impulse radio technology means transmission is done power-efficiently in short bursts. Narrowband systems using a modulated carrier have to continually transmit the carrier signal during data transfer, requiring higher transmit power (**Figure 2**).

SPARK UWB uses proprietary technology to further lower the power consumption of the impulse radio. Combined with the low spectral output power benefit, UWB significantly reduces power consumption. By way of comparison, SPARK’s implementation of UWB consumes up to 40x less power than BLE.

This ultra-low power operation makes UWB extremely advantageous for small battery-powered applications, regardless of the data rate. Current consumption can be further optimized for battery operated devices including earbuds, headsets, microphones, portable speakers and more.

The combination of UWB’s wide frequency range, allowing for multiple channels, and the underlying impulse radio technology means that UWB is also less impacted by multipath phase distortion issues and can filter out unwanted interferers, making it more robust.

Co-Existence—UWB, Bluetooth/BLE, and Wi-Fi can readily coexist. Bluetooth occupies a different frequency band than UWB, so interference between them isn’t a pressing concern. But Bluetooth/BLE radio signals have a larger sphere of influence than required for BANs/PANs, and this can lead to congestion when many devices are in close proximity.

Wi-Fi connectivity is most beneficial for devices that are spaced apart (large homes, low density living areas, etc.). But the extended range becomes a disadvantage for live streaming applications for multiple users in high density areas where numerous Wi-Fi networks overlap (particularly in urban neighborhoods/cities).

These represent potential sources of interference—and they’re all competing for the available bandwidth in the area of interest.

In some applications, it will likely make sense to use more than one radio type. A hybrid RF solution greatly improves the overall end user experience and offers system manufacturers considerably more flexibility. Co-existence between Bluetooth/BLE, Wi-Fi and UWB makes this easier.

Latency—UWB excels at enabling low latency wireless links. Link latency in a UWB system can be very low (less than 5ms for most audio applications), making it easy to keep audio signals in sync with each other and with other related sources such as video. Additionally, due to low latency control signals, audio/multimedia channels can be tightly synchronized (within 20μs) to deliver exceptional performance across a BAN/PAN or home sound system.

Distance Measurement—UWB can also measure distance between two radios using time-of-flight calculations: An impulse is sent from one device to another, and we measure the time it took from transmit to receive. The distance between objects can be determined accordingly, and this can be measured with picosecond accuracy with UWB chips.

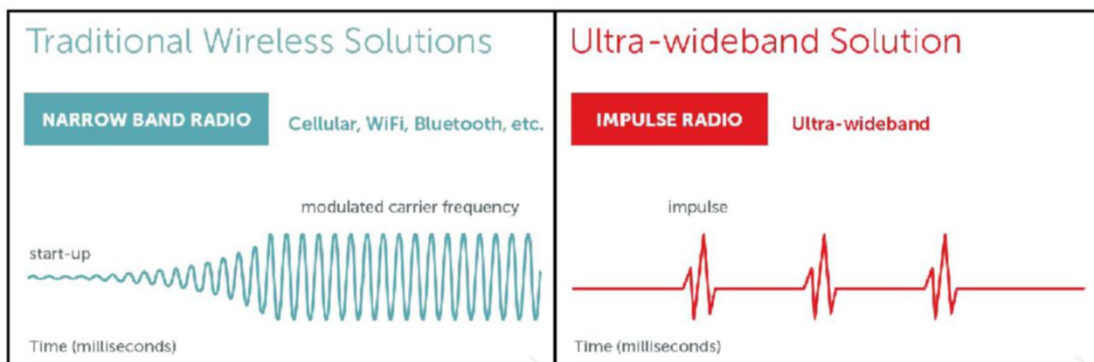
We see this capability in action today via UWB-based positioning devices like Apple’s AirTags, designed to help identify the precise location of tagged objects with accuracy that’s within mere centimeters. This same UWB capability can be used to determine distance between audio components and devices, enabling the calibration of user-friendly spatial audio configurations for improved listening experiences.

SPARK Microsystems is the first mover in leveraging UWB for data delivery, enabling the ability for high-performance wireless audio/multimedia transport. Other UWB technology vendors, as well as the Institute of Electrical and Electronics Engineers (IEEE), are currently focusing their efforts on the positioning and distance measurement attributes of UWB.

Flexibility—UWB links can also contain piggyback data for configuration and diagnostic purposes, and bi-directional links are also possible. What’s more, UWB wireless audio links can dynamically scale up and down and adjust link rates to optimize audio quality for a given environment.

UWB’s support for multiple channels ensures that multiple audio and AR/VR devices can be used on the same BAN/PAN. UWB can also support broadcast and mesh modes, in addition to point-to-point (direct connection) schemes.

Figure 2: Traditional narrowband RF systems and UWB operate on different underlying transmission methods. UWB uses an impulse style radio whereas the majority of narrowband radio systems use a modulated carrier frequency.



Unlicensed Spectrum—The UWB spectrum is unlicensed, so it's free for everyone to enjoy! Each technology—Bluetooth/BLE, Wi-Fi and UWB—brings compelling attributes to the table that should be carefully weighed as we take a long view of our evolving short-range wireless needs into the future. UWB provides distinct advantages that blend the best attributes: ultra-low power and latency with high data throughput.

In Conclusion

These robust connectivity and communication properties make UWB ideally suited for the next generation of wireless audio devices. As wireless audio/multimedia delivery converges with HID and positioning data delivery for future AR/VR devices, UWB will ensure higher overall performance that's optimized for mobile BANs/PANs.

Moreover, UWB provides significant benefits for high-quality, short-range wireless delivery in congested areas for smoother, seamless consumer audio and AR/VR experiences. For more information about UWB technology and SPARK Microsystems, please visit www.sparkmicro.com 

About the Authors

Dr. Frederic Nabki is a Co-Founder and the Chief Technology Officer of SPARK Microsystems where he directs the technological innovations that SPARK Microsystems is bringing to market. He has 17 years of experience in research and development of RFICs and MEMS. He obtained his Ph.D. in Electrical Engineering from McGill University in 2010. Dr. Nabki has contributed to setting the direction of the technological roadmap for start-up companies, coordinated the development of advanced technologies and participated in product development efforts. His technical expertise includes analog, RF, and mixed-signal integrated circuits and MEMS sensors and actuators. He has published several scientific publications, and he holds multiple patents on novel devices and technologies touching on microsystems and integrated circuits.



Nicolas Paillusseau joined SPARK Microsystems as Applications Team lead in November 2021, with more than 15 years of professional semiconductor experience. Prior to joining SPARK Microsystems, Nicolas was a Senior Application Engineer at STMicroelectronics where he was involved in the development of various products, including mobile phones SoC with ST Ericsson, then TV set-top box SoC, and finally, the application development of STM32 32-bit Arm Cortex MCUs. Nicolas started his career as Hardware engineer at TCL & Alcatel Mobile phones.

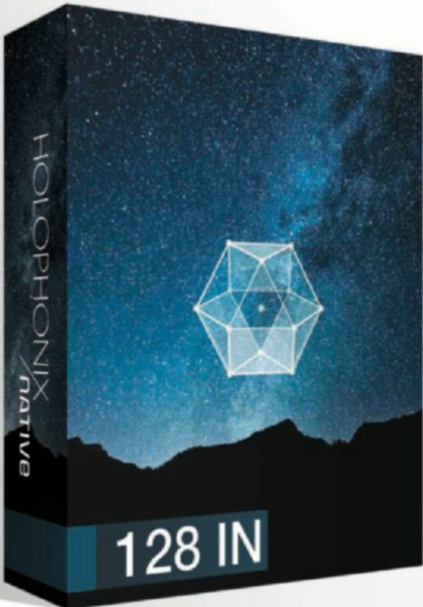


Ron Rinkenbach joined SPARK Microsystems as Senior Applications Specialist in April 2021. Ron is an engineering professional with background in electronic design and customer support roles working in industries such as Semiconductors, Avionics, Aerospace, and Communications. His previous roles include Field Sales Engineer at Axis Communications and Field Applications Engineer at Introspect Technology.



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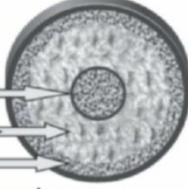


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An Approach to Designing RIAA Networks

This technical article addresses an issue that often keeps designers busy when calculating the RIAA parts' values for a phono amp. The reason is that the surrounding circuit parts always somehow influence the RIAA correction. So it is often a matter of trial and error. With the author's method, it is much easier to quickly calculate the correct values.

By
John Ellis

The well-known Recording Industry Association of America (RIAA) standard for vinyl recordings was pioneered by RCA and became widely adopted in the mid-1950s. It introduced three time constants of 3180 μ s, 318 μ s, and 75 μ s with the corresponding frequency break points at 50.049Hz, 500.49Hz, and 2122.11Hz respectively. **Figure 1** shows the ideal response with such a characteristic, ignoring the later 20Hz addition.

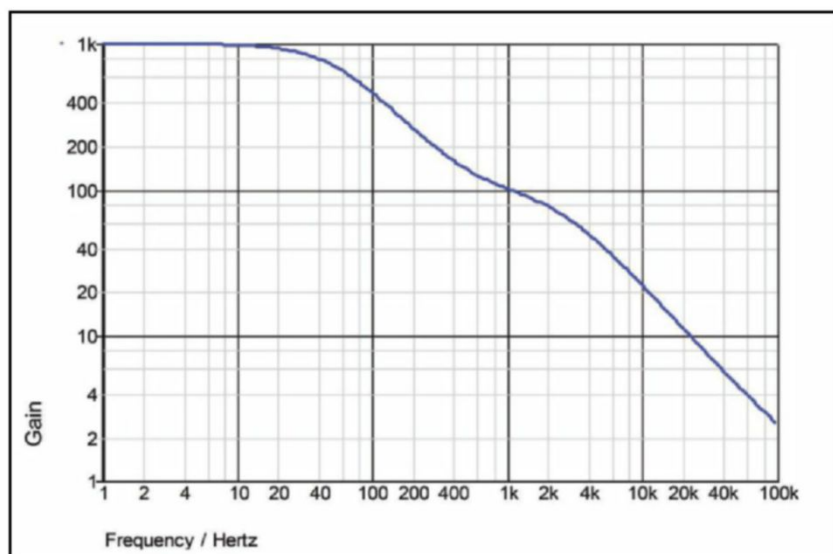


Figure 1: RIAA playback characteristics show the ideal response with such a characteristic, ignoring the later 20Hz addition.

In his noteworthy paper [1], Stanley Lipshitz drew attention to the fact that many designs were incorrect. He pointed out a common error where the time constants are applied without considering the whole network. To calculate the correct values of components using analytical methods quickly leads to one or more pages of equations, particularly when real amplifier gains and additional resistors may be required for stability. An alternative to analytical expressions is to use numerical simulation.

A Numerical Approach

One of the most widely used algorithms for solving simultaneous equations is to use Newton's (or the Newton-Raphson) method. For this method to apply to RIAA networks the problem is broken down as follows:

First, the RIAA characteristic is defined at several particular frequencies, which will be used to solve the network. **Figure 2** shows three such points, but a fourth, not shown, is located at 0.1Hz and is used essentially to solve the "DC" gain where the influence of the capacitors is negligible.

This gives a set of spot frequencies $sf(i)$, where $i=1...4$ but others could be proposed. This particular set selected represents the geometric mid point of each frequency band in the slope regions. Thus, $sf(1)=0.1$, $sf(2)=\sqrt{50.049 \times 500.49}$ and $sf(4)=\sqrt{2122.1 \times 21221}$, all in Hertz. Other

frequencies and gain targets can be used. The central spot frequency and the gains at all the specified frequencies are determined from an idealized RIAA characteristic defined by a transfer function with the three independent time constants:

$$TF_{\text{RIAA}(\omega)} = \frac{G_0(1 + j\omega\tau_2)}{(1 + j\omega\tau_1)(1 + j\omega\tau_3)} \tag{1}$$

where $\tau_1=3180\mu\text{s}$; $\tau_2=318\mu\text{s}$ and $\tau_3=75\mu\text{s}$ are the defining values, and G_0 is the maximum gain. Solving this for an attenuation of 0.1 (the mid-band) with $G_0=1$ yields a frequency of $\text{sf}(3)=1027.37\text{Hz}$, not the commonly assumed 1000Hz. The gains at the chosen frequencies are summarized in **Table 1**. For a real amplifier, the generalized feedback network is that shown in **Figure 3**.

For each frequency, the network impedance $z(i)$ is calculated. If used as the feedback loop around an operational, or discrete amplifier, the attenuation is:

$$\alpha = \frac{rg}{rg + z} \tag{2}$$

where rg is the “grounding” resistor in the feedback network as shown in Figure 3, and z is the complex impedance of the network. For Newton’s method, functions that equate to zero must be defined. Either the network gain or attenuation can be used. A comparison of the required gain with a target value for each frequency point as given in Table 1 can be defined using a transfer function for the open loop gain of the amplifier, A so that:

$$G = \frac{A}{1 + A\alpha} \tag{3}$$

where the closed loop gain G , open loop gain A , and attenuation are all complex values. If a transfer function for an actual amplifier is known that may be used too.

The set of equations required for Newton iteration is defined as:

$$f = \text{abs}(G) - g = 0 \tag{4}$$

where G is the achieved gain from Equation 3 and g the target gain as given in Table 1. Note that while the impedances are calculated using complex numbers the vectors f and g are real. A typical network is shown in **Figure 4**.

If **comp** represents the array of component values for both resistors and capacitors, that is: $r1, c1, r2$ and $c2$, (optionally with or without $R3$), then Newton iteration produces an error correction:

$$\text{comp} = \text{comp} - \Delta\text{comp} \tag{5}$$

An example of the Newton functions $f(i)$ for the spot frequencies $\text{sf}(i)$ is:

$$z(i) = \frac{1.0}{\frac{1.0}{r1} + \frac{1.0}{zc(1,i)}} + \frac{1.0}{\frac{1.0}{r2} + \frac{1.0}{zc(2,i)}} [+r3] \tag{6a}$$

$$f(i) = \text{abs}(G(i)) - g(i) \tag{6b}$$

where $z(i)$ and $zc(j,i)$ are the impedances of the network and capacitors at $\text{sf}(i)$.

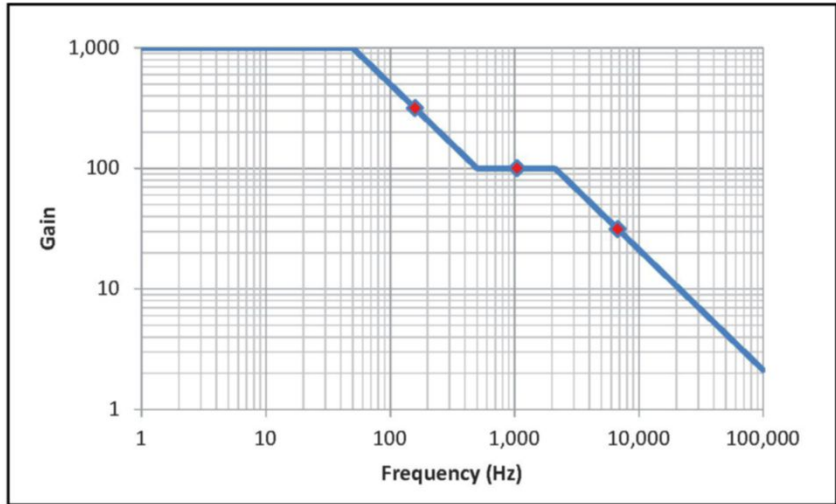


Figure 2: This graph shows the idealized RIAA response and matching points.

Reference Point	Frequency	Target Gain
1	0.10Hz	1000.00
2	158.27Hz	315.35
3	1027.37Hz	100.00
4	6710.56Hz	30.234

Table 1: The frequencies and their target gains are summarized with $G_0=1000$.

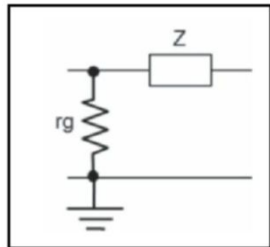


Figure 3: The schematic shows the generalized feedback network.

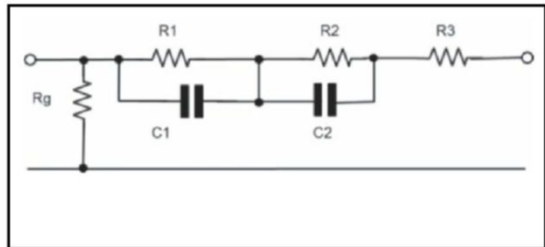


Figure 4: Here is an example network for a RIAA replay.

Component	Value
r1	92.152kΩ
r2	7.748kΩ
c1	34.51nF
c2	9.687nF

Table 2: The components for $rg=100$ and $r3=0$ are shown.

In the absence of r_3 , the Jacobian is calculated from the differentials of the four functions with the variations in component values. The sequence of entries in the Jacobian are:

$$J = \begin{bmatrix} \frac{df(1)}{dr1} & \frac{df(1)}{dc1} & \frac{df(1)}{dr2} & \frac{df(1)}{dc2} \\ \frac{df(2)}{dr1} & \dots & \dots & \dots \\ \frac{df(3)}{dr1} & \dots & \dots & \dots \\ \frac{df(4)}{dr1} & \dots & \dots & \dots \end{bmatrix}$$

with a line entry for each function, and where the columns align to the components. Note that $f()$ in the Jacobian is the Newton function for the gain, while the spot frequencies were defined by $sf()$. **$\Delta comp$** is solved by using LU decomposition on **J** to give:

$$LU \Delta comp = f \quad (7)$$

which can be solved with forward elimination and back substitution using an intermediate vector x :

$$L \cdot x = f \quad (8)$$

$$U \Delta comp = x$$

The whole process is then cycled until successive changes in the component values is less than a desired preset resolution.

Potential Hazard

Newton's method is not guaranteed to work every time. The potential hazard in this case is that the values of the capacitances are very small. This makes the differentials even smaller, giving rise to an overall differential function df/dc , which can be many orders of magnitude larger than that for the resistors which could lead to inaccuracy. Clearly double precision is mandatory. Scaling may help.

Results

Newton's method is well known for rapid convergence once the estimated values are close to the solution. Therefore, a useful guess for a starting point is to use the nominal component values as might be chosen directly from the time constants. After a few iterations, the components are, for $r_g=100$ and $r_3=0$, shown in **Table 2**.

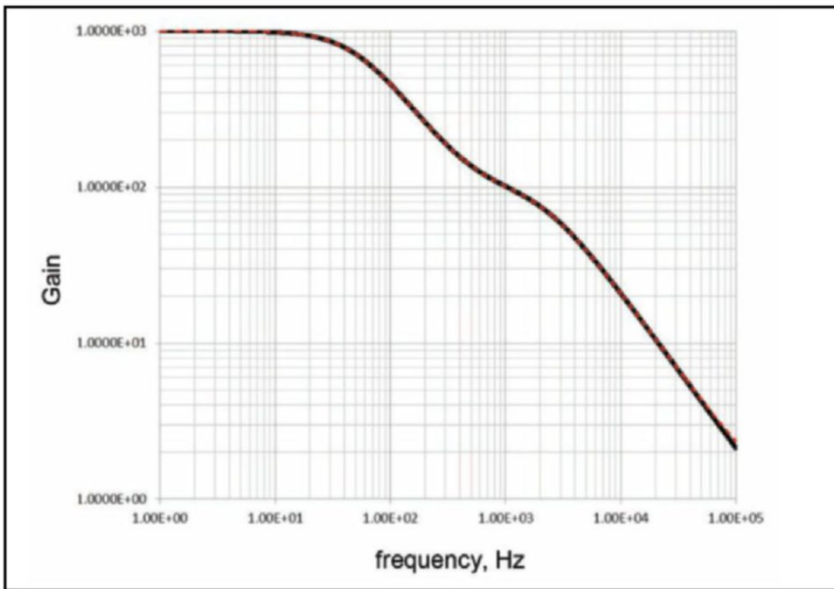


Figure 5: This graph shows a comparison of the ideal op-amp. Replay characteristics: solid line, transfer function response; dashed line, response with component values. These differ by less than 0.002dB between 1Hz and 3kHz and only deviates at high frequencies as the feedback network asymptotically approaches unity gain, which is correctable with an additional filter.

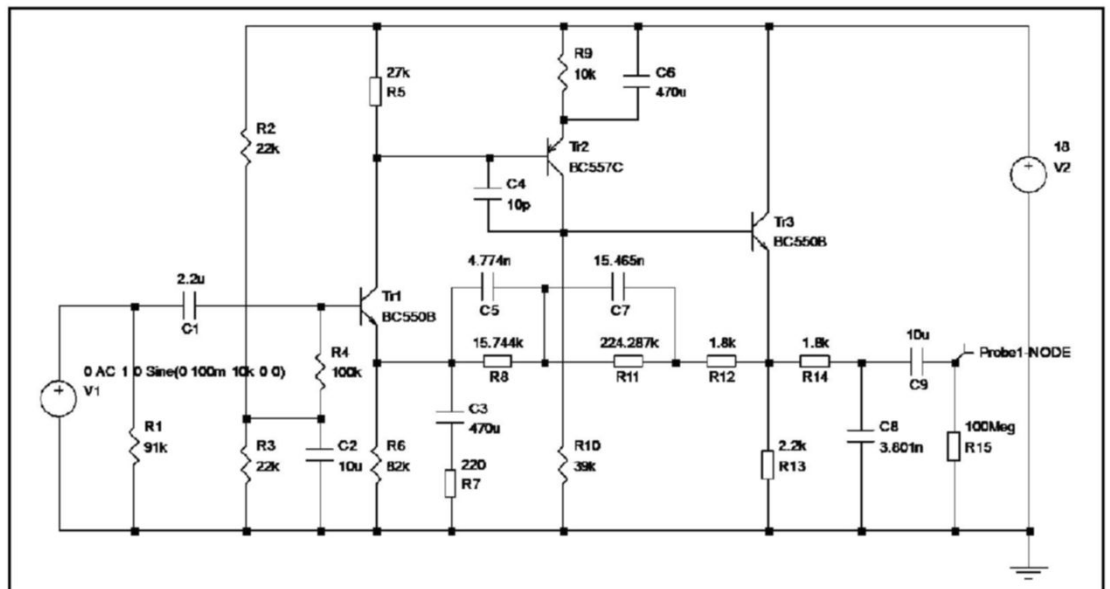


Figure 6: This schematic shows a simulated RIAA replay preamplifier.

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Table 3: The components' values for a three transistor circuit are determined from the Newton iteration.

Component	Value
r11	224.287k Ω
r8	15.744k Ω
c7	15.465nF
c5	4.774nF
c8	3.801nF

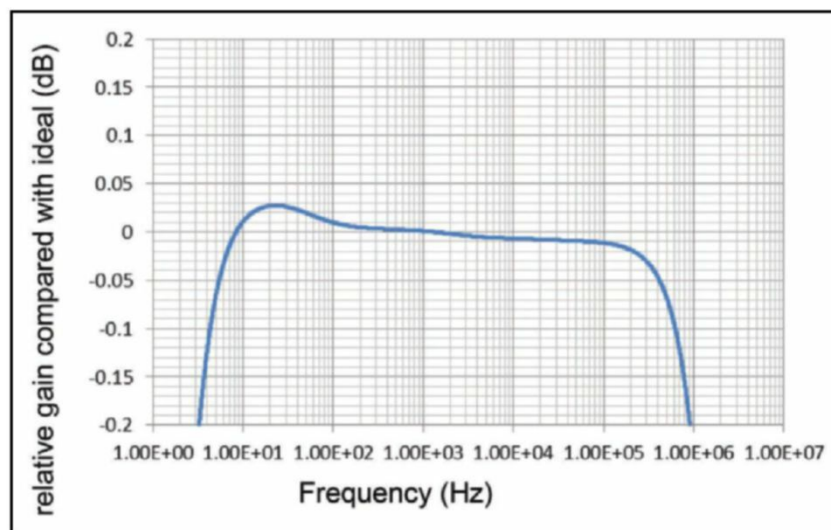


Figure 7: The three-transistor circuit performance is shown with calculated values of the RIAA network.

Table 4: Gain targets with the initial target and then revised from the results

Frequency	Exact Gain Target	Adjusted Target
1.0	1000	1000
100.0	455.91	455.91
1000.0	101.03	101.03
10000.0	20.784	20.817
100000.0	2.1216	2.1256

Table 5: Component values for the revised circuit open loop gain

Component	Value
r1	224.289k Ω
r2	15.744k Ω
c1	15.466nF
c2	4.7645nF
c3	3.792nF

About the Author

John Ellis learned to play the piano at an early age, and later the organ while at school. The combination of music and electronics led to his interest in hi-fi and audio design. Since graduating with first and higher degrees from the University of Kent at Canterbury, he has worked in the semiconductor industry, as a Device Engineer then as a Technology Development engineer. His responsibilities included developing processes for MOS technologies and devices including EEPROM cells, a BiCMOS process, and photodiodes for both blue light detection and image sensors. He has used commercial software for device and process simulation as well as writing programs to model devices at the physics level as well as compact models for circuit simulation. More recently he has worked on developing GaN LEDs.

A comparison of the results for an ideal op-amp is shown in **Figure 5**. The deviation due to the non-inverting configuration can be corrected with an additional external filter.

The method is now applied to a classical three-transistor RIAA preamplifier circuit as a practical example. The circuit shown in **Figure 6** is based on a classical two-stage amplifier plus emitter follower. Stability considerations demand an additional resistor R12 (R3 shown in Figure 4) in series with the RIAA network. To compensate the flattening of the response at higher frequencies a passive RC filter is added to the output, R14 and C8. The network expands to five unknowns to be solved, and the Newton functions in Equations 6a and 6b includes the additional resistor.

The amplifier gain is described by a simple transfer function of an open loop gain of 11000, as simulated using a BC307C or BC557C transistor in the position of Tr2, and a single time constant of 50 μ s due to the Miller capacitor C4 (with the collector-base capacitance of Tr2 in parallel). The components R8, R11, C5, C7 and C8 are determined from the Newton iteration shown in **Table 3**. The error between the response and the ideal RIAA curve is shown in **Figure 7**.

There is a systematic error toward the top of the spectrum. This may be due to the assumed simple gain model for A rather than the actual circuit. A correction can be applied by choosing a gain target, which differs from the exact RIAA by the error and re-calculating. The low-frequency error is mainly due to the input transistor emitter decoupling capacitor interacting with the input capacitor. That is addressed by increasing the input capacitor to 10 μ F. The second iteration with adjusted gain targets gives the values shown in **Table 4** and **Table 5**.

The changes to the circuit components are essentially to c2 and c3, which control the high-frequency performance. The updated simulated results are shown in **Figure 8**. Note that the scale is 10 times finer than that shown in Figure 7. The error is just -0.007dB at 100Hz and less than 0.002dB from 200Hz to 20kHz. The low-frequency error can be reduced by using larger coupling and decoupling capacitors.

This approach to designing an RIAA network should be regarded as an aid. Using an approximate transfer function for the amplifier yields an almost exact match to the RIAA characteristics (Newton typically reaches errors less than 10⁻⁸) but applying the results to an actual circuit may require adjustments to the gain targets by comparing the simulated output with the RIAA transfer function, and modifying accordingly. In

practice, variations in components in a circuit may give greater differences than the simulations shown. For example, a change from BC557C to a BC557B affects the open loop gain. And if the nearest preferred values are used for the resistors and capacitors, the error may reach up to $\pm 0.25\text{dB}$. It is, as Lipshitz mentioned, worth making the network as close to the target values as possible using series or parallel pairs. That works well with resistors, which are now routinely available at 1% tolerance. Capacitor values are more difficult as these typically are available to 5%. Paralleling several smaller devices to reduce the error only works if you can be certain that your samples are truly random. Ten devices from a production strip may well have a systematic offset, which makes this concept fail. You can scale the components by a small, constant amount, (resistors inversely to capacitors) without significantly altering the response, should that help.

Conclusion

Newton's method has been used as a design aid to solve an RIAA network. The discrepancy between the simulated circuit and ideal values may

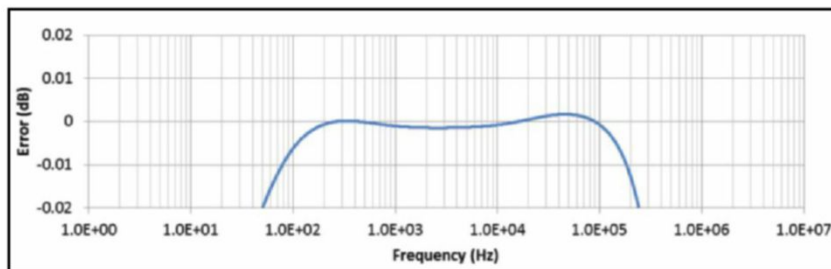


Figure 8: Error plot for the updated target gains, as simulated is shown.

be related to the simple amplifier model used, and needs further investigation. Nevertheless small deviations can be offset by modifying the target gains to obtain an improved match. 

Author Acknowledgement: Some of the simulations carried out for this article were conducted using SIMetrix software.

References

[1] S. Lipshitz, "On RIAA Equalization Networks," *Journal of the Audio Engineering Society*, Volume 26, No. 6, June 1979.



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The LNA-100

A Low Noise Amplifier with Filters and a RMS Voltmeter

This project describes the design and construction of the LNA-100 device, which comprises a low noise, high gain amplifier with three different gain settings, six filters for bandwidth restriction, and a True RMS voltmeter that gives a DC voltage at its output.



Photo 1: The completed LNA-100 device will be a handy companion on the test bench of any audio designer.

By

George Ntanavaras

The low noise amplifier (LNA), or the LNA-100 as I've dubbed this device, is based on top-quality op-amps and almost all the components are contained on a single PCB for easy construction. An external low noise power supply is used to supply the device. The completed device is shown in **Photo 1**.

Functionality of the Device

The device includes several functionalities as the block diagram of **Figure 1** shows. The low noise amplifier stages can switch between three gains: x100, x1000, or x10000 before the signal goes to the five different filters that are selected with a rotary switch, including a low-pass filter at 1kHz, a low-pass filter at 10Hz, a band-pass filter from 20Hz to 20kHz, a low-pass filter at 20kHz, and a "A-weighting" filter. The Monitor output can be used to measure or check the signal with an oscilloscope or an external voltmeter. The True RMS voltmeter measures the signal and gives a DC output voltage, which is directly analogous to its RMS value.

The Low Noise Amplifier Circuit

The more detailed block diagram of **Figure 2** shows the gain structure of the three-stage low noise amplifier. The first stage with a constant gain of x100 (40dB) is a very low noise circuit based on the ultra-low noise op-amp LT1028C from Linear Technology (now Analog Devices). According to its datasheet, the voltage noise is less than the noise of a 50Ω resistor. It has excellent high-speed specifications (the gain-bandwidth product

is 75MHz), distortion-free output, and true precision parameters (0.1μV/°C drift, 10μV offset voltage).

The second and the third stages are not very critical and I used the OPA2134 op-amps. Each one of them can have gain of x1 or x10 depending on the position of S1a and S1b switch.

When the switch is in position 1, the second and the third stage have 0dB gains and the total gain of the amplifier is x100 or 40dB. In position 2, the second stage has a gain of 20dB and the third 0dB, and the total gain is x1000 or 60dB. In position 3, the second and the third stage have gains of 20dB, setting the total gain to x10000 or 80dB.

Figure 3 shows the electronic schematic. Ferrite bead FB3 at the input rejects any EMI before it enters to the circuit. Capacitor C45 protects the circuit from any DC voltage component on the input signal. Resistor R32 provides a DC path for the bias of IC4 op-amp and resistor R33 with diodes D3 and D4 offer some protection of the input stage against large signals.

The first stage with the LT1028C achieves an excellence noise performance of 0.85nV/√Hz at 1kHz and 1.0nV/√Hz at 10Hz. A necessary condition for such low voltage noise is operating the input transistors at nearly 1mA of collector currents, since voltage noise is inversely proportional to the square root of the collector current. Current noise, however, is directly proportional to the square root of the collector current. Consequently, the LT1028C's current noise is significantly higher than most monolithic op-amps so the source resistance should

be kept as low as possible and according to the datasheet, less than 4kΩ.

The gain of the first gain is defined by the low value resistors R36, R37 for minimum noise. The capacitor C50 with R36–R37 give a high-frequency roll-off to additionally reduce the total noise. Resistor R56 buffers the output of the first stage while capacitor C47 blocks any DC offset of the first stage to enter the second stage, which is based on the half part of IC5 OPA2134 op-amp. The gain is defined by resistors R46 and R52. Switch S1 selects if R52 will be connected or not to the circuit and thus the gain could be x10 or x1. The capacitor C42 with R46 give a high-frequency roll-off.

The third stage is similar to the second stage and is based on the other half part of OPA2134. Switch S1 can select if resistors R49 and R53 will be connected to the circuit giving a gain of x10 or x1 in combination with resistor R47.

Resistor R3 buffers the output of the preamplifier and C49 blocks any DC offset from entering to the next stage.

The Filters

Filters play an important role for audio noise measurement and are applied to reduce signal components at selected frequencies.

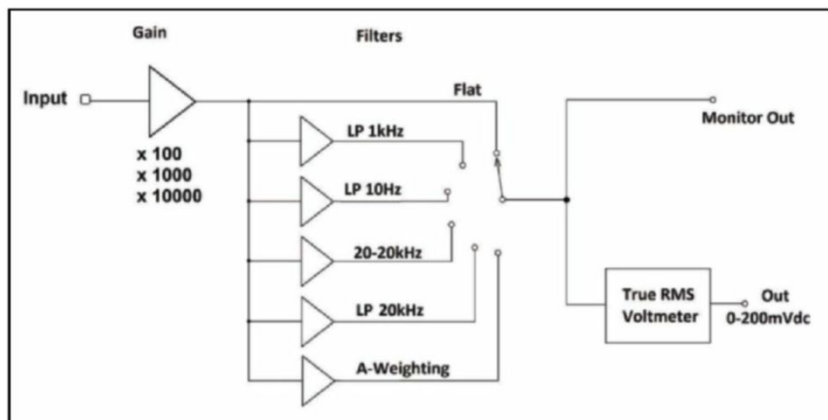
I included the following filters in the device:

- “A-weighting” filter
- 20kHz low-pass filter
- 20Hz to 20kHz band-pass filter
- 10Hz low-pass filter
- 1kHz low-pass filter

The electronic schematic diagram of all the filters is shown in **Figure 4**. A-weighting is the most commonly used curve for noise measurement since it is responding at pure tones similar to the sensitivity of the human ear. The A-weighting filter is based on the components around IC3. I chose a combination of passive and active components because I wanted to design a very accurate “A-weighting” filter.

To achieve this, I used the demo version of the LspCAD 6 software (see Resources). The demo version contains all the functionality of the unlimited version except it is not possible to save the projects and to export data.

The program has a lot of capabilities mainly for the design of loudspeaker cabinets and crossovers, but I used only its Optimizer tool for the design of the filter. I entered the initial topology of the “A-weighting” filter and the target response as an ASCII text file. I spent a lot of time changing the



topology of the circuit and the initial component values, until finally I succeed to have a very accurate response using only standard values for the components.

I used the LTSpice simulation software to compare the response of the calculated filter to that of the ideal A-weighting filter. The differences that I found between the two curves were less than ±0.05 dB, an excellent result.

Practically, this can be done only if the exact calculated values of the components as shown in the schematic are used in the assembly of the real circuit. This is not very easy to achieve because of the tolerance of each component. I didn't want to spend a lot of money by buying very high accuracy components, so I bought a larger quantity from what I needed and I measured the value of each resistor and capacitor, choosing the ones with a value closer to the calculated values. I have found that this is a good method to construct higher accuracy circuits without increasing the cost too much.

A special note is required concerning the value of the capacitor C24. The optimum value to keep the high accuracy of the filter above 10kHz is 455pF. This is nominally achieved by paralleling three capacitors of 220pF, 220pF, and 15pF. Since the value of 455pF is about 3% lower from 470pF, if you have a batch of 470pF, 5% capacitors, by measuring

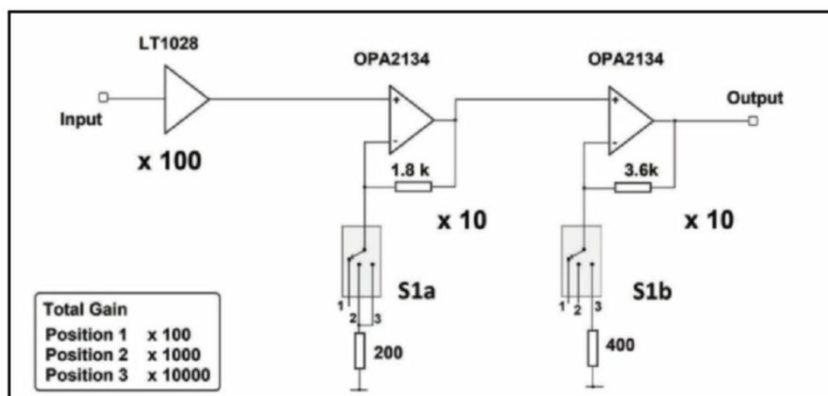


Figure 2: This more detailed block diagram shows the gain structure of the three-stage low noise amplifier.

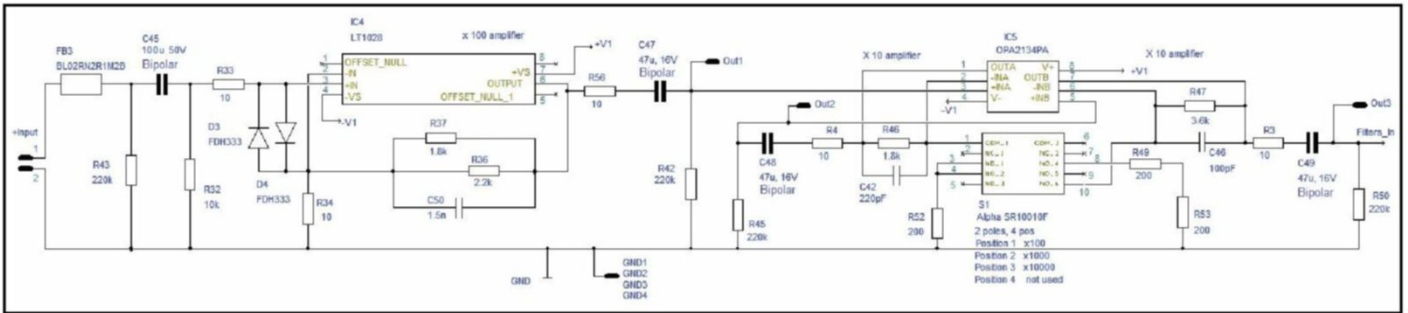


Figure 3: This is the electronic schematic for the gain stages of the LNA-100.

them with a capacitance meter you could find one that has the exact value of 455pF. This is what I did in my prototype. I measured the accuracy of the assembled filter and it was ± 0.1 dB between 20Hz and 20kHz.

The other filters are active second-order Butterworth filters using the Sallen-Key topology. Butterworth filters have maximally flat-magnitude response with an attenuation of -3dB at the cutoff frequency. Above or below the cutoff frequency, depending on the type of the filter, the attenuation is 12dB/octave. The transient response of a Butterworth filter to a pulse input shows moderate overshoot and ringing.

The 20kHz low-pass filter is built around IC2A

with the components R14, R21, C15, C16 and C23 and the 20Hz high-pass filter is built around IC2B with the components R11, R15, C14 and C19. The 10Hz low-pass filter is built around IC3A with the components R5, R9, C3, C4 and C8, while the 1kHz low-pass filter is built around IC3B with the components R6, R10, C1, C5 and C9. Switch S2 selects which filter will be used by the device. The last position bypasses the filters and connects the output of the low noise amplifiers directly for a flat response.

The Power Supply Circuit

The device has a consumption of about ± 50 mA and is supplied from an external very low noise

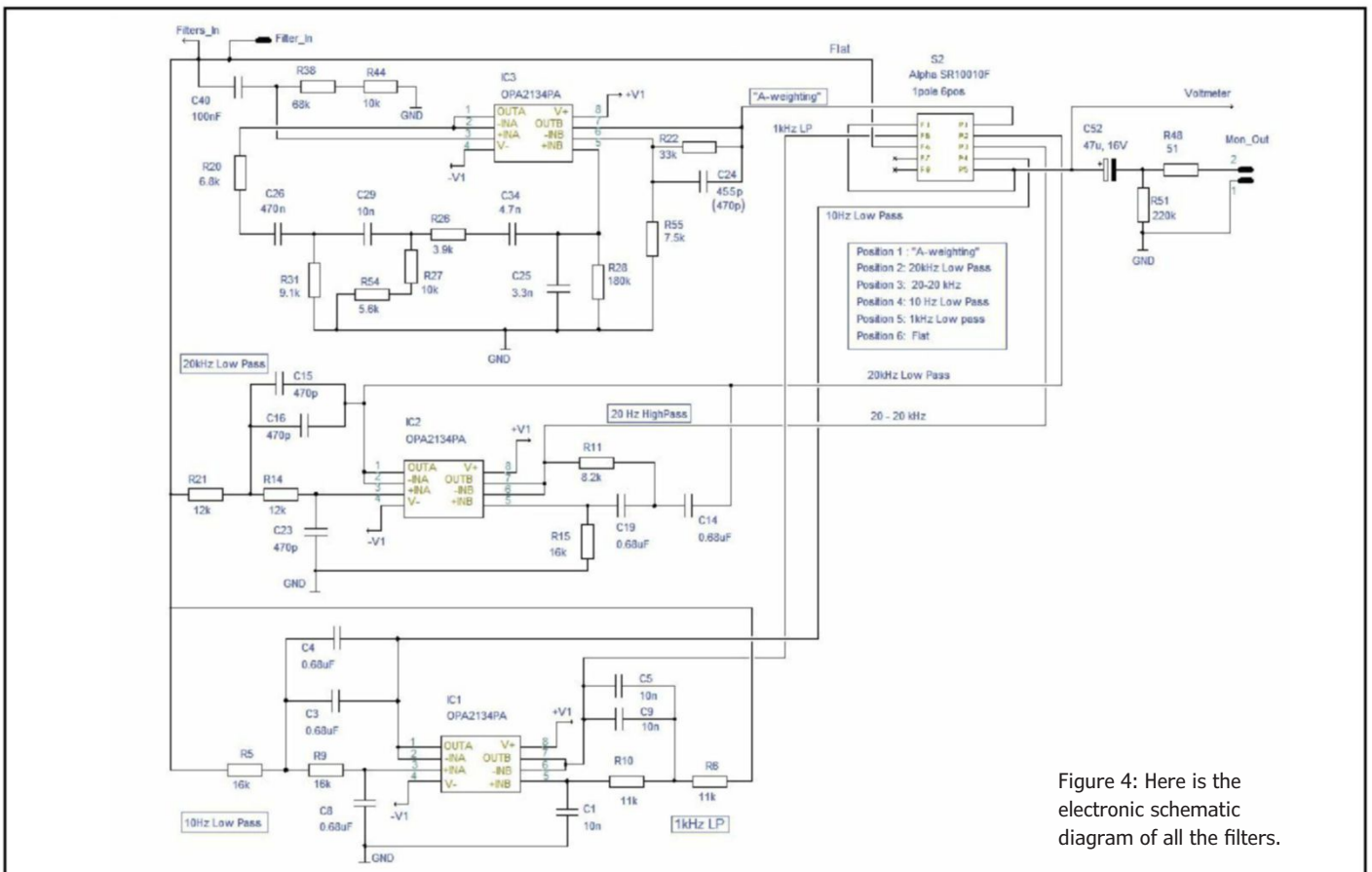
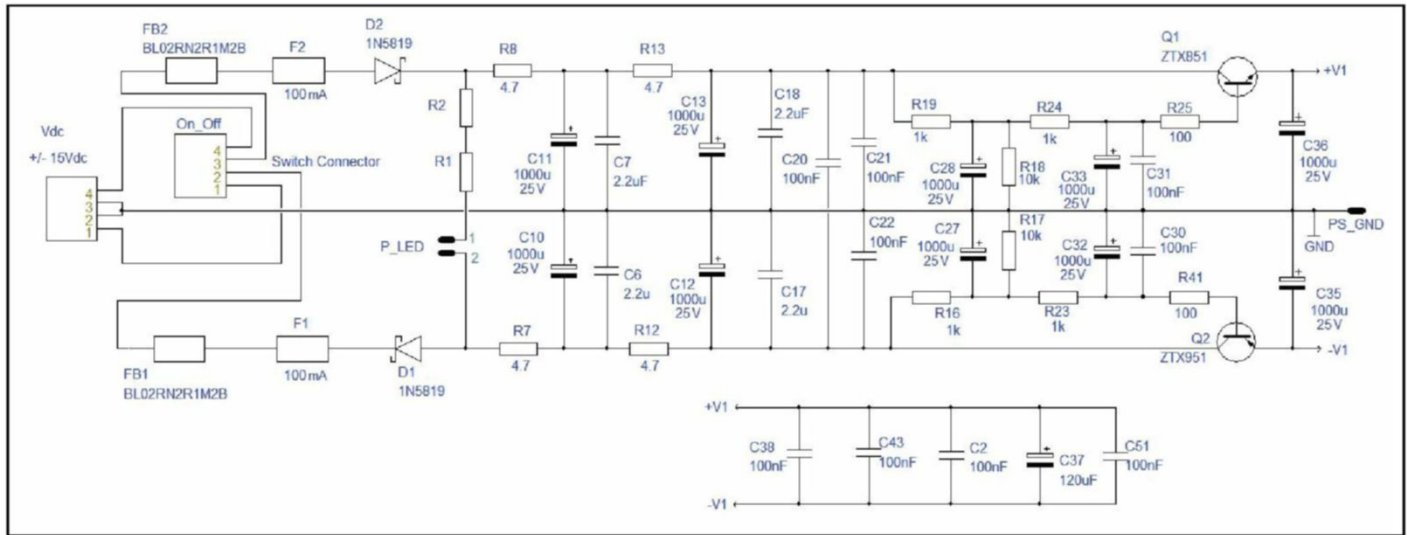


Figure 4: Here is the electronic schematic diagram of all the filters.



regulated power supply of $\pm 13.4\text{V}$ DC. I built the power supply using an old PCB left from my PR-3 preamplifier project. This power supply uses the LM317 and the LM337 as pre-regulators with their output set at $\pm 15\text{V}$. These voltages drive the inputs of the SMD regulators LT1763 and LT1964, which are micro-power, low noise, low-dropout regulators with their output voltage set to $\pm 13.4\text{V}$ DC. More details can be found within the PR-3 article, published in this magazine in 2019 (see Resources).

Through a 5-pin DIN connector that is placed on the back panel of the device, the voltage goes to the PCB's power connector. The schematic of the internal power supply circuit of the LNA-100 is shown in **Figure 5**. Fuses F1 and F2 are slow-blow fuses, mounted on the PCB to protect the external power supply from short circuits within the device. FB1 and FB2 are ferrite bead inductors offering electromagnetic induction (EMI) protection from several megahertz (MHz) up to gigahertz (GHz)

by suppressing any noise coming from the power supply. Resistors R7, R8, R12, R13 with capacitors C6, C7, C10, C11, C17, C18 form low-pass filters to eliminate any possible noise from the external power supply and the connecting cables.

The low noise transistors Q1 and Q2 form a capacitor multiplier circuit for the positive and the negative side, respectively, to further eliminate any noise. Resistor R17 and R18 are used to set a 2V DC voltage drop between the collector and the emitter of Q1 and Q2 for the proper operation of the circuit.

Figure 5: This is the schematic of the internal power supply circuit of the LNA-100.

The True RMS Voltmeter Circuit

Figure 6 shows the electronic diagram of the True RMS voltmeter. It is based on the AD736 chip of Analog Devices, which is a low-cost, low-power, precision, monolithic true RMS to DC converter with a maximum error of $\pm 0.3\text{mV}$, $\pm 0.3\%$ of reading with sine wave inputs.

Its frequency response is 190kHz when the

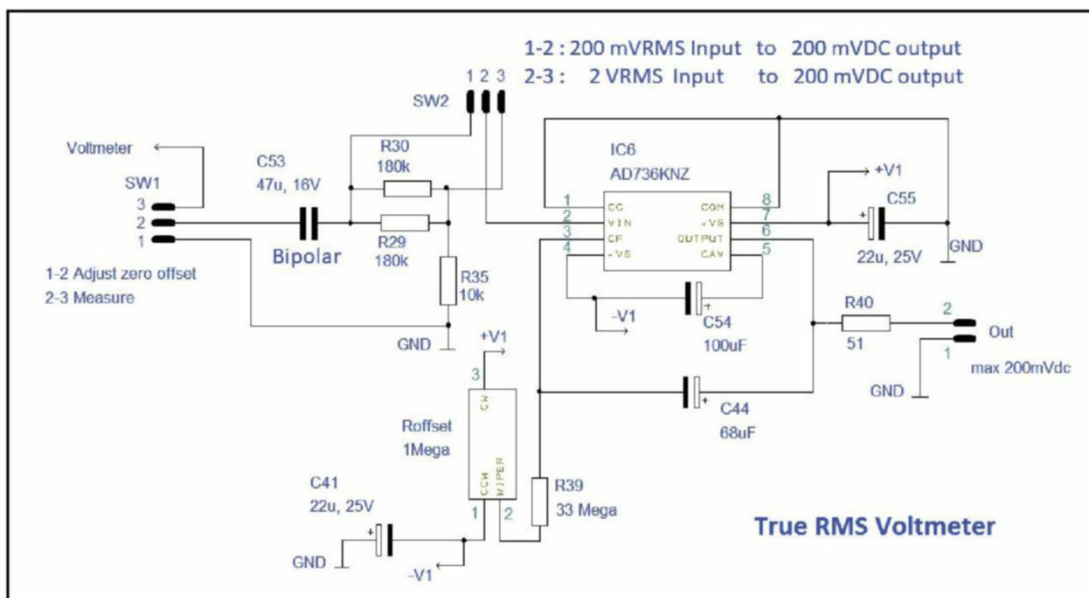


Figure 6: The electronic diagram of the True RMS voltmeter is based on the AD736 chip of Analog Devices, which is a low-cost, low-power, precision, monolithic true RMS to DC converter.

Figure 7: Here is the PCB's complete guide assembly.

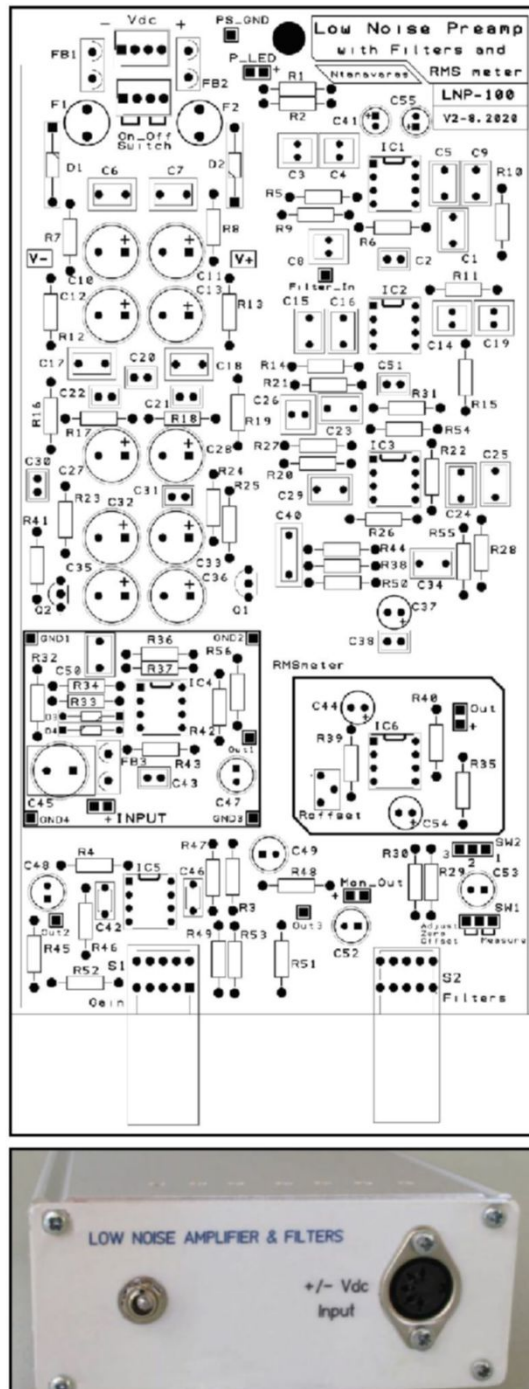


Photo 2: The back panel of the LNA-100 has a 5-pin DIN connector for the connection of the external power supply to the device on the right side, and on the left side the ON-OFF switch.

Project Files

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

Resources

DesignSpark, www.rs-online.com/designspark/pcb-software

Front Panel Designer, Schaeffer AG, www.schaeffer-ag.de

LspCAD program, IJData, www.ijdata.com/LspCAD_demo.html

G. Ntanavaras, "PR-3, A High Quality Stereo Preamplifier," *audioXpress*, January 2019, <https://audioxpress.com/article/you-can-diy-the-pr-3-a-high-quality-stereo-preamplifier>

input signal level is 200mVrms, it drops to 55kHz at 10mVrms input and becomes only 5kHz with an input signal of 1mVrms. This means that it is very important to adjust the total gain of the device so that the input voltage to the voltmeter will be more than 10mVrms in order to avoid measurement errors due to the limited bandwidth at very low levels.

The AD736 accepts a maximum 200mVrms full-scale AC input voltage giving a DC output of 200mV and maintains high accuracy while measuring a wide range of input wave forms. An input attenuator can be used for larger voltages. R29, R30, and R35 form a 20dB (x10) input attenuator and switch SW2 (which is the scale switch in the front panel) selects if it will be active or not. With the attenuator in the circuit, the voltmeter has the capability to measure a maximum 2Vrms input signal giving an output voltage of 200mV DC.

I also included the offset adjustment circuit indicated in the datasheet of AD736. The jumper switch SW1 selects if the input of the AD736 will be connected to the input signal (connection pin 2 with pin 3) or to the Ground (connection pin 1 with pin 2) for adjusting trimmer offset to give zero offset at the output of the voltmeter.

Resistor R40 buffers the output of the IC. The rest of the circuit is according to the datasheet of the IC, which should be studied if more details are required.

The Construction of the Device

I used the Design Spark PCB software to design the PCB. The outcome was a 200mm x 130mm double-sided PCB with ground planes on the top and bottom layers. **Figure 7** shows the PCB's complete guide assembly. The Parts List, which can be found in the Supplementary Material section of the audioXpress website, shows what is required to build the device.

For the assembly of the PCB, I started by placing the components from the smaller to the larger ones. I measured the value of each resistor that is used to adjust the gains of the op-amps and I chose the ones that were most closed to the nominal values. In this way the gains of the op-amps were very accurate without the need to use high precision resistors. For all the op-amps, I used high-quality IC sockets with gold-plated pins to facilitate the testing or any repairs if necessary.

I used a silver anodized aluminum extruded box to house the components of the device. The enclosure had the following external dimensions: Width=108.5mm, Depth=220mm, and Height=45mm.

The front panel of the device is shown in Photo 1. On the left side, a BNC connector accepts the

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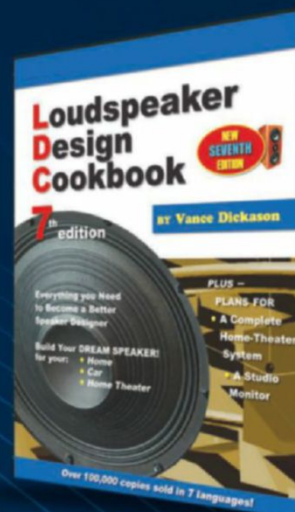
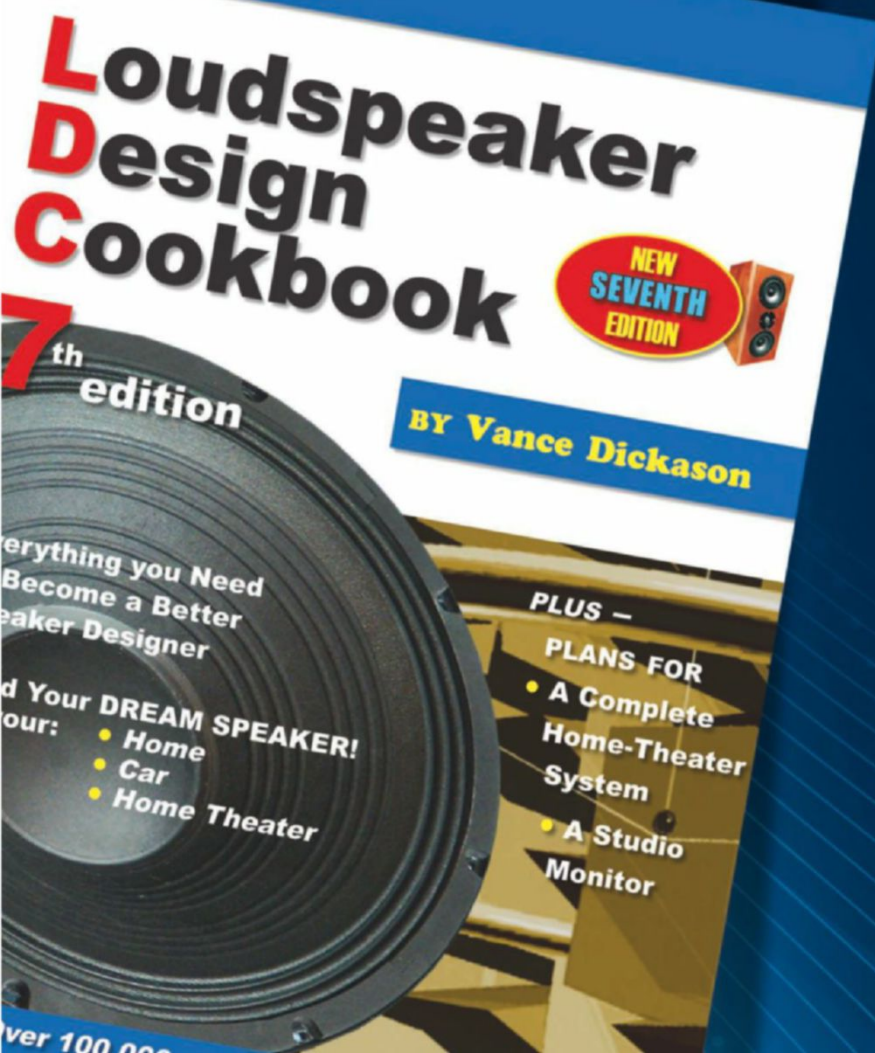
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input signal, a red LED indicates the operation and a rotary switch selects the gain. In the middle of the front panel, a BNC connector gives the Monitor output of the device. On the right side there is another BNC connector for the DC output of the RMS voltmeter and the 'Scale' toggle switch for the

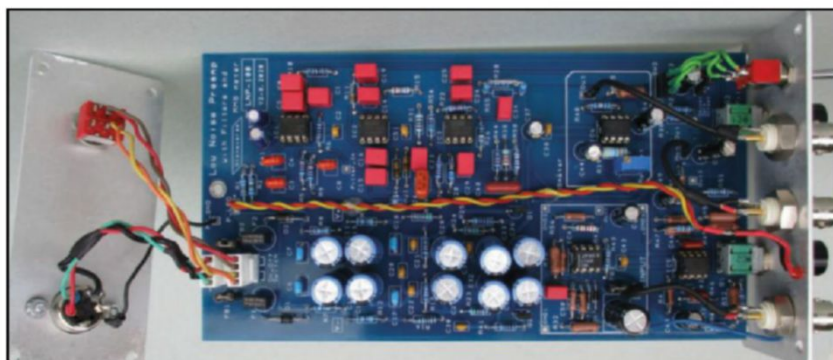


Photo 3: This view inside the device shows the wiring with the connectors and the switches of the front and the back panel.

Total Output Noise with Input Short-Circuited		
Noise Filter	LTSpice simulation	Measured
A-weighting	1.19mVrms	1.2mVrms
20Hz – 20kHz	1.48mVrms	1.6mVrms
Total Output Noise with 1000Ω at Input		
Noise Filter	LTSpice simulation	Measured
A-weighting	5.48mVrms	5.6mVrms
20Hz – 20kHz	6.78mVrms	6.9mVrms
Total Output Noise with 11000Ω at Input		
Noise Filter	LTSpice simulation	Measured
A-weighting	18.88mVrms	20.8mVrms
20Hz – 20kHz	23.61mVrms	25.9mVrms

Table 1: The table summarizes the calculated versus the real measured noise voltages.

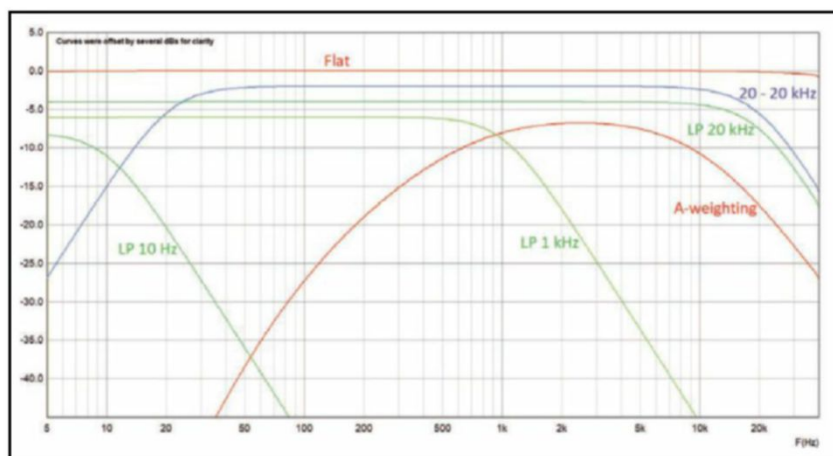


Figure 8: This graph shows the frequency response of the five filters used in the device as measured.

selection of the meter's scale. Below is the rotary switch that selects which of the filters will be active or if the bandwidth will be flat.

The text on the front panel was created using the "Front Panel Designer" program (see Resources). The file was printed on a transparent self-adhesive sheet with my ink-jet printer.

The back panel of the LNA-100 is shown in **Photo 2**. On the right side, there is a 5-pin DIN connector for the connection of the external power supply to the device and on the left side the ON-OFF switch. **Photo 3** shows the inside of the device, including the wiring with the connectors and the switches of the front and the back panel.

When the assembly of the PCB was completed (and without the ICs in their sockets), I connected the external power supply and I checked that the voltages at the supply pins of all ICs sockets were correct. Then, I inserted the ICs and I measured to ensure that the voltage at the supply pins of the ICs were close to $\pm 11.7V$ DC.

The trimmer offset should be adjusted to measure 0V DC at the True RMS voltmeter's DC output with no signal at the input. To do this, the internal jumper switch SW1 should be set to position 1-2, the amplifier input should be short-circuited, and the Scale switch SW2 should be set to position x10. The trimmer is adjusted slowly until the output shows 0mV DC. This will take a few minutes to stabilize and should be readjusted over the course of an hour or two. It is also a good idea to re-check it and adjust the offset, if necessary, after the device has operated for a few weeks. The Scale switch SW2 should also be set at position x1 and confirm that the offset is also zero. The offset in my prototype remained over time within $\pm 0.02mV$ DC. Don't forget to restore the SW1 jumper to the 2-3 position after the offset adjustment for the normal operation.

I tested the operation of the device using a signal generator connected to its input and an oscilloscope connected to its output. I measured the voltage gains of the amplifier for each one of the three positions and they were very close to the nominal values. To do this measurement, it was necessary to connect an attenuator of 40dB before the input of the LN-100, otherwise the output will clip when the gain is very high.

Other Measurements of the Device

As a crosscheck for my noise measurements, I used the LTSpice simulator to calculate the total noise levels for all the three amplifier stages comprising of IC4 and IC5. For the simulations, I set the total gain to x10000 and I used two type of the filters: the A-weighting and the 20Hz to 20kHz.

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.



Table 1 summarizes the calculated versus the real measured noise voltages. If the output noise is divided by the gain of the amplifying circuits, it will give the noise referred to the input. This was calculated at 160nVrms with the input short-circuited and the bandwidth defined from 20Hz to 20kHz.

Figure 8 shows the frequency response of the five filters used in the device as measured. The curves shown were offset by several decibels for clarity. In the Flat position, the $\pm 3\text{dB}$ bandwidth of the device is from 1.5Hz to about 90kHz. I measured the total distortion of the LNA-100 at the gain of $\times 10000$ and the result was below 0.02% at 1kHz.

I checked the True RMS Voltmeter accuracy using my TTI 1604 True RMS Multimeter, which provides 4.75 digits for measuring AC voltages with an accuracy of 0.5% for the frequencies between 45Hz and 400Hz. From 400Hz to 5kHz, the accuracy drops to 1% and as the level and the frequency are increased, the accuracy gets worse.

The results using sinus waveform as input were: at 220Hz the TTI measured 152.4mVrms and the device 152.22mV DC, at 1kHz 152.8mVrms and 152.65mV DC and at 20kHz 149.4mVrms and 149.31mV DC.

I also checked the accuracy with a square wave input. At 20kHz, the TTI measured 187.9mVrms while the DC output of the LNA-100 measured 187.81mV DC. I was very surprised because all the measurements were almost identical in comparison, but apparently this was to be expected because, as I found out later after a little search, the TTI is based on the similar AD737 true RMS converter IC.

Conclusion

The LNA-100 is a versatile and easy-to-construct device. It can measure very low level signals and has a lot of features. If you need such a device, I am sure you will enjoy building and using this one as I did. 

Author's Note: I have a few PCBs available for the construction of the LNA-100 device. If you are interested in obtaining one, please send me an email at: gntanavaras@gmail.com

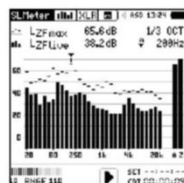
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Power Transformer Parameters, Selection, and Testing

Part 2 — Transformer Cores and Their Construction Methods

In the second part of this article series, Chuck Hansen details the history of transformer cores and shares information about their various construction methods.

By
Chuck Hansen

In 1878, the Ganz Works in Budapest, Hungary designed a transformer based upon two copper wires wrapped around an iron ring (toroid). That first toroidal transformer was used to construct AC lighting systems throughout Europe.

The first practical E-I core transformer was developed by William Stanley in 1885 using his lower-cost iron laminations. George Westinghouse hired him to develop the transformers to provide the first long-distance AC lighting to customers in Great Barrington, MA in 1886.

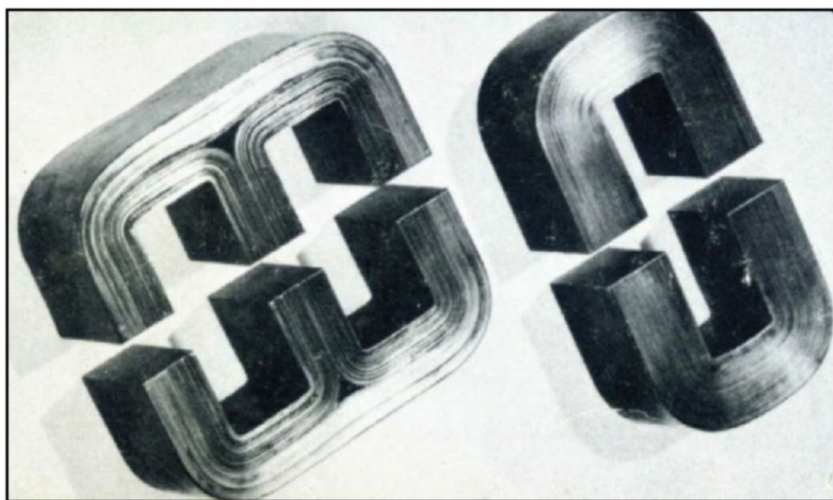


Photo 1: C or E cores are cut in half, and the mating halves ground, lapped, and polished so the mated cores have a total air gap of less than 0.004 mm (0.1 mils). (Image Source: Arnold Engineering)

The introduction of silicon steel in 1932, grain-oriented silicon steel in 1933, and insulated surface coatings were major improvements in transformer core materials.

C cores were first used by Westinghouse in distribution transformers around 1940. In my work, we used smaller C and E cores made under license by Arnold Engineering for our aerospace products. The first Arnold catalog I have is dated August 1966. In the early 1980s, the first MetGlas amorphous metal cores were introduced by Honeywell to reduce the energy losses in utility power transformers. In 1996, Nicore in China started to produce the first R cores and O cores. Let's look at each of these core types in more detail.

Tape Wound Cores

A customer making tape wound C or E cores will slit the sheet steel to final width for winding on rectangular mandrels. C or E cores (**Photo 1**) are cut in half, and the mating halves ground, lapped and polished, so the mated cores have a total air gap of less than 0.004 mm (0.1 mils). The core halves are ink-stamped with the core part number printed across the gap so it is obvious how they should be assembled, then packaged in shrink-wrap plastic (**Photo 2**).

Toroidal cores are made in a similar fashion, except the winding mandrel is round and produces finished bare cores. These cores can then be painted, coated with epoxy or, for nickel

alloy cores, have a protective core box installed (**Photo 3**).

O cores and R cores are also tape-wound cores. The core tape is cut from flat sheets of cold-rolled grain-oriented silicon-steel using electrical discharge manufacturing (EDM) with specially programmed automated cutters (**Photo 4**).

The width is narrow at the start of the core, winding then gradually grows larger until the maximum outer diameter (OD) is reached half way through the wind. Next, the taper decreases to the minimum width again at the outer end of the core. You can see the start and finish tape ends of the O core (**Photo 5**) and the R core (**Photo 6**).

Laminated Cores

A manufacturer making laminated cores for E-I or the I portion of C-I core transformers and inductors (**Photo 7** and **Figure 4**) can cut the strips into laminations using a blanking die punch press, laser, photochemical etching, shearing, waterjet, or EDM.

For some commercial transformers, cold-rolled non-grain oriented (CRNGO, or just NGO) silicon steel laminations might be more cost-effective, since the extra grain recrystallizing annealing processes required for CRGO steel are not needed. As mentioned earlier, generators and motors use NGO since the rotor pole grain orientation changes constantly with respect to the stator windings as the shaft rotates. For large industrial and utility machines the stator can be segmented such that the grain orientation always effectively aligns with the rotor poles.

For large diameter or high-speed machines, the manufacturer might need to use a high-strength steel such as 4140 in order for the rotors to withstand the high centrifugal force.

There have been alternatives to the CRGO grades of transformer steel, such as amorphous MetGlas 2605SC which, compared to the CRGO steel, has much lower core loss when used in distribution transformers. Still, CRGO 3% silicon steel is the most common choice in the industrial and utility transformer industry, and meets or exceeds many industry standards, such as American Iron and Steel Institute (AISI), American Society for Testing and Materials (ASTM) International, and International Electrotechnical Commission (IEC), among others. There are also transformer steel MIL-specs based on the Society of Automotive Engineers (SAE) International specifications.

Cobalt Magnetic Cores

Many aerospace transformers also use grain-oriented 3% silicon-steel, but high-performance



Photo 2: The C core is packaged for shipment. (Image Source: Magnetic Metals)



Photo 3: Toroidal cores are made in a similar fashion to the C and E cores, except the winding mandrel is round and produces finished bare cores (Image Source: Magnetics, Inc.)



Photo 4: Sheet metal manufacturing is conducted with electrical discharge manufacturing (EDM). (Image Source: SK Electric)



Photo 5: These are the start and finish tape ends of the O core. (Image Source: James Transformer)

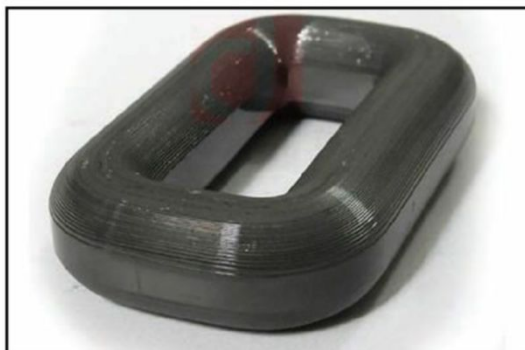


Photo 6: These are the start and finish tape ends of the R core. (Image Source: James Transformer)

Photo 7: To make laminated cores for E-I core transformers and inductors cut the strips into laminations using a blanking die punch press, laser, photochemical etching, shearing, waterjet, or EDM. (Source Image: Alibaba.com)



laminations such as vanadium permendur Fe49-Co49-V2 (Supermendur) are required to meet SWaPc.

Supermendur has the highest maximum flux density (2.4 Tesla) of any magnetic material, as compared with CRGO silicon-steel (1.97T), see **Figure 5**. They use a high-temperature ferric oxide film insulation rather than C5. They also don't have any particular grain orientation.

Cobalt is a very expensive metal even though it is the 23rd most abundant element. More than half the world's cobalt ore supply is mined by child labor in the Congo and is therefore considered a conflict mineral.

Nickel Magnetic Cores

Nickel-iron (NiFe) alloys are characterized by relatively high permeability and low core losses. Nickel-iron alloys can also be brittle and sensitive to shock and vibration, so NiFe E-I laminated transformer cores are not generally available.

Nickel alloys can be non-grain oriented, or processed to have a desired grain orientation. Also, nickel is highly magnetostrictive, so even tape-wound nickel toroidal cores require a nylon, phenolic or epoxy coated stainless steel or aluminum core box.

Copper has a higher thermal coefficient of expansion than any of the magnetic transformer core materials (16.5 ppm°C vs. 11.7 for SiFe or 13-15 for nickel and cobalt alloys). At low temperatures,

the copper wire shrinks, and rather than compress the sensitive magnetic core material, it compresses the core box instead.

The stainless steel and aluminum core boxes that are used with nickel alloy cores do not wrap around the full cross-section of the core, since that would make a single-turn short-circuit. The metal part of the case is interrupted on one side with an insulated plastic spacer (more on that later). For higher frequencies, powder metal nickel cores are used, which are less sensitive to the mechanical issues.

Plastic or epoxy core boxes are also used on silicon-steel cores to prevent the copper wire insulation from being compromised by the edges of the steel core tape. Painted SiFe cores are available for use where space is at a premium, but they need an inspection to make sure there are no exposed core tape edges. I prefer to use phenolic or epoxy core boxes.

Invar is a 36% NiFe alloy that also has the lowest coefficient of expansion. It was first used for the timekeeping balance wheel in mechanical watches. Now it is also used for high-frequency inductors and transformers.

Smith in the UK makes three versions of another nickel alloy; Radiometal (36% NiFe), Radiometal-4550 (45% NiFe), and Superradiometal (48% NiFe). These alloys have maximum flux density of 14 kgauss to 16 kgauss (1.4T to 1.6T), and permeability greater than 40,000. They are annealed in a dry hydrogen atmosphere.

In the early 1970s, we began to use a new 54% nickel core material called SatmuMetal for aerospace voltage and current instrumentation and protection transformers. It was developed by British company Telcon Metals, Ltd., with the toroidal cores wound in the US by the Magnetics division of Spang.

SatmuMetal required careful handling since any shock caused by sharp bends during sheet cutting and spooling could disrupt its grain alignment, leading to localized loss of permeability, which if severe enough, could require a repeat of the annealing steps.

Satmu has a decent maximum flux density of 15 kgauss (1.5T), compared to 1.97T for silicon-steel. But it has a much lower 400Hz H1 reset coercive force of 0.16 Oersteds, compared to 0.450e to 0.650e for 3% silicon-steel. It also has an initial permeability μ_0 of 65,000 vs. 9,000 for 3% silicon-steel.

These properties produce very low core losses and more accurate current and instrument transformers. SatmuMetal is longitudinally annealed in a dry hydrogen atmosphere and a magnetic field

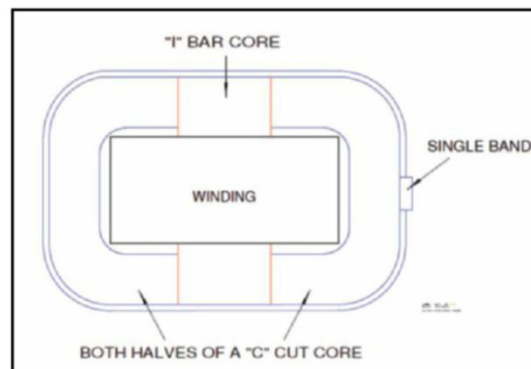


Figure 4: The layout for the C-I core laminations is shown. (Source Image: Magnetic Metals)

to maximize its permeability, and it became very expensive. Fortunately, we could go back to 3% silicon-steel for most of our current transformers without requiring re-qualification or increasing the size, weight or cost (SWaPc again).

MuMetal, with 77% nickel (Ni77-Fe16-Cu5 and Cr2 or Mo2) is another permalloy material. It is used for electromagnetic shielding both in aerospace and high-end instrumentation and audio equipment. Its maximum flux density is about 0.75T, but the core losses are an order of magnitude lower than 50% nickel alloys.

The shield metal absorbs electromagnetic energy and converts it to heat. For a magnetic shield, you use materials with the highest peak permeability. This will minimize the thickness of the shield and maximize its effectiveness, but you have to make worst-case magnetic field measurements to design it properly and keep it out of saturation.

As with SatmuMetal, MuMetal is made with an annealing heat treatment process in a hydrogen atmosphere and magnetic field that increases the number of magnetic domains, which gives it a high E-field (electric) and H-field (magnetic) attenuation factor.

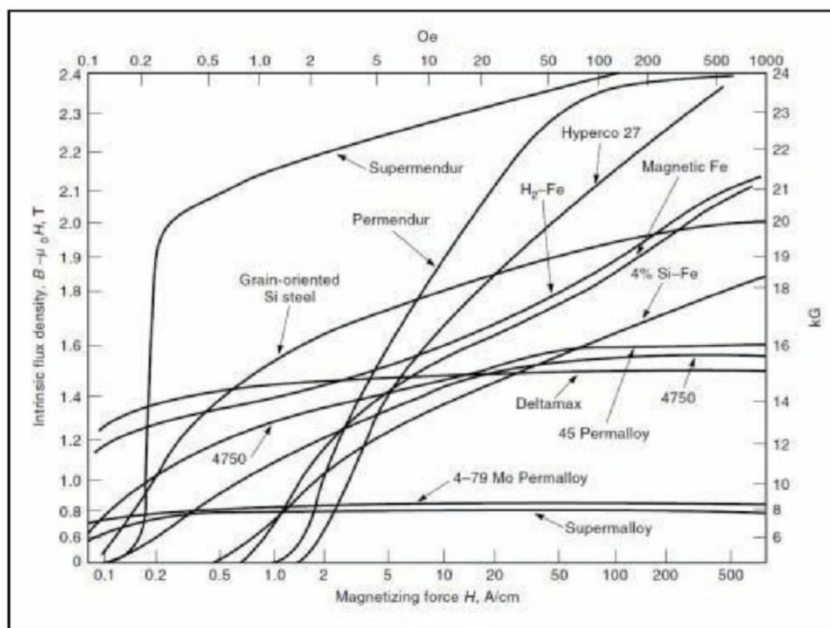


Figure 5: Initial B-H curves are shown for selected soft magnetic materials (Source Image: Arnold Magnetic Technologies)

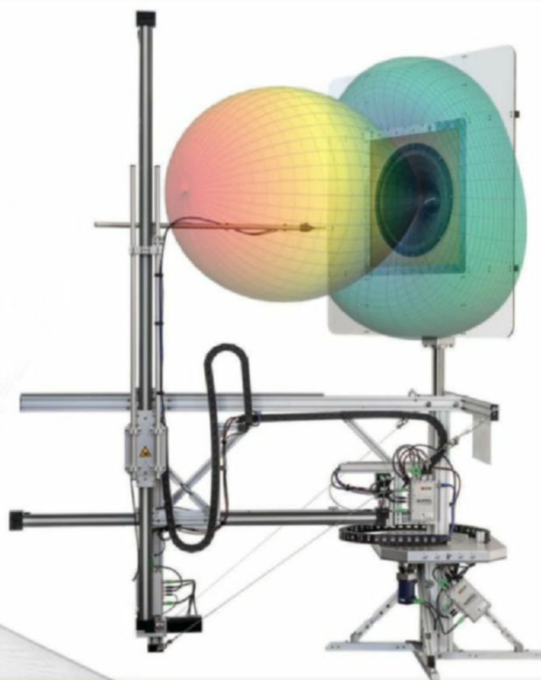
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Nickel alloys 4750, 48, and 49 are also shielding alloys made of Ni48-Fe50. The balance of the alloys are Chromium (Cr), Manganese (Mn), Silicon (Si), Carbon (C), Phosphorus (P), and Sulfur (S). Their other properties depend on the annealing process.

We made the emergency static inverter (28V DC to 115V AC 400Hz) for several airliners. To meet SWaP-c, we nickel-plated a thin piece of aluminum that functioned as the electromagnetic shield between the switching power transistors and the low-level sensing, control, and protection circuit board. This was much lighter than a piece of MuMetal of the same thickness, and it didn't require any annealing.

Another popular series of aerospace laminated core material is 80% nickel-based permalloy. Supermalloy (Ni80-Fe15-Mo5) has much lower losses than cores with higher iron content. It has a variation of flux density with temperature

comparable to silicon-steel, although its maximum flux density is only 0.84T.

We also use a number of nickel alloys in the Orthonol family. These cores are grain-oriented Ni50-Fe50 alloys, and they are available in both square and round hysteresis-loop characteristics (**Figure 6**). Square and round are used to describe the shape of the B-H curve between maximum linear flux density and core saturation.

Square loop alloys are manufactured to meet exacting circuit requirements for very high squareness and high core gain, and are typically used in power transformers, saturable reactors, high-gain magnetic amplifiers, and power inverter-converter applications. Other applications such as flux counters and transducers, demanding extremely square hysteresis loops, require Square Orthonol. Silicon steel (Microsil) and Supermendur are also square loop alloys (see the yellow highlight in Figure 6).

Round loop alloys have lower squareness and core gain than the square type, as these characteristics are sacrificed to produce their high initial permeability and low coercive force properties. Round Permalloy 80 is especially useful in designing highly sensitive input and inter-stage transformers where signals levels are extremely low and DC currents are not present. It is also useful in precision potential (voltage) and current transformers where losses must be kept to a minimum and high accuracy is a necessity.

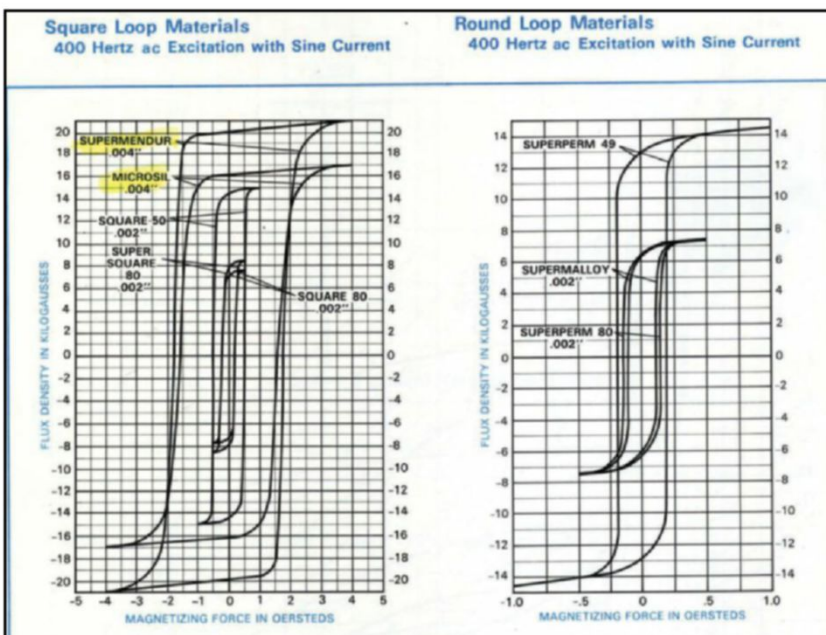


Figure 6. Composite hysteresis curves for square loop and round loop materials (Magnetic Metals Company)

About the Author

Chuck Hansen is an Electrical Engineer and holds five patents in his field of electric power engineering. He has worked in both the electric utility and aerospace electric power industries. He was assigned as the Instrument and Controls Startup Group Lead Engineer for two nuclear power generating units. He was the Electrical Systems and Controls Supervisor for a number of military programs as well as commercial and business jet electric power programs. He has written two books for Audio Amateur publications, and more than 260 magazine articles. He began building vacuum-tube audio equipment in college, and enjoys restoring and modifying audio equipment and test equipment. In his most recent roles as Senior Design Engineer and Lab Calibration Manager, he designed specialized test equipment particular to the aerospace electric power sector.



Amorphous and Powder Metal Alloys

The MetGlas family consists of amorphous Fe-Si-B alloys that also can be annealed to obtain either round-loop or square-loop characteristics. It is widely used in utility pole-top and line transformers to increase efficiency, but its low curie temperature—the temperature at which its maximum flux density starts to rapidly decrease (see Figure 2 from Part 1 of the article, November 2022)—limits its use to applications below 150°C.

The degree of sensitivity to temperature changes depends on the composition of the magnetic alloy, other added metals and the level of impurities. It also depends on the alloy processing from ingot through finished sheet. Metglas 2605HB1M transformer core alloy is used for high-efficiency 50Hz and 60Hz commercial, industrial, and distribution transformers.

Next Month

In the next part of this series, we will discuss transformer insulation materials and winding bobbins.

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Unusual Tubes

Most aficionados of hollow-state electronics—at least most of the people reading this column—are familiar with the tubes most commonly used in classic or contemporary audio amplifiers. In this article, we explore some of the less common tube types used for audio applications.

By
Richard Honeycutt



Photo 1: Many audio devices built in the last 7 decades employed only a few common tube types.

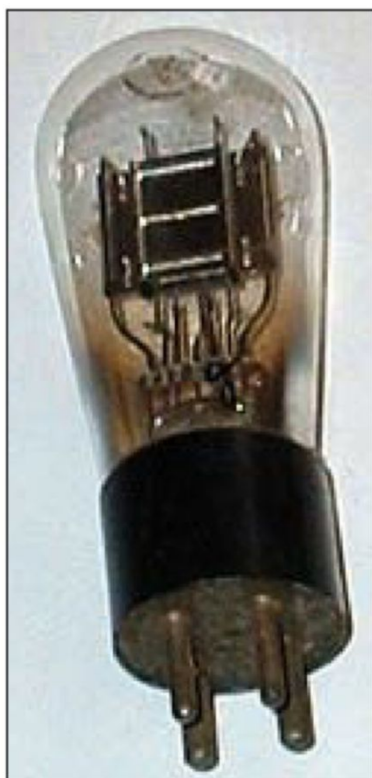


Photo 2: The 01 triode, like many antique tubes, had a 4-pin base.

To begin, most of our readers are probably familiar with preamp triodes (e.g., the 12_7 family, which includes the 12AU7, the 12AX7, the 12AY7, and the 12AT7); preamp pentodes (e.g., the 6AU6); power triodes (e.g., the 2A2, the 811, and the 300B); and power pentodes and beam power tubes such as the 6AQ5, the 6BQ5, the EL34, and the EL84 (**Photo 1**). I have discussed these and other common tubes in past *audioXpress* articles.

Some tube audio fans are also acquainted with some of the antique tubes used in “pre-war” (meaning 1930s and earlier) radios, including the type 01, 35, 56, and 76 triodes; type 80 and 88 rectifiers; and others (**Photo 2**).

A few of our readers might remember the tubes made specifically for TV use, some of which can be re-purposed for audio amps including horizontal deflection output tubes such as the 6BQ6GTB and low-power audio output tubes such as the 6K6, the 6F6, the 6W6, and the 6Y6 (**Photo 3**). And some of you might even recall the 7-pin loctal tubes used in auto radios and a few guitar amps in the 1940s, such as the 7F7 high-mu twin triode and the 7A5 beam power tube (**Photo 4**).

However, there are a number of lesser-known tubes that were made for early radio receivers, telephony, and military uses. Let’s investigate some of these.

Western Electric

Most of our readers probably know that Lee de Forest invented the Audion vacuum triode in the early years of the 20th century. The inventor and most of his contemporaries considered the Audion to be simply a sensitive radio detector. In 1912, about five years after inventing the Audion, de Forest began investigating the basis of its sensitivity as a radio detector: specifically its ability to amplify electrical signals.

Most audiophiles know that Western Electric introduced the classic 300B power triode in 1938. What is not as well-known is that Western Electric bought de Forest’s US patent rights to the Audion in 1914, subsequently developing it into a more useful and reliable device. Demonstrations by Fritz Lowenstein and John Stone in 1912 showed the Audion’s potential as a practical amplifier for telephone repeaters in long-distance transcontinental and intercontinental telephony; hence, the patent purchase. However, the Audion amplifiers had erratic





Photo 3: The 6BQ6GTB horizontal output tube and the 6K6 audio power pentode were used primarily in TVs.

performance and provided insufficient output in their original form. H. D. Arnold of the Bell Telephone Laboratories noticed that attempting to increase the Audion's output by raising the plate voltage caused a blue glow, suggesting that the vacuum was so incomplete as to allow ionization of remaining gas molecules in the tube. By manufacturing the triode using a high-vacuum pump, Arnold improved its performance enough that it became the basis of amplifiers used in transcontinental telephony, which were placed into service in 1914-1915.

Further improvements pioneered by Arnold included the design of a high-emissivity filament made of platinum with a baked-on coating of alkaline-earth oxides. These improvements were incorporated into the Western Electric type 101 vacuum tube (**Photo 5**). As further improvements were made, letters were added to the type number. By 1926, the 101D had been in use in telephone repeaters for 12 years, and it was the most reliable and durable type of tube yet produced. Unlike today's ubiquitous 12AX7 with its μ of 100, the 101D had a μ of 6.2. Complete data for the Western Electric 101D is available online (see Resources).

Although we often think of miniaturization as a relatively recent



Photo 4: A variety of loctal tubes were used in car radios and guitar amps in the 1940s.

phenomenon in electronics, the market introduction of the Western Electric 215A "peanut tube" in the early 1920s was a major advance in reducing the size of electronic devices.

At 2.36" long by 0.63" in diameter, the 215A was not only quite small, but it used a unique type of connector: a bayonet base with four silver contacts (**Photo 6**). In addition to its small size, the 215A boasted a very low filament power; it operated at 0.25A with a nominal 1V filament supply. This was about one-fourth the filament power required by the 101D. The μ was about 5.7. Its microphonic noise—as determined by the then-standard WE test—was -27dBV, about 5dB above that of the 101D. A complete datasheet for the 215A is available online (see Resources).

Much of the design effort at Western Electric Laboratories—renamed Bell Telephone Laboratories (BTL) in 1925—was concerned with high-frequency and high-power tubes for radio broadcasting. But with the introduction of "talkies" in 1927, a market quickly developed for cinema sound systems. A major problem was the hum resulting from the electromagnetic fields of the tube heaters, and BTL's development of the 259B tetrode and 260 triode made much quieter amplifiers possible. The 260 is shown in **Photo 7**, and the



Photo 5 (left): The Western Electric type 101 triode incorporated significant improvements over the Audion.

Photo 6 (center): The Western Electric 215A was the smallest vacuum tube of its time.

Photo 7 (right): BTL's development of the 259B tetrode and the 260 triode (shown here) made much quieter amplifiers possible.

About the Author

Dr. Richard Honeycutt bought a *Popular Electronics* magazine from the rack at a bus station in 1960. It contained an article on building a transistorized audio amplifier, which captured his interest. He followed up by subscribing to several electronics magazines. To avoid the cost of parts, he built up a "junk box" by disassembling trade-in TVs from his uncle's furniture store. His dad bought him the Van Valkenburg, Neville, and Nooger Basic Electronics book set, so he learned about tube circuits as well as transistors. He started repairing electronic devices about 1965, earning his First-Class Commercial FCC license in 1969. He has worked part-time in broadcast radio engineering and audio electronics repair since 1968, in addition to full-time work in acoustical and audio system design, plus a 20-year stint teaching electronics at the college level.



259 is very similar in appearance except for its base, a medium 5-pin bayonet type rather than the 4-pin base of the 262. The 259 tetrode had a μ ranging from 400 to 610, depending upon operating conditions, but as was true for most tetrodes, it required very careful circuit design to avoid parasitic oscillation. The 262 triode's μ was 15. Datasheets for the 259 and 262 can also be found online (see Resources).

With the 1942 introduction of the 6AK5 pentode with its 7-pin miniature base, high-gain pentode voltage amplifier tubes began to take on a look recognizable to today's hollow-state enthusiast. The 6AK5's diameter was only 3mm greater than that of the 215A peanut tube, but its performance was far superior. Having a gain equivalent to a μ of 255, the 6AK5 has been used as a microphone preamplifier and in guitar effects pedals. Many manufacturers have released their own versions of the 6AK5; the datasheet for the RCA version can be found online (see Resources).

Other Manufacturers

During the 1950s and 1960s, RCA and other manufacturers developed a number of new tube types for audio applications. Some of these types never achieved sufficient sales numbers to remain in

production, even though they were included in some commercial products. The 7189, a plug-in replacement for the EL84/6BQ5, was introduced during the 1950s, and was used in some H. H. Scott, Eico, and Fisher amplifiers. Both the 7189 and the EL84 are rated for a maximum plate dissipation of 12W, but the maximum plate voltage of the 7189 is 400V, whereas the EL84 is limited to 300V.

The 7199 is a 9-pin miniature tube combining a medium- μ triode and a sharp-cutoff pentode in the same envelope. It was introduced in 1959. Typically, the tube was used in circuits similar to the 15W, 30W, and 50W amplifiers described in the 1973 *RCA Receiving Tube Manual*: the pentode was used for the input stage, and the triode as a phase inverter. The 7199 was used in a number of American guitar amplifiers, such as the 1961 Gibson Falcon (in the reverb circuit) and many Ampeg models.

The Dynaco ST-70 stereo amplifier introduced in 1959 used a 7199 tube in the driver section of each channel. The 7581 was a plug-in replacement for a 6L6GC beam power tube. Introduced in 1960, this tube has a maximum plate dissipation of 35W, compared to the 6L6GT's 30W. Otherwise, it is substantially identical.

The 7868 power pentode has a Novar base, and thus is not a plug-in replacement for the more common power pentodes or beam power tubes. With its 550V maximum plate voltage and 9W plate dissipation, it is limited to amps outputting about 44W or less (in the push-pull configuration). The 7868 was introduced in 1962.

The 7591 is basically a glass octal twin of the Novar 7868. This tube was commonly used in vintage Gibson and Ampeg guitar amplifiers, and in tube hi-fi equipment manufactured by H. H. Scott, Fisher, and Harman-Kardon.

Even though in most cases the original manufacturers of these tubes no longer produce them, new stock is available from several manufacturers. Data for each tube in this section can be found in the 1973 *RCA Receiving Tube Manual*.

Other Unusual Tubes

Many beam power tubes and power pentodes have European-type numbers as well. For instance, the beam power tubes were first called "kinkless tetrodes" in Europe, and had numbers such as KT66 (6L6GC), KT77 (6CA7/EL34), and KT88 (6550). Electro-Harmonix builds a KT90, (KT90, pocnet.net), which is a plug-in upgrade for a KT88, boasting a maximum plate voltage and dissipation of 750V and 50W versus the 600V and 35W of the KT88. There was a KT100 tube made in China a few years ago, but stock and datasheets are difficult to find. 

Resources

R. Honeycutt, "Trying Different Output Tubes," *audioXpress*, March, 2017, pp. 62-65.

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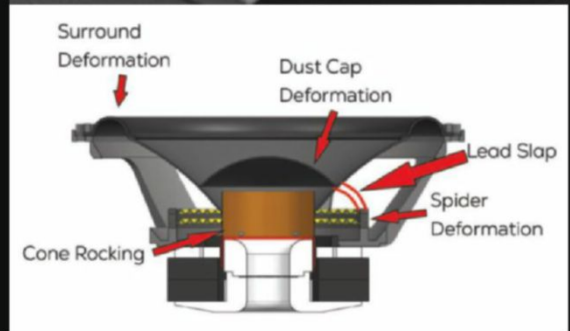
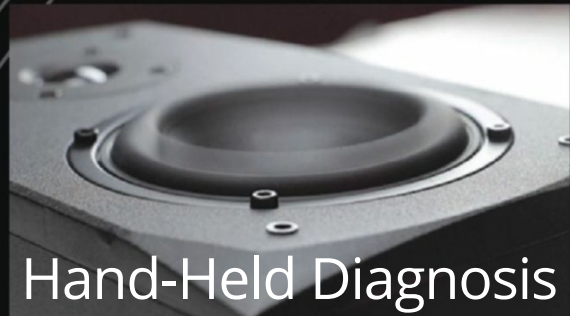
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