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Sure Thing. The Future is Being Invented

There's no innovation happening. Everything has already been invented. Audio today is about cheap plastic speakers, and compressed MP3 files. Music today is only about guys in computers mixing loops and sampling ad nauseam, with artificially auto-tuned voices.

Sound familiar? It doesn't end there... Nothing beats the sound of tubes. Digital will never sound as good as analog. 16-bit at 44.1kHz is good enough for what humans are able to hear.

Really? Did you noticed the iPhone was introduced 10 years ago? Did you notice that no science fiction book, TV series, or film ever anticipated the smartphone? I was watching John Carpenter's *Escape from New York* (1981), which happens in the "distant future" of 1997, and the first sequence in the movie immediately shows people using heavy landline phones with rotary dials. No cellphones. *Blade Runner*, supposed to take place in 2019, still shows Harrison Ford dialing a number to place a call on a "public video phone," with a CRT screen. Well, the same can be said of networks. No sci-fi film anticipated the Internet or social media, while spinning discs and physical media remain at the core of most "futuristic" narratives. And yes, *Back to the Future* shows a scene in 2015 using... a fax!

Yes, I admit I find the Marantz 2270 Stereo Receiver from 1971 a thing of beauty, and I wish I could own an original pair of Klipschorns, but please, look at those speakers from the 1970s, with three or more drivers and a bass port all in the front (flat) baffle. Did they sound great? Yes, but take those woofers, midranges, and tweeters, and compare it with any current driver in terms of measured performance. Let's not even mention the crossovers.

For anyone who loves technology like we do, we wouldn't mind having a mint condition JBL L100 Century pair, a McIntosh MC275, or the amazing Quad 33 and 303 pre- and power amplifiers. But I also admire the average Joe's products, such as the unique plastic design of the Sennheiser HD414's with the yellow foam, the original Sony Walkman, or even the "cheap" Dynaco ST-70, because there's true merit in products that actually improve people's lives while being affordable.

Let's jump forward, and here you are reading *audioXpress* February 2018, featuring the latest advancements on graphene—the "wonder material of the future"—applied to earphones, speakers, and microphones. And there's products shipping in 2018!

This, while Neville Roberts writes about his rediscovery of analog reel-to-reel machines, and how they still sound as amazing at home as in the studio, to the point of being encouraged to spend $25,000 on a used Studer A810 professional machine... and invest in copies of master tapes costing upwards of $400! "Just" to experience the sublime sound quality that is able to take us back in time and into another dimension…

What those glories from the past, or the graphene developments of the future represent are truly amazing engineering feats that make our lives better. And it doesn't have to stop there. Would you have guessed that we would carry supercomputers in our pockets and on our wrists, have truly wireless earbuds that sound amazing and are able to guide us while walking unknown cities, and digital assistants that understand our voices, autonomous electric vehicles, and "wear" universal translation devices? That we have speakers with the ability to perform digital signal processing to optimize a speaker's response in accordance to its acoustics surroundings? Or personalized headphones that make music sound amazing for each individual's hearing, and can be used as hearing enhancers? And that we would have DNA recordings (http://bit.ly/2BnCTXT)? Yet, all that is happening now.

I sincerely have a wish for 2018. Let's understand the lessons and the accomplishments from the past, let's revere true advancements in the evolution of technology in the present, and let's work toward innovations that improve our lives and for the good of humanity.

J. Martins
Editor-in-Chief

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COLUMNISTS

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

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

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

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

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

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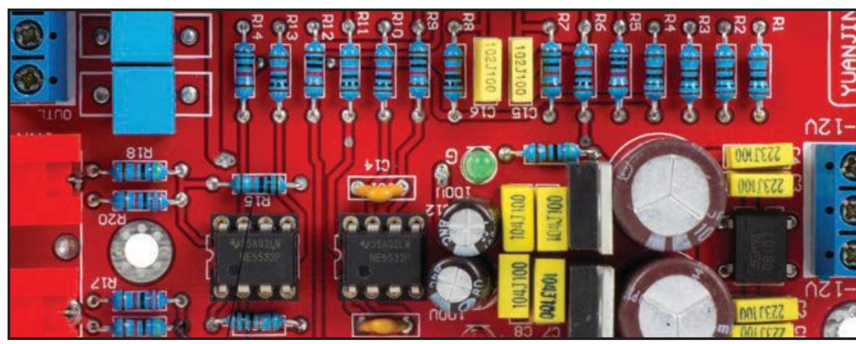

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Realistic Stereo from Headphones

In this article, I explore four DAW plug-ins, which provide added realism when listening to stereo using headphones. Basically, the plug-ins provide an interesting re-creation on headphones of the same auditory cues that reach our ears in the real world, when listening to loudspeakers.

By
Ron Tipton
(United States)

Photo 1: My DIY HRTF head is shown above. The black dot in the ear is the left-side electret microphone with a corresponding one in the right ear. The HRTF (the transfer function) is the math equation relating the input sound to the two-channel output signal. So, it's useful to look at a head photo to help understand what the HRTF does. The DIY head is perhaps not as good as a commercial model, but it's much less expensive—and it does work. A complete article detailing its construction and results will be featured in *audioXpress* April 2018.

Two of the plug-ins offer surround to stereo, but the other two don't have this feature so in this article I evaluated only stereo to stereo. I will review the plug-ins in licensing price order, low to high. As usual, I used the Reaper DAW version 5.40, which is current at the time of writing. I also used Sennheiser HD-280 pro headphones, and a TDL model 444A headphone amplifier to evaluate the plug-ins.

Background Information

If we use headphones to listen to stereo music, we have the left channel going to just our left ear and the right channel going just to our right ear. But when we listen to a pair of loudspeakers, each ear gets a mixture of the left and right channels. The left ear signal heard by the right ear and vice-versa is known as crosstalk or crossfeed. For true binaural sound from the loudspeakers, the crosstalk must be reduced as much as possible.

In my article "Binaural from Two Loudspeakers: Part 1" (*audioXpress*, October 2015), I point out that "Conceptually, crosstalk cancellation is straightforward but the practical problems abound." In this article, I will present the other side of the coin—crosstalk needs to be added to each channel. Of course, a stereo signal already has some crosstalk because even directional microphones share some pickup.

To some extent, musical realism is a personal preference. Having the experience of stereo or surround from headphones with its expanded and more detailed sound stage is appealing to many listeners, as is true binaural from loudspeakers. Again, crosstalk addition is conceptually straightforward but there are practical problems. From my listening tests, I suspect each plug-in author used a different algorithm.

All four plug-ins use "head" modeling. This refers to the Head-Related Transfer Function (HRTF), a model that describes how a sound from a specific point will arrive at the ears. We localize a sound by comparing the loudness and arrival times of the sound as heard by our two ears. We each have a pair of HRTFs: a left ear and a right ear. The modeling works best if we measure our head size, ear size, and head location and then enter this data into the plug-in. But it's also possible to construct an average HRTF that is reasonably accurate for most people (see **Photo 1**).

JB Isone

This VST is free and is for Windows only (see **Photo 2**). It appears to be an early version of TB Isone Pro but it's offered by Jeroen

Breebaart on his website. The HRTF's controls are on the screen's left side with Room Acoustics on the right. Clicking on the knobs displays the setting just under the Isone heading at top center. I adjusted the controls during listening and found a fairly good sound stage with CUE Strength at 75%, head size at 50%, ear size at 30%, distance at 2.5 m, T60 at 0.35 seconds, and room size at 50%. I think CUE Strength is basically a crosstalk control and the User Manual suggests an initial setting of 90% or higher, but I found that increasing it above 75% didn't make any improvement. (Adobe pdf copies of all four User Manuals are included in the Supplementary Material content found on the audioXpress website.)

Photo 2: This is a screenshot of my Reaper DAW hosting the JB Isone plug-in. Although it's free, the stereo enhancing is not as good as the other plug-ins I tested.

Photo 3: This Toneboosters plug-in performed fairly well but I found it difficult to set up because of its many controls.

Enabling the Inter-aural Time Differences (ITD) switch illuminates its yellow bar. ITD means different internal delay times for each ear, which is probably the more realistic.

Loudspeaker-to-listener distance, reverberation time (Room T60), and room size should be adjusted for the most pleasing sound. They will often need readjusting with changes in the type of music being played. Channel mode and cabinet type are also the listener's choice and it's interesting to experiment.

I listened to several types of music: classical, 1950s, popular, and jazz and I used the same tracks for all the testing. Compared to the other plug-ins I found the stereo effects to be minimal. But it is free and probably somewhat out of date: The User Manual is dated 2009.

TB Isone

Photo 3 shows a screenshot of this VST plug-in using the HiFi Speaker preset. It is available for both Windows and Mac but no longer as an individual plug-in. Toneboosters has recently changed its marketing policy so Isone is now bundled with nine other plug-ins in its BusTools Version 3 package. However, the licensing price is only €40 (about $47.80), which is just about twice the previous cost of a single plug-in.

The Preset button (shown in Photo 3) pops down a speaker list that also includes Flat, Small monitor, Flat panel, Mono speaker, and Monitors A, B, and C with their frequency response curves shown in the JB Isone User Manual. Also, speaker angle is measured from the center to each speaker rather than speaker separation.

As you can see from the screenshot, there are quite a few knobs to adjust. The User Manual gives some hints about adjusting them, but I discovered some interactions that does not make it easier. With the settings as shown, I got a pretty good soundstage but it lacked the "presence" that I heard from CanOpener Studio. Isone sounded like I was hearing it through a thin wall or a curtain. Good sound but not great sound.

CanOpener Studio

This Version 3 plug-in from GoodHertz is available for both 64-bit Mac OS equal or greater than 10.7 and 64-bit Windows 7 and higher (see **Photo 4**). It is priced at $49 with a free upgrade if you have a licensed copy of version 1 or 2. The installation program asks what to install. I chose both VST2 and VST3 for Windows but using VST3 is recommended for best performance. (For Mac, it's 64-bit AU or AAX with a 64-bit DAW.)

As you can see from the photo, the left-side pane controls are rather simple compared to the Isone's, just Crossfeed Amount and Loudspeaker Angle. The User Manual says an amount of 150% is the most realistic but I found it best to re-optimize depending on the music type. An angle of 60° is the default but I found 40° or 45° produced a "tighter" sound stage. I didn't use the equalization and just left the sliders at 0.

The center pane is a stereo spectroscope. The horizontal axis is the left and right stereo field width and the vertical axis is frequency. The size and brightness of the circles is loudness. Besides being interesting to look at, it's also useful. Quoting the User Manual: "If the scope shows a wide, dispersed stereo field (especially at the low end), then you might want to increase the crossfeed amount. If the scope is showing a narrow, straight-lined stereo field (especially at the low end), then less crossfeed is probably appropriate."

In the right-hand pane, there are buttons for Mono, Flip L/R, Polarity (inverts the right channel phase), and Dim (reduces output volume by the number in DimLevel.) Below these buttons are the gain (volume) control and an output level meter. Clicking the three white dots on the right screen edge loads the Advanced menu. "Most realistic" is the default and is usually the best choice, but the other choices are described in the User Manual.

I love this plug-in's sound stage and it has great presence, as if the performance were taking place right in front of me—very realistic.

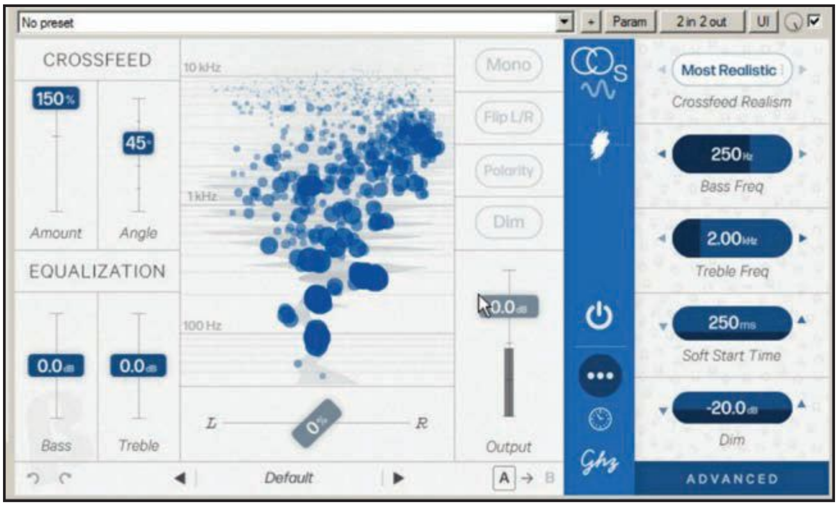


Photo 4: This screenshot shows the simplicity of CanOpener Studio. It's easy to set up and I am impressed by the quality of its soundstage and presence.

Waves Nx

Priced at $99 and available for both Windows and Mac, this Windows VST plug-in has only a few controls (see **Photo 5**). Although I have one, I didn't use the head tracker because it would not have added anything to the review. I did measure my head circumference and ear-to-ear distance around the back of my head. They are somewhat different from the default values and the User Manual encourages making and entering the actual measurements.

For the Headphone EQ line, I turned it ON and selected my headphone model from the drop-down menu. I accepted the default values for room ambience and speaker position.

The sound stage was very good and the presence was okay but not quite as good, I think, as CanOpener Studio. The Room Ambience control affects the presence. As I increased the value, it seemed to increase my sense of distance from the source.

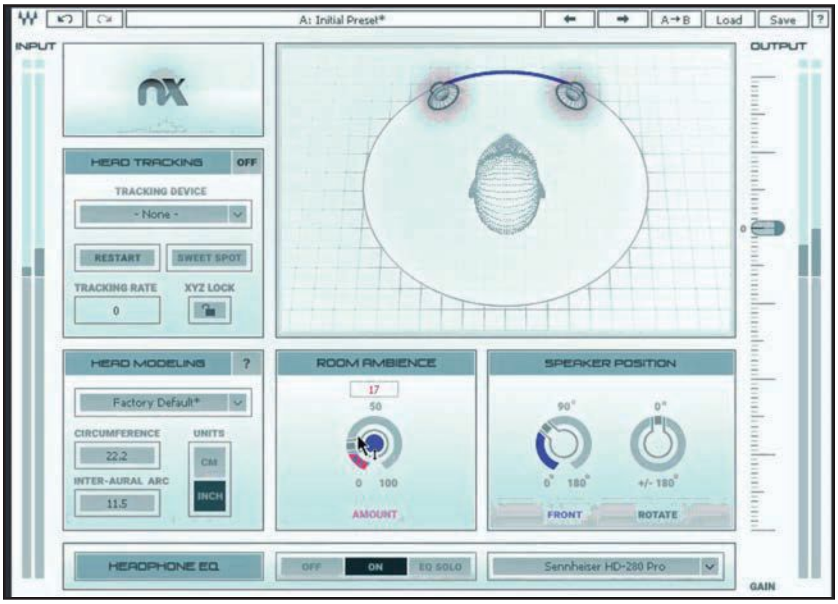


Photo 5: The Waves Nx plug-in is the only one that uses actual head measurements for its HRTF calculations and the only one to tailor its output to a specific headphone model. The stereo enhancing is close to CanOpener Studio. (Photo colors reversed to avoid the all-black background.)

Project Files

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

Sources

JB-Isone
Jeroen Breebaart | www.vst4free.com/free_vst.php?id=2069

TB-Isone
Toneboosters | www.toneboosters.com

CanOpener Studio
GoodHertz | http://goodhertz.co

Waves Nx
Waves Audio | www.waves.com

The Reel Thing

With the revival of interest in vinyl, many people are realizing how fantastic this analog medium can sound, even when compared with "studio master" quality digital formats. Now it seems that another analog format is coming back into style—open reel tapes.

Photo 1: Professional 1/4" stereo recorders, such as the Studer A810 shown here, usually record on two tracks and in one direction only.

By
Neville Roberts
(United Kingdom)

It was only about 10 years ago that we started to see a gradual increase in vinyl sales, and initially this was dismissed as a passing fad. Since many vinyl buyers were too young to remember vinyl as a primary music form, the renewed interest in this format was simply considered to be part of a revival of retro style. Nevertheless, more and more audio enthusiasts discovered the quality of sound produced by good quality LPs exceeded that of even the higher resolution digital formats, such as the 24 bit, 192 kHz PCM files often used in recording studios to record the master files.

Of course, analog signals do have their own issues, such as susceptibility to noise and certain types of distortion. However, the analog format certainly gives the sound a greater sense of realism and clarity, and this is particularly evident when playing music on high-end audio systems. There are a number of reasons for improved sound, but many attribute the positive qualities of analog to the fact that an analog signal has a theoretically infinite resolution—something a digital signal can never attain.

I must confess to thinking that a 180 gram, direct-cut LP, one-take recording of a live concert session was about as good as it gets—until I heard the sound from a half-track stereo, 1/2" master tape of the same session recorded at 30 IPS. It was evident that the tape recording was significantly better, so I started looking into why analog tape offers such superior sound quality compared with other mediums.

Analog Magnetic Recording

The idea of recording an analog audio signal as a magnetic pattern dates back to the late 1890s. Valdemar Poulsen, a Danish engineer, came up with the idea of magnetically recording a signal on length of piano wire wrapped around a cylinder. The Telegraphone, as it was called, was a hand-cranked device used for telephone recording purposes. In those days before amplification, the benefit of the Telegraphone over the phonograph was its ability to record directly from the electrical signal transmitted via the telephone lines. Later, the wire was fed from a supply spool, passed across a static record/playback head, and then wound onto a take-up spool. This constituted the first wire recorder and it was used as a Dictaphone in the 1920s as an alternative to the acoustic phonograph.

Initially, the two poles of the record/playback head were arranged to be on each side of the wire as it passed through, so the wire was magnetized transversely to the direction of travel. One of the problems with this approach was that when, inevitably, the wire twisted during playback, there were times when the direction of magnetization of the wire was at right angles to the two poles of the head. As a result, no signal was picked up and consequently there were long periods of dropout. Later, the two poles of the head were repositioned to be on the same side of the wire so that the wire was magnetized along its length. This system is still in use today with tape recorders.

Apart from the issue of twisting, steel wire as a recording medium had other limitations. For example, it required a relatively high wire-to-head speed of around 24 IPS, so a one-hour spool contained approximately 2,200 m of wire. In order to hold this much wire, the wire itself had to be very fine and often became tangled or kinked. This, coupled with the need to improve the recording's sound quality and reduce the speed of the media, led to the development of magnetic tape as a recording medium in the 1930s, using a special iron oxide powder coated onto a plastic tape.

The Rise and Fall and Rise of Analog Tape

In the mid-1950s, hi-fi equipment for home use really took off, and by the late 1960s an open-reel recorder was an essential component of any hi-fi system. These recorders could take cine spools of up to 7" diameter with tape speeds of 7.5 IPS for hi-fi and 3.75 IPS as standard speeds for music reproduction.

Professional tape recorders in the studio could accommodate NAB spools of 10.5" diameter or more, recorded at speeds of 15 IPS or 30 IPS. Of course, one big advantage of tape recording for professional use is that the tape can be physically edited and it soon became the standard way of making professional recordings. These recordings were used to cut a master lacquer that was then used to produce an LP. A second replay head, which was positioned ahead of the main playback head, enabled LP masters to be cut with a variable pitch of the groove. This made it possible to pack the quieter music closer together, which enabled more music to be recorded on one side of an LP.

Professional machines used 1/4" or 1/2" tapes, with wider tapes being used for multi-track recordings. Professional stereo recorders (e.g., the Studer A810) usually record on two tracks and in one direction only (see **Photo 1**). Domestic machines (e.g., the famous Sony TC377) use 1/4" tape, but for stereo, four tracks are used with tracks 1 and 3 (numbered from the top) for the left and right channels, respectively (see **Photo 2**). The tape can then be turned over and used again, with track 4 on top lining up with the left channel of the tape head, and track 2 lining up with the right channel of

a)

b)

Photo 2: Domestic machines, such as the famous Sony TC377 reel-to-reel tape recorder shown horizontally (a) and vertically (b), use 1/4" tape, but for stereo, four tracks are used with tracks 1 and 3 (numbered from the top) for the left and right channels, respectively.

the head. Professional tapes are usually stored "tail out," which means that the tape must be re-wound before playing.

Philips introduced the world to the cassette tape recorder in 1963 and with a tape speed of 1⅞ IPS, it was only suitable for voice recording. However, Nakamichi applied its experience of open-reel machines to cassette systems, and by 1973 the company was producing stereo cassette decks with such high-quality reproduction that they offered a serious challenge to the open-reel machine. The Nakamichi 700 and 1000 machines with their three heads and dual capstan drive were regarded as two of the finest cassette recorders of the mid-1970s, but few audiophiles could afford them. Nakamichi responded to the potential demand by releasing less expensive two-head models (e.g., the Nakamichi 500 and 600).

Further developments in the oxide coatings of cassette tapes and the implementation of noise-reduction systems (e.g., Dolby) established the cassette recorder as an essential element of an audio system by the mid-1970s. As a result, by the early 1980s, the open-reel recorder was truly a "legacy format."

At the start of the digital era in the 1990s, many enthusiasts turned to Digital Audio Tape (DAT). Then with the availability of personal computers, most people moved away from cassette and open-reel analog tape and started recording their own CDs and digital audio files. As a consequence, analog tape as a media faded from use in both professional and domestic settings.

However, now that people are realizing that vinyl has a place in an audio system due to its excellent audio quality, many audio enthusiasts are also digging out their tape collections and realizing that this old analog format still has a lot to offer as well. With the renewed interest in analog formats, spearheaded by the start of the vinyl revival in 2007, we are now seeing a growing number of second- or third-generation master tape copies being offered for sale by a number of companies across the globe.

Of course, the one main difference between the vinyl revival and the tape revival is the increased cost of producing the recorded media, and the cost of the equipment needed to play the master tapes. First, at a cost of around 300 Euros (approximately $355) per 10.5" NAB tape, it is much more expensive to produce master tape copies than an LP. Second, many domestic tape recorders are unlikely to be able to play the professional tapes at speeds of 15 IPS, which necessitates the purchase of a suitably refurbished high-end domestic or ex-professional machine. Therefore, the renewed interest in open-reel tape is focused on audio enthusiasts who are prepared to pay the price for the machines and the media. Many feel, however, that this is worthwhile as the quality of the master tape is even better than that of an LP.

Photo 3: The Tape Project in the US is focused on re-releasing copies of the master tapes of original analog recordings.

Current Recorded Tape Suppliers

Over the past couple of years, we have been seeing an increase in the number of companies

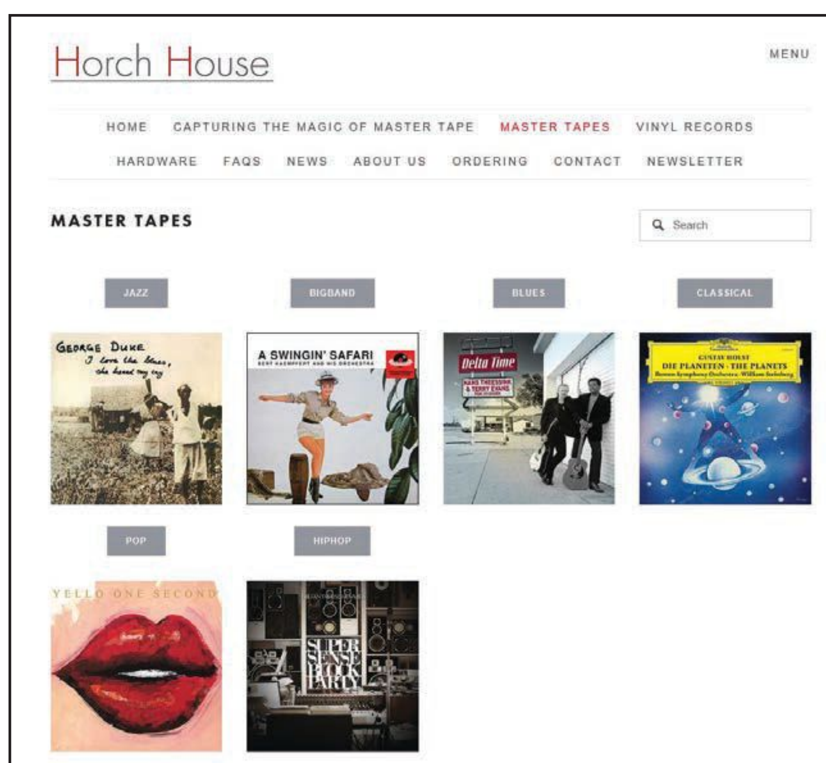


Photo 4: Horch House in Slovakia is another company focused on re-releasing copies of the master tapes of original analog recordings.

Photo 5: Chasing The Dragon in the UK is one of a few companies that are now producing modern recordings on analog tape.

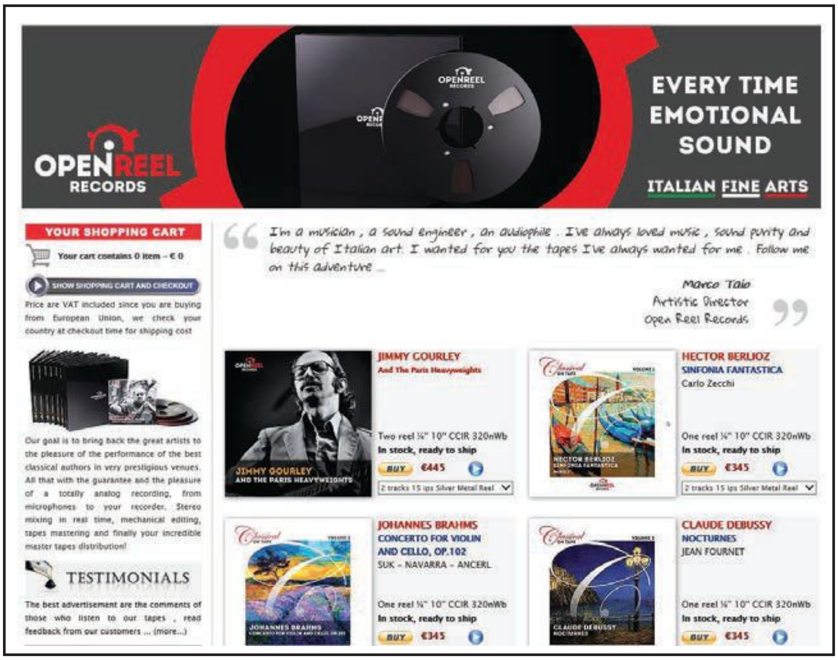


Photo 6: A fantastic demo tape from Marco Taio of Open Reel Records is truly breathtaking.

About the Author

Neville Roberts, a graduate in Physics, is a Chartered Scientist, Chartered Engineer, Chartered Physicist, and a Fellow of the Institution of Engineering and Technology. He is a retired Director of the UK National Health Service (NHS) and has also worked for the UK Ministry of Defence and in the private sector. He is a regular contributor to a UK audio magazine, *Hi-Fi Choice*, and has a particular interest in tube audio design and is a vinyl devotee. Apart from being an audio enthusiast, he enjoys a wide range of music including classical, especially baroque, light orchestral and jazz. He is a keen photographer and enjoys growing both tropical and hardy orchids at his home near Bournemouth in Dorset. He is also on the Committee of the Bournemouth Orchid Society.

(currently around 20) that are producing music on tape. Some companies, such as The Tape Project in the US (see **Photo 3**) and Horch House in Slovakia, focus on re-releasing duplicates of the master tapes of original analog recordings (see **Photo 4**). These companies have acquired the rights and the original master tapes of some famous recordings, ranging from Leopold Stokowski's *Rhapsodies* through to the legendary big band music of Bert Kaempfert and the like. Other companies, such as the Swedish Opus 3 Analogue Master Tapes, Open Reel Records of Italy, and Chasing The Dragon in the UK, are now producing modern recordings that have been made on analog tape (see **Photo 5**).

In parallel with this, there are companies that are starting to manufacture open-reel machines again, albeit aimed at the high-end market. The aforementioned Horch House is working with the Revox Group on what they call Project R2R to produce a new open-reel machine. A German company, Ballfinger, has also launched its new open-reel tape machine called the Tonbandmaschine M 063.

Having recently acquired a superb Studer A810 machine, which is classed as a "professional portable" tape recorder (though weighing in at 68 lb., it's only just "portable"), I tried some of the tapes, playing them through my audio system.

A fantastic demo tape from Marco Taio of Open Reel Records (see **Photo 6**) is truly breathtaking, and the recording of Venetian contralto Sara Mingardo singing Vivaldi's "Nisi Dominus RV608" really leaves me spellbound. Her singing, combined with the splendid playing of Accademia degli Astrusi, is absolute perfection, and the ambience of the recording venue (Chiesa di Santa Maria della Vita in Bologna) is perfectly captured in this recording. Turning to some jazz, the excerpt of Arrigo Cappelletti playing Thelonious Monk's "Round Midnight" on the piano is compelling. The piano, which is a notoriously difficult instrument to record accurately, is superbly captured by the Schoeps microphones.

I have also been able to compare a copy of Chasing The Dragon's direct-cut vinyl "Espãna" with one of the company's new master tape copies that was made from the 30IPS 1/2" tape. The tape was recorded from the same audio feed that was fed to the vinyl cutting lathe, and this provided a rare opportunity to compare the tape with the very best that vinyl has to offer.

In an A/B test comparing the LP with the tape, the tape took the sound quality to a noticeably higher level. This tape recording conveyed a bit more of the magic of hearing the concert live—as indeed I did, having been present in the actual studio during this recording. I could almost smell the rosin that is used on the bow strings of the stringed instruments!

Photo 7: I had the opportunity to compare an original Studio Master from the 1960s and modern master tapes from Chasing the Dragon and Open Reel Records.

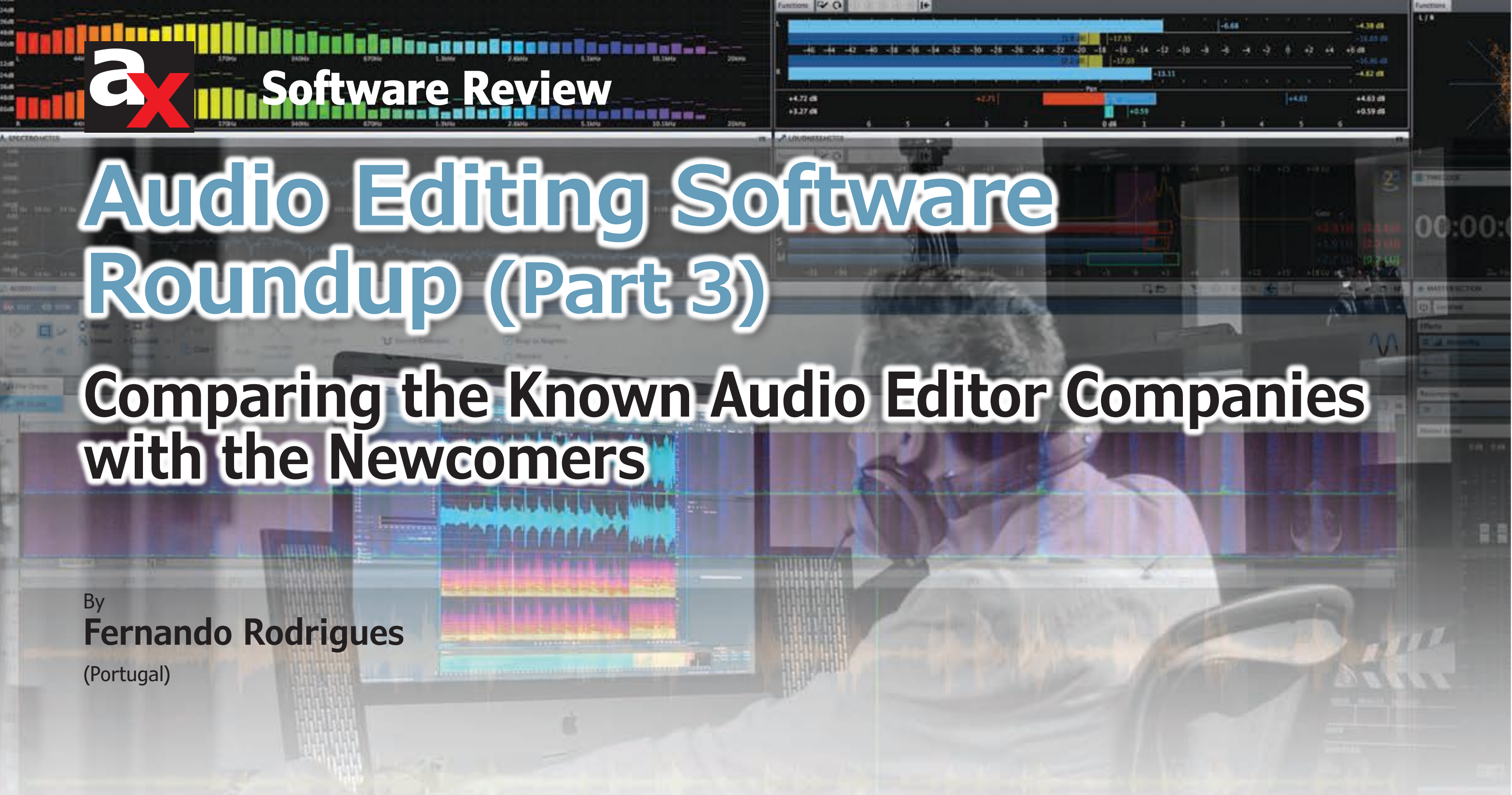
Playing a set of original master tapes of light music recorded in Germany in the 1960s and featuring the Manfred Minnich orchestra, I once again find myself captivated by the incredible fullness, clarity and presence of the recordings (see **Photo 7**). These are not second-generation copies, they are the original tapes of the actual recordings. It is interesting to note that so much of the magic of the performance is preserved on the recording, which was probably made using the high-quality Neumann tube microphones, similar to the famous Neumann U47 microphones used by Chasing The Dragon on many of their recordings.

Clearly, the quality of reproduction is not just down to easily quantifiable parameters, such as frequency response, distortion, or noise. Surely, there must be other as yet undetermined factors that contribute to the high quality that analog offers over other forms of technology.

Indeed, it has been noted by some people in Japan that audio enthusiasts can appreciate high-quality reproduction independent of good hearing. Some elderly audiophiles have confessed that although clinically they have quite limited hearing, they can still easily appreciate high-quality reproductions. In blind listening tests, they are able to appreciate the differences between live performances, high-quality audiophile recordings played on high-end equipment, and conventional commercial recordings played on domestic audio equipment.

Conclusion

Having heard for myself all that tape has to offer, I can now understand why companies are starting to invest again in analog tape recording and replay technology and manufacture of the media. Good quality examples of professional tape equipment are becoming increasingly hard to find and prices are rising rapidly. I personally believe that the growth of interest in the tape medium will be considerably faster than it was for vinyl, as the vinyl revival has already proved that high-quality analog can seriously challenge even the best of the modern formats. In my opinion, it will only be cost factors that limit the appeal of open reels.



Audio Editing Software Roundup (Part 3)

Comparing the Known Audio Editor Companies with the Newcomers

By
Fernando Rodrigues
(Portugal)

Fernando Rodrigues continues to examine some of the available software audio editors currently on the market. In this third article, we look at some established names, such as Sound Forge, now under the Magix brand, followed by Audacity, one of the most popular free audio editors. He also checks out Gold Wave, one of the oldest audio editors available, and an interesting newcomer, Ivosight Soundop.

Magix Sound Forge

Sound Forge was the first product of Sonic Foundry. Later, Sonic Foundry was bought by Sony, which recently sold its entire software division to German company Magix. Consequently, Magix, which also bought Samplitude from SEK'D some years ago, has been integrating Sound Forge into its own portfolio, and Sound Forge became part of the bundle of software included in the Samplitude Pro X Suite.

It's a commercial application, and still 32-bit only, which is a pity, and perhaps the major con—at least until an expected update is released. In an era where more and more products (even plug-ins) are 64-bit only, it is urgent that Magix launches a 64-bit version of Sound Forge. We hope to see this done soon (Magix has already announced a new version of Sound Forge for Windows to be launched in 2018).

Sound Forge for Mac was recently updated to version 3.0, but it still isn't on par with the Windows version in terms of features, although it seems to be closer now. In Windows, there are two versions available—Sound Forge Pro and Sound Forge Audio Studio.

Sound Forge can handle files with sample rates up to 192 kHz and a resolution up to 64-bit floating in the pro version (32-bit floating in the light version). In terms of number of channels, it can handle up to 32 channels.

The main strengths of the program reside in the fact that, being a relatively "old" audio editor, it keeps some features now absent in most of its competitors, such as the ability to deal with external samplers, and a dedicated loop editing window with GUI (for those to whom this is important, of course). As a matter of fact, when it comes to creating and editing loops, Sound Forge is our tool of choice in Windows, and has little competition, even in the Mac platform.

VST support is a little convoluted, and the program is not very tolerant. Nevertheless, if we keep the VstPlugIns folder relatively clean (we did that by creating a special folder), it handles them better. Unfortunately, we have to use a workaround to handle the scanned plug-ins, otherwise we would be presented with an unsorted and unfriendly window. There is a window to stack plug-ins and create multi-processing, but we would prefer a more elegant method (e.g., the Master Channel in Wavelab or in DSP Quattro). Sound Forge is also one of the few programs to still support Active-X plug-ins.

Sound Forge comes with many DSP processing tools, among them Acoustic Mirror, a great convolution plug-in (Sonic Foundry was a pioneer in the use of the convolution technique), as well as Noise Reduction (it was once one of the best noise reduction tools, but that was several years ago). Perhaps the most important special feature is the Zplane Élastique Pro time stretch and pitch shift plug-in that's also included. Other special features worth noting are Crash Recovery, Acid Loop creation tools, as well as Volume and Pan envelopes, MIDI timecode, and Timestamp recording.

Sound Forge features many analysis tools, and the VU indicators are very good, as is the Goniometer (fast and precise). The spectrum analysis is also very precise, although not real time. Unfortunately, Sound Forge has no spectrum editing capabilities Magix sells Spectra Layers, another audio editor, for that specific task. Although Spectra Layers is a fantastic editor, with great spectrum features, in our opinion, it doesn't make sense to have two separate programs for what is basically the same job—audio editing—and the price tag of Sound Forge would justify having these features integrated.

In terms of reading and writing capabilities, Sound Forge Pro is very comprehensive, as would be expected, considering its age and tradition: AA3, AC3, AIFF, ATRAC, AU, AVI, DIG, DLS, FLAC, FRG, GIG, IVC, M1A, M1P, M2A, M2P, M2T, MMV, MOV, MSV, MP1, MP2, MP3, MP4, MPA, MPEG, MPEG-1, MPEG-2 video, Sony MXF, Ogg Vorbis, PCA, RAW, SF2, VOX, W64, WAV, WMA, and WMV.

It is also one of the few audio editors that handles video files directly (a feature only present in the Windows version), enabling us to edit the audio track still in the video, and save the file directly. Formats supported are AVI, WMV, MPEG-1, and MPEG-2. Almost all audio editors only deal with video files (those that do) through import/export of audio. Another important add-on is CD Architect, another application especially tailored to prepare tracks, create metadata (PQ code, CD text, etc.) and burn CDs directly (DAO mastering), according to the Red Book standard. It includes high-quality resampling technology licensed from iZotope.

Despite some shortcomings, Sound Forge is still our platform of choice in Windows for sample editing and mangling, and loop creation.

Sound Forge for Mac is not as complete as its Windows sibling, and we did not try the newest Mac version. Nevertheless, some features are not present in the Mac version, although it comes with iZotope's RX Elements and Ozone Elements. Also, licenses are platform specific, therefore, a user who owns a license for Windows must purchase another license to use the Mac version—something we always found strange, and definitely not what is the common practice among vendors (as a matter of fact, we don't know any other vendor that does this—any other cross platform program uses the same license). This is another con, and again something Magix should correct, in our opinion.

Right before going to print, Magix announced a new version of Sound Forge Audio Studio, working at 64-bit, and with some new and interesting features. Sound Forge Pro, though, remains at the same version, but it indicates we will have a new update launching soon.

Magix Sound Forge: www.magix-audio.com/int/sound-forge
Sound Forge Pro 11 (Windows 7 and later): $ 399.95
Sound Forge Audio Studio 12 (Windows 7 x64 and later): $ 59.99
Sound Forge Pro Mac (OS X 10.9 and later): $ 557.00

Audacity

Audacity is a free cross-platform audio editor (Windows, Mac OS X, and Linux). It's 32-bit only, and one of the oldest, most well-known and widespread audio editors available. In Mac, it was Universal Binary (up to OS X 10.5—a version is still available), but now supports only Intel OS X (10.6 and later). Up to the launch of version 2.2 OS X 10.12 Sierra was only partially supported, but it is now fully supported. In Windows, Version 2.2 also ceased support for Windows XP, therefore, the program now supports Windows Vista, 7, 8, and 10. Installation packages for Audacity Linux are provided by many GNU/Linux and Unix-like distributions. Although it offers a good feature list, there are some quirks that we noticed during use.

However, the program is well supported, and just before we went to print, we received notice of the launch of the already mentioned version 2.2 and even an update numbered 2.2.1, with some new features and many bug corrections. One of the most visible changes is the GUI, which gained new themes (skins), and also customization capabilities. The menus were also reorganized, and compatibility with Mac OS X 10.12 (Sierra) was finally completed, after solving the problems still existing in the previous version.

Audacity supports several tracks. For recording, the number of tracks is tied to the hardware in use, but it can open and mix a large number of tracks. Multiple clips are allowed per track. Sample rates go up to 192 kHz when recording, and up to 384 kHz when editing. Audio resolution includes 16-bit, 24-bit, and 32-bit floating.

File support is very dependent on third-party libraries. Natively, it supports WAV, AIFF, AU, FLAC, and Ogg Vorbis files. Through libsndfile library, it supports GSM 6.10, 32-bit and 64-bit float WAV, RF64, as well as a-Law and µ-Law raw files. Through FFmpeg library, it supports AC3, M4A/M4R (AAC), and WMA. This also enables users to import audio from video. The libmad library enables users to import MPEG audio (including MP2 and MP3 files), but to export MP3, we need to install the well-known LAME encoder. This library dependency is common with Linux software, but Mac/Windows users might find it awkward having to find, download, and install these libraries separately. Finally, with the launch of version 2.2, Audacity also gained the ability to import and play MIDI files. These files are played through the internal synthesizer. The manual mentions a software synthesizer, but it is not clear if we can use any virtual instrument besides the system one. However, we managed to use one of our external multitimbral modules connected through a MIDI interface. We loaded a multitrack MIDI file, and it played it correctly. So, what's this for? According to the release notes, this feature can be used to allow comparisons with recorded audio.

Audacity has a somewhat convoluted way to deal with plug-ins. This may be related to the need to support several different operating systems and many formats: It supports Nyquist, LADSPA, LV2, VAMP, VST, and Audio Units, with real-time preview. We confess, we weren't even familiar with some of these formats. Management of plug-ins is done through the Plug-in Manager, which handles plug-in installation and addition/removal of effects and generators from the menus. The problem is that this Plug-in Manager is not very transparent, and management is complicated when we have many plug-ins. We were struggling to have some plug-ins recognized, and decided to create a new VstPlugIns folder inside the Audacity folder. Then we managed to have the same plug-ins twice in our list. We deleted the Audacity folder, and now, although the plug-ins appear in the recognized list, they became Disabled, with no apparent reason. Somehow, after installing the updated version 2.2 the list appears cleansed. We were able to enable more modern plug-ins, such as Ozone 8 and Neutron 2, but every time we tried to instantiate Ozone 8, it crashed. We had to choose the previous Ozone 7 instead.

The way Audacity deals with VST plug-ins still has work to be done. Besides the numerous incompatibilities, anyone with a reasonable number of plug-ins (and users can easily have hundreds of them these days…) will struggle to simply maintain a list of usable plug-ins. A true challenge to our good will and patience. Also, we have to choose the plug-in from a long vertical list, because they don't show inside nested folders and there is no way to organize the list.

Anyway, Audacity already comes with many plug-ins. Included processors are Equalization, Distortion, Echo, Limiter, Paultratch (extreme stretch), Phaser, Reverb, Reverse, Truncate Silence, and Wah. The program includes Batch Processing and Plug-In Chains. Actually, these are basically the same thing in Audacity, since everything is done through Chains. Chains are a kind of "scripts" that the user creates, and where several processes are defined, to be applied sequentially to one or several files. We can create one of these through the menu File > Edit Chains, but to apply them we will use File > Apply Chain. There is one "Chain" already pre-programmed, for MP3 Conversion. The rest is up to the user.

Although creating a chain of plug-ins is pretty straightforward, the method is not the most intuitive, quite the opposite. We found it somehow "geek-oriented," and we didn't find a way to dynamically alter the plug-in settings. It seems that we define a set of controls for each plug-in and then the chain is applied "as is." So, the only way to use plug-ins the way we are used to is one at a time.

As mentioned, Audacity also allows Time Compression/Expansion and Pitch Shifting, using the standard algorithm methods. It also has a Noise Reduction tool to remove static, hiss, hum, and other background noises.

There are no loop tools whatsoever. This audio editor seems more concentrated in the audio conversions (of sample rates, formats, etc.) and restoration. It allows us to edit at sample level with the Draw Tool, to alter individual sample points.

We also have some spectral editing tools, like the ability to filter a frequency selection created in Spectrogram view or Spectral Selection Toolbar using spectral edit effects.

Analysis tools include Spectrogram, Plot Spectrum (no real-time), and Contrast Analysis for analyzing average RMS volume differences between foreground speech and background music. Other special features include automatic sample rate and format conversion of different tracks in real time, and track labeling with selectable Sync-Lock Tracks feature for keeping tracks and labels synchronized.

Audacity: www.audacityteam.org
Available for Windows (from Vista onward); Mac OS X (10.6 and later, current version is Intel only); GNU/Linux (several builds available): Free

GoldWave

GoldWave is one of the most ancient audio editors still available (we remember it from Windows 98, but maybe even earlier). Currently, it supports Windows 7 x64 or later (it is now 64-bit only).

As it happens with many audio editors, it supports multi-channel files (up to 48 tracks). It also supports high sample-rates, although slightly less than some competitors (the maximum sample rate is the strange figure of 352.8 kHz, or 8 × 44.1 kHz, meaning that the 48 kHz ratio that became standard in many others isn't the basis here). In terms of audio resolution, we have the now standard 32-bit floating.

Regarding formats recognized and supported, it reflects the fact that it's an historic application, and it's one of the most comprehensive: WAV, MP3, Extended Audio (XAC), AIFF, AIFC, MOV, AC3, IFF (Amiga), AU (Sun), VOC, Raw, SDS (Sample Dump Standard), Sample Vision (SMP), VOX, MAT, Numerical Text (TXT), MP4, M4A, AAC, AVI, FLAC, Ogg Vorbis, Opus, WMA, and APE (Monkey Audio). It's one of the few in this software roundup to support the Monkey Audio APE format natively (a lesser known but very good lossless compressed audio format), as well as the only one to support the now exotic Sample Vision format (Sample Vision was an audio editor from Turtle Beach, now Voyetra).

Regarding plug-in support, it supports DirectX and VST2 plug-ins, but only through its own wrapper. This contributes to making the plug-in support rather convoluted, and also creates some GUI sluggishness. An example of that is the wrapper, also a 32-bit bridge, which means we have to work with 32-bit plug-ins (strangely, despite being a 64-bit application, GoldWave doesn't support 64-bit plug-ins), but, although we can select a full directory to have it scanned, by default we are forced to answer to a dialog box asking if we want to use THAT plug-in. This happens because there is an option ticked that forces that confirmation. To avoid this, we have to untick that box. After scanning the entire VstPlugIns folder, we have to wait for the program to update the list of effects. Then, we must restart GoldWave, and only after that, the VST plug-ins are available to work. However, the program expects us to mainly work with them individually. We may work with plug-ins within chains (this is a supported feature) but chains are not easily created, as we'll see. So, the best way is to use some third-party chaining host (e.g., Plug and Mix Chainer). However, we noticed another problem: Maybe due to the bridging, or for some other reason, the GUIs of the plug-ins have very poor performance. So, real-time indicators, such as those present on Ozone, T-Racks, Chainer, or even in many other plug-ins (e.g., the ones from FabFilter) are basically useless inside GoldWave. It's a pity that GoldWave doesn't have a better support for plug-ins, because in many other aspects, it is a truly advanced piece of software. In our opinion, and in the current state, the use of VST plug-ins inside GoldWave is problematic, to say the least. After some intense dialog with the developer, it was recognized that work needs to be done, and will be done in the next months.

That said, the included plug-ins (GoldWave proprietary format) are very good, have good GUIs and some of them are even creatively excellent. Just the included plug-ins would justify the price of GoldWave. With these and the included analysis tools, this could be a good competitor in Windows, if it wasn't for the mentioned quirks. Anyway, it's still one excellent audio editor, and one of the most comprehensive in terms of included DSP processing.

There are no loop tools and no sampler support, despite being an old audio editor. This is another program that seems more concentrated in the audio conversions and restoration—no CD mastering and burning either. However, batch processing is available with a dedicated window, and we also have Plug-in FX chains (as we saw, it's basically the only way to access VST plug-ins) and a multimixer.

Analysis tools are great. Many real-time analysis tools, including spectrum bars, 2D Spectrograph, and VU meters. We also have real-time spectrum view (2D and 3D) and a spectrum filter. The analyzers are showed in the control window, which is very customizable. Unfortunately, our third-party plug-in analyzers (e.g., iZotope Insight) showed a sluggish graphical performance, as we said, which made them basically useless.

The usual features of time compression/expansion and pitch shifting are also present, again through proprietary algorithms. Good, but no special features either. Other features worth mentioning include the File Merger (to mix audio files), and the Expression Evaluation Tool (a tool to create audio through math expressions).

Being focused on audio restoration, it's no surprise that the software includes standard restoration functions, like Pop and Click remover, Smoother (to reduce hiss), and Noise Reduction.

GoldWave: www.goldwave.com/goldwave.php
Windows 7 and later—64-bit only: $45

About the Author

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

Ivosight Soundop

Ivosight Soundop is another commercial audio editor that is just starting out, and much like ReSample, is struggling to find its own place in the market. Again, as is ReSample, it is a "work in progress" and some things have already been corrected through our dialog with the programmers, although more corrections and improvements are yet to come.

This program, apparently, was born using Adobe Audition as a paradigm. Like Audition, Soundop has a multitrack editing window (called Mix Space), with its own special file format, track editor, mixer (complete with send and aux busses), a flexible routing and side-chain support, and multiple automation lanes. The file formats are different, and for the aim of this roundup, what interests us is the other window, called Editor.

We found the program prone to errors while scanning VST plug-ins. Offending plug-ins crash the program, and upon relaunch it "forgets" where it was and always starts from scratch—really annoying. After contacting Soundop, they said they were checking a new way to scan the plug-ins to correct this. Right before going to press, they released a new version (they are now at version 1.4), and the program went through all the plug-ins flawlessly. We were up and running after some time (we have quite a lot of plug-ins to scan).

Plug-in support seems to be a problem common to many audio editors. We found many have problems at scanning audio plug-ins, and this seems one of the lines that separate the market. Anyway, we have been in contact with the programmers, they have been listening, and they actually keep their promises. They are fast developers.

The program is Windows only, and supports XP up to Windows 10. Since it has a multitrack part, the audio editor supports only mono and stereo files. Sampling rates go up to 192 kHz (and go as low as 6 kHz, in case anyone finds that useful). Recording audio resolution goes up to 24-bit, but internal processing can be done much higher—up to 64-bit floating point.

In terms of supported audio formats, we have a broad choice, as usual: MP1, MP2, MP3, AAC, M4A, CAF, AC3, WAV, WAV64, AIFF, AU, Ogg Vorbis, FLAC, APE, WMA, and raw PCM files. This is another program that supports all the lossless compressed formats available, including Apple lossless and Monkey Audio, besides the more widespread Ogg Vorbis and FLAC.

Plug-in formats supported are VST2 and VST3. After managing to finish a scan (the more problematic step we had at the beginning), we found an extra tool that's welcome: a Plug-in manager, which lets us choose which ones to keep and which ones to let go. For someone who has tons of plug-ins, many of which are virtual instruments (useless inside an audio editor) this is welcome. Besides the plug-ins support, we have several DSP functions available, although the GUIs are very simple (a window with a small bunch of faders, usually.) We have Normalize, Gain Envelope, Noise Reduction, and the "bread-and-butter tools" (Compressor, Limiter, Chorus, Flanger, Phaser, Parametric EQ, and Reverb).

There is no support for loops nor for sampler—this is a strictly audio-oriented application. Despite that orientation, we have no support for special master formats (e.g., DDP) nor CD track lists. Perhaps Ivosight might be aiming to eventually develop Soundop into a full-featured DAW.

We have Noise Reduction and Adaptive Noise Reduction, and these tools performed well in our tests. Time Compression and Pitch Shifting are also present, and again we found the results quite good, although the algorithms have been apparently developed in-house.

We can zoom up to sample level, but again we lack the special tools (pencil, pen, whatever) to perform detailed editing. As with Audition, Soundop's audio editing window can display both audio waveform and spectrum, and we have some extra editing possibilities in the spectrum view (copy, paste, delete, and mixing).

As with almost all the audio editors mentioned here, Soundop can import audio from video, but it doesn't support working directly with video.

Finally, as an extra, the audio editor has an extensive and pretty comprehensive Metadata Editing section. Soundop can also open several audio and multitrack projects simultaneously, working concurrently, switching between them, and even export audio projects to a Mixing project, or import a track into the Editor for detailed editing.

Ivosight Soundop: http://ivosight.com/soundop
Windows XP and later—32-bit and 64-bit: $129.00

Requirements for Audio Control Rooms

Photo 1: This is the control room at the Friday Conference Center, University of North Carolina. (Photo courtesy of www.fridayconferencecenter.com)

Richard Honeycutt shares practical design considerations for people who want to build an audio control room.

By
Richard Honeycutt
(United States)

The need, function, and requirements for audio control rooms in performance and worship spaces is a widely misunderstood subject, to the detriment of the listening public **(see Photo 1)**. Back in the 1950s, sound-system controls were often minimal and were placed in a closet from which the operator could hear what was going on in the venue only through headphones, and most of the available headphones were of telephone or voice-communication quality **(see Photo 2)**.

Separate Sound Control Rooms

In facilities that have the sound operator in a separate room, the operator can get some idea of the sound in the venue by using a calibrated audio monitoring system. This would consist of excellent control-room monitor loudspeakers, equalized to match the frequency balance that a listener would hear in a typical seat. Equalization should typically be accomplished via a digital signal processor (DSP), with the settings locked so that they cannot be tampered with by operators seeking to achieve their preferred frequency response. If the audio mix is stereo or left-center-right, two or three equalized monitor speakers are needed. For better realism, suspended omnidirectional mics can be hung over rear seating areas and a surround monitor system can be employed.

Photo 2: Headsets like these (using either crystal or telephone-quality magnetic elements) were often the sound operator's only way of getting an idea of the sound in a venue in the 1950s.

More recently, many facilities have placed the sound controls in a projection booth near the rear of the venue, affording sound operators the ability to see what's going on, but not to hear. Some churches have a large glass window in the back of the room that provides the operators a good view of the sanctuary. Sometimes, the control room is used only for the television broadcast, and the sound controls are inside the sanctuary. The audio feed for the TV broadcast is taken from the audio mixer. The TV production crew hears the audio through monitor speakers in the TV room.

In less well-designed venues, sometimes a small openable glass "drive-thru" window allows some small sampling of the sound in the venue. This is always a poor compromise, since the sound operator is really located in a different acoustical space than the one for which (s)he is controlling the sound.

Separate Recording Rooms

An increasing number of venues are being built with the sound controls in the seating area, and a separate room for audio recording and/or streaming. The architectural plan of a university chapel having a separate recording room is shown in the lower left of **Figure 1**. Note that no provision was made for the recording operator to see into the venue. The reason is that in this venue, there are pan/tilt/zoom video cameras controlled by the sound operator, and there is a video monitor in the recording room. The absence of a window decreases sound leakage from the recording room (where recording operators may be talking) into the chapel, and also allows the lights to be on in the recording room without affecting the lighting in the chapel. Evenstar Audio Peregrin monitor speakers are used to provide accurate sound for the recordist.

Designing a Control Room

In designing a recording/streaming control room, the first consideration is location. If a window is to be provided so the operator can see into the venue, good sight lines are a must. If video cameras and a video monitor will be used, location is less important.

A common concern raised by owners and architects who are planning or designing a control room is the need for sound isolation. If the sound in the venue will be loud, then good isolation is a must. If the venue is used for serious music concerts for which sounds leaking from the control room would be distracting, good isolation is important. Typical conversation averages about 65 dBSPL. A loud rock, orchestral, or band concert can reach 100 dBSPL or above. The ambient noise inside the control room usually should not exceed about 35 dBSPL. The ambient noise inside a church sanctuary or an auditorium for serious music concerts should be as low as possible: typically not more than 10 to 15 dBSPL. The degree of sound isolation needed equals the difference between the sound level in the noisier space and the level in the quieter space. For example, to keep the level in the control room at or less than 35 dBSPL if a loud concert is going on, at least 100 dBSPL − 35 dBSPL = 65 dB of isolation is needed.

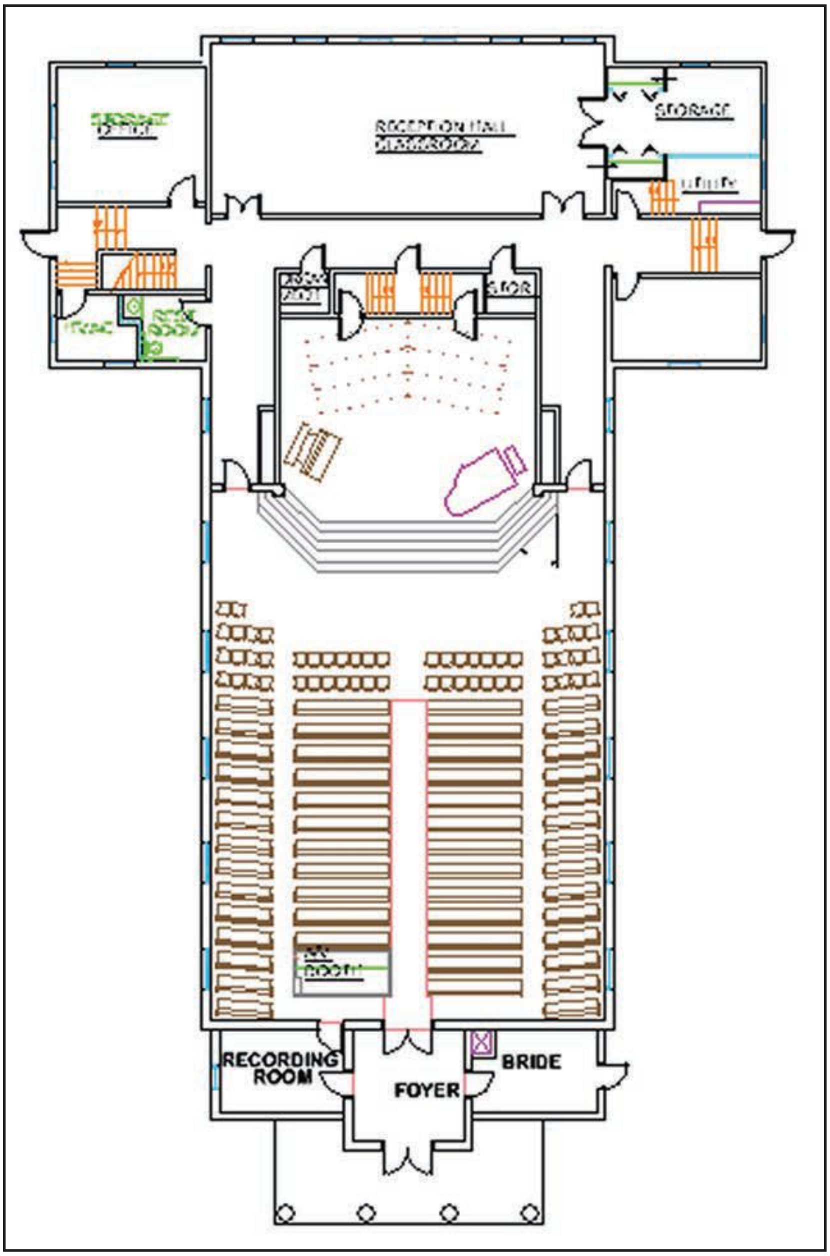


Figure 1: This university chapel has a recording/production room (lower left corner, adjacent to foyer) served by calibrated monitor speakers.

Acoustical isolation is provided by massive materials, sometimes with the aid of viscoelastic layers added. Typical massive building materials include brick, concrete, and gypsum board. The sound-isolating ability of a material is quantified by a rating called Sound Transmission Coefficient (STC). A 3″ thick solid concrete wall has a surface weight of about 40 pounds per square foot (lb/sf) of wall area, and has an STC of 47. Doubling the thickness doubles the surface weight to about 80 lb/sf, and increases the STC to 53. To get an STC of 59, we'd have to double the thickness again, winding up with a 12″ thick concrete wall.

Consider a 10′ × 15′ control room in which the 15′ wall is the separator between the venue and the control room, and an STC 59 wall is used. The sound level in the control room will typically not match what would be calculated by subtracting 59 dB from the sound level in the venue, depending on the interior acoustics of the control room. The frequency of the sound in the venue and the total sound absorption in the control room affect the sound isolation.

More important, in many cases, is the effect of doors and windows. A well-sealed 0.25" thick single-pane glass window has an STC of about 33, and a solid-core wood door has an STC of about 20 (26 if the door is gasketed). A rated acoustical door can achieve an STC of about 53. Thus, the isolation of a heavy wall can be compromised by the performance of doors and windows.

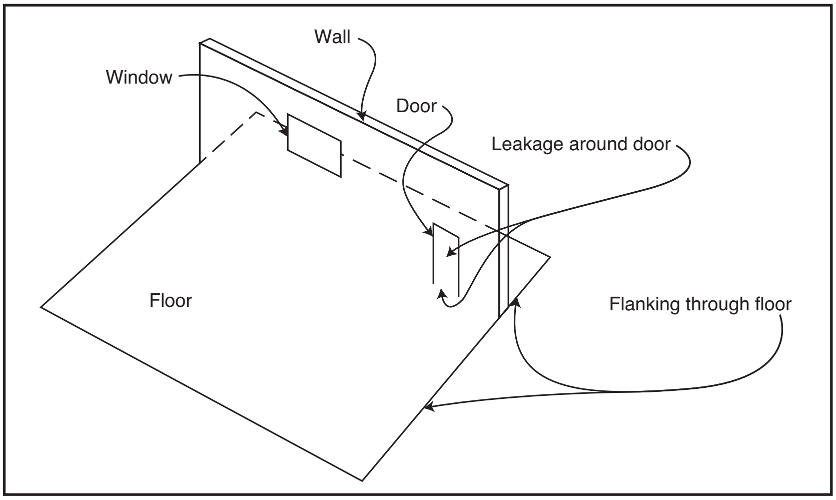


Figure 2: Sound can enter a recording/production/streaming room via flanking paths, without passing through walls, doors, or windows.

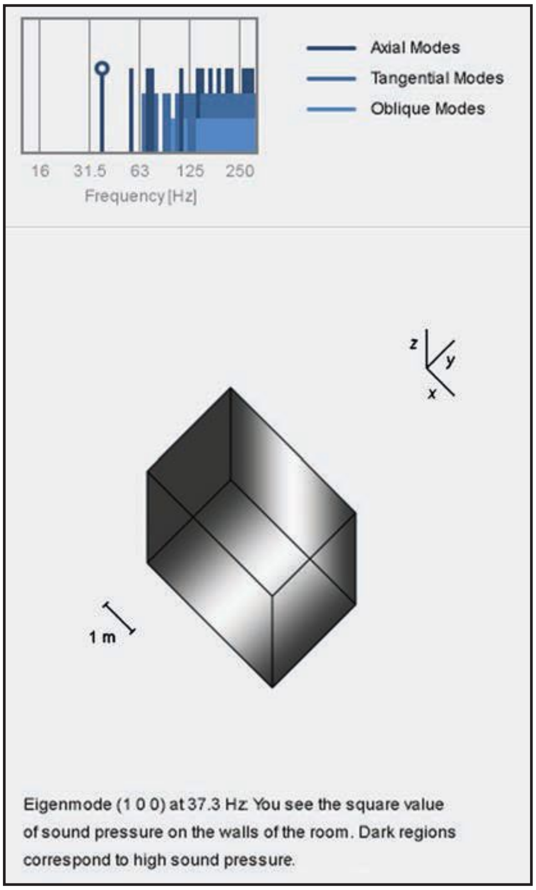


Figure 3: This figure illustrates the lowest mode in a 15′ × 10′ × 8′ control room, both graphically and pictorially. At the modal frequency of 37.3 Hz, the sound level will be high at the ends of the room and lower near the center.

In addition to sound leakage through walls, windows, and doors, sound can enter the control room through flanking paths. These are paths that sound can travel, bypassing an isolating partition. Examples of flanking paths include HVAC ducts or plenums shared among different rooms; walls that only extend to the top of a suspended ceiling, rather than to the roof deck; solid floors extending from the "loud" room to the "quiet" room; and non-acoustically sealed walls and ceilings. A 10% open area around a door can degrade sound isolation by 10 dB. A good acoustical consultant can design a control room so as to achieve any reasonable level of sound isolation needed. **Figure 2** illustrates some common paths for sound leakage.

Acoustical Considerations

Room size and dimensions are often selected without regard to acoustical considerations, but buildup of sound at specific frequencies due to room modes can negatively impact the accuracy with which the recording operator hears the sound. A 15′ × 10′ × 8′ control room will have significant modes up to about 243 Hz. A room having twice those dimensions will only have significant modes up to about 86 Hz. Modes having frequencies above the lowest expected frequency response limit of the source audio or the final listener's sound system should be treated using bass traps.

If a streaming room is used to provide sound from performers using electronic instruments, modes below 40 Hz don't matter, since the lowest frequency produced by the performers will probably be the low E of an electric bass, which has a frequency of about 40 Hz. If the listener will hear the program on computer speakers or low-quality earphones, modes below 100 Hz may not matter. So the considerations regarding room modes depend on the situation. **Figure 3** illustrates the lowest mode in a 15′ × 10′ × 8′ control room (see Resource).

In a control room, you don't want the sound to be colored by comb filtering from early reflections, or by excessive reverberation, so proper treatment with acoustical absorption is necessary. Typical target Reverberation Times (RTs) range from about 0.3 to 0.6 seconds for rooms ranging from a volume of 1,000 ft^3 to 10,000 ft^3, respectively. To avoid a "boxy" sound caused by comb filtering, it is good to treat opposite areas of opposing walls (see **Figure 4**). It is seldom necessary to completely cover any wall with acoustical absorption.

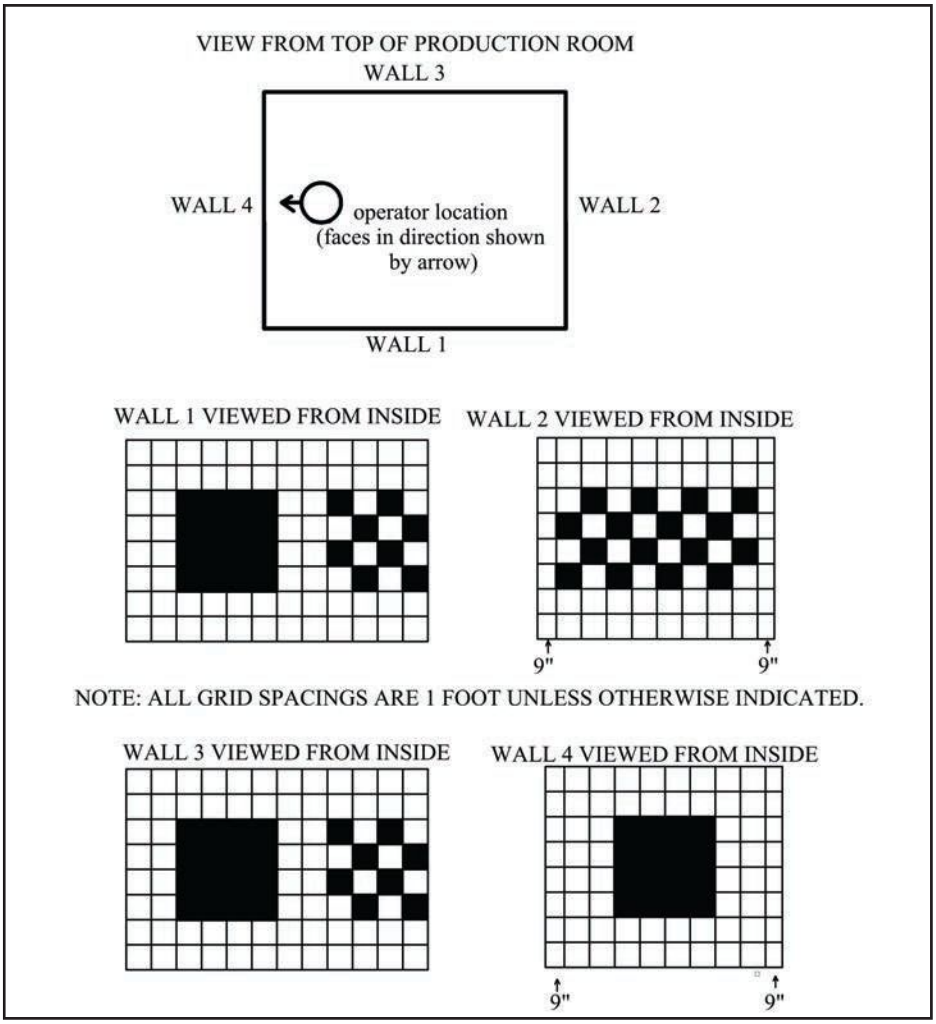


Figure 4: Alternate placement of acoustical absorption (shaded areas) on opposing walls prevents a boxy sound in control rooms.

Three pointers for recording/streaming operators are apropos. Sometimes, event recordings or audio streams include audience sound. There are two important considerations regarding audience sound pickup. First, the mic must be located far enough above the audience that it will not "solo" individual audience members. About 10′ above the heads of the audience is a good minimum.

Second, if the audience pickup is intended to capture group singing with the accompaniment of a musical instrument or group located at the front of the venue, a bidirectional microphone can be used, oriented so that it picks up from the sides, with the pickup null pointing toward the instrument(s). This helps avoid having the accompaniment drown out the audience.

Third, if the recording will be based on a send from the front-of-house (FOH) mixer, the FOH mixer operator should be cautious about the use of effects such as reverb enhancement. What sounds like just enough reverb in the venue can easily sound like a cave when a recording of the event is heard. When possible, it is best to use mic splitters to send the mic feed to a dedicated recording/streaming mixer located in the control room. Then the recording mix, including effects, will not depend on the FOH mix.

Finally, although the practice of "mixing by muting" is common in theaters, and can also work in live sound, it makes for choppy-sounding recordings or Internet streams. Use the faders to bring sources up and down gradually, and the result will be more pleasing for the ultimate listener.

Resource

Dr. J. Hunecke, "Room Eigenmodes Calculator," Wörthsee, www.hunecke.de/en/calculators/room-eigenmodes.html

Exploring Graphene Speaker Development

Graphene is hailed as a wonder material that should be intriguing for speaker engineers—strong, light, and electrically and thermally conductive. In this article, we will discuss the progress being made by ORA Sound and GraphAudio, two pioneering companies with distinctly different approaches to the use of graphene-based drivers that can be used in earphones, speakers and even... microphones.

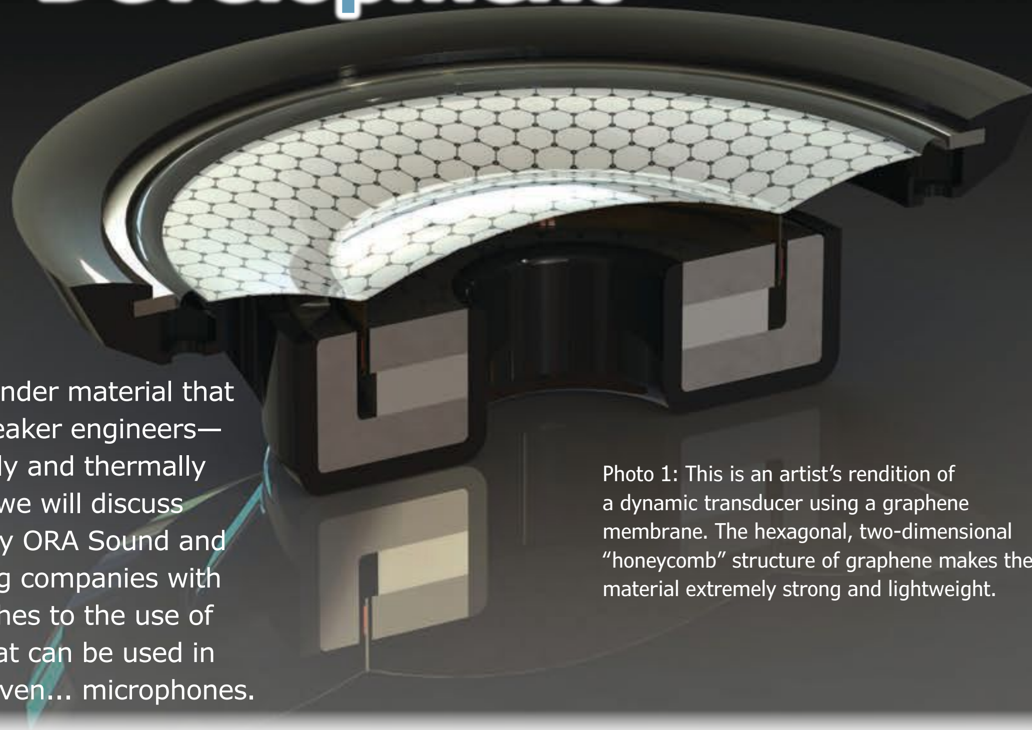


Photo 1: This is an artist's rendition of a dynamic transducer using a graphene membrane. The hexagonal, two-dimensional "honeycomb" structure of graphene makes the material extremely strong and lightweight.

By
Mike Klasco and J. Martins

Paraphrasing from *Wikipedia*, Graphene is a form (allotrope is the more precise technical term) of carbon in a two-dimensional, one atom thick film. Graphene is the basic structural element of other carbon variants (allotropes), including graphite, charcoal, and carbon nanotubes. It is more than 200 times stronger than high-strength (high-carbon) steel. It efficiently conducts heat and electricity, and carbon materials typically have good acoustical properties.

It was originally observed using electron microscopes in 1962 and characterized in 2004 by Andre Geim and Konstantin Novoselov at the University of Manchester. This work resulted in their Nobel Prize in Physics in 2010 "for groundbreaking experiments regarding the two-dimensional material graphene."

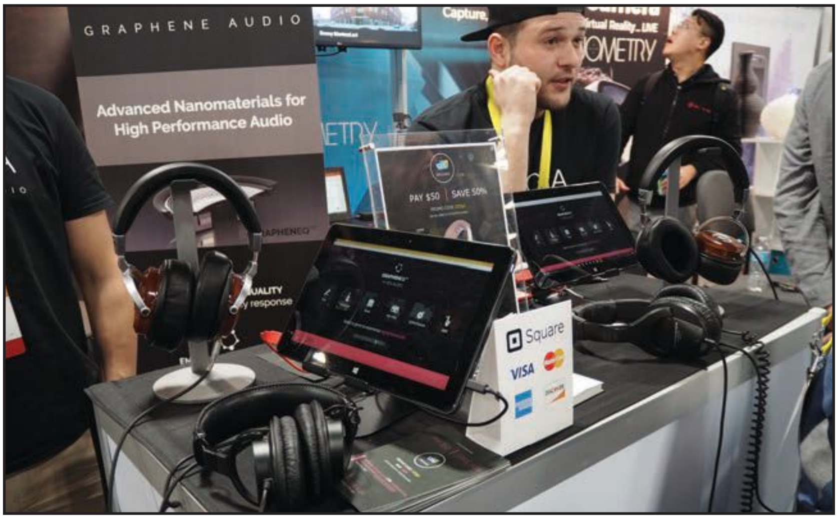
The Tip of the Spear

So how is graphene used in headphones and speakers and why does it matter? The first audio products to be commercialized with a graphene-derivative material are from ORA Sound (ORA Graphene Audio, Inc.), an early-stage start-up that has announced, but not yet released a product. ORA has devised traditional cone drivers from a composite material called GrapheneQ (graphite oxide). Its low density and high stiffness enable potentially higher output drivers.

ORA was conceived at the engineering labs of McGill University. Dr. Robert-Eric Gaskell headed research into how the properties of graphene oxide could be tuned to create the conditions for high-quality sound. Shortly after producing the first prototype audio transducer based on graphene oxide technology, Gaskell joined TandemLaunch, an incubator for university research and assembled the ORA team. If ORA was going to succeed where other commercial endeavors of graphene had failed, they would have to put manufacturability at the forefront. Turning to Northwestern University, they found the research of Professor SonBinh Nguyen,

Dr. Robert-Eric Gaskell introduced a prototype of a GrapheneQ speaker at the ALMA International Symposium and Expo (AISE) 2017. ORA Sound was the first company to bring to market its development of GrapheneQ, a graphene oxide it is commercializing in composite speaker diaphragms.

An earlier prototype speaker membrane formed from a Graphene composite material (GrapheneQ) from ORA Sound



At the Eureka Park zone during CES 2017, ORA Sound successfully introduced the idea of GrapheneQ-based headphones and provided side-by-side comparisons with their first prototypes and well-established over ear models.

In 2016, GraphAudio licensed the graphene audio work and patents from The Lawrence Berkeley National Labs at UC Berkeley. GraphAudio's goal is to develop a new generation of graphene-based microspeaker transducers with high sensitivity and ultra-low distortion. The image shows a comparison of an 8 mm dynamic speaker and an 8 mm GraphAudio Gen1 transducer.

who had a way to manufacture graphene sheets in a simple process that could be easily scaled to production quantities. Then the focus was on tuning GrapheneQ into a high-performance diaphragm material for audio.

The team made its graphene oxide by reducing graphene and then adding a proprietary blend of cross linkers to create the composite. The intrinsically high tan delta means that it requires less damping mass (especially at low frequencies) than commercial devices to prevent unwanted spurious breakup response characteristics. Less damping at lower frequencies also means that the bass and treble response are both extended, which ideally results in faster transient response and quicker settling times.

GrapheneQ is stiff and distorts very little thanks to its high Young's Modulus of up to 130 GPa. This enables sound waves to travel quickly through the material, pushing diaphragm "break-up" to beyond audio frequencies where they are more easily damped. The material has a very high thermal conductivity, which enables design for enhanced heat dumping from the voice coil and out from the diaphragm using various techniques. Specifically, the voice coil bobbin can be thermally conductive or in a microspeaker or headphone driver a bobbinless coil can be bonded (using thermally conductive adhesive) directly to the diaphragm.

The ORA researchers hope that GrapheneQ will become a standard material for loudspeaker membranes in the future. They have a Kickstarter crowdfunding program for a headphone and are establishing an OEM program for their diaphragm technology.

But graphene's promise is that not only can it be used as an additive for speaker diaphragms in traditional moving coil, magnetic dynamic speakers, such as ORA is doing, but also theoretically as a speaker itself.

More Research and Development

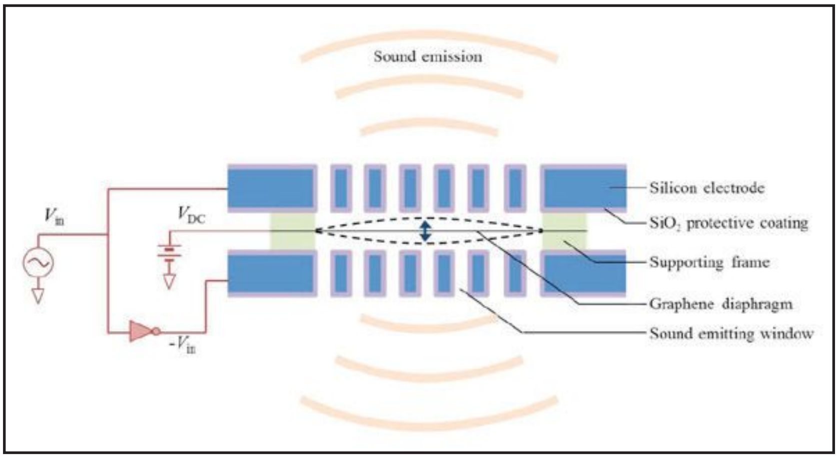
In 2012 at the Lawrence Berkeley National Labs (LBNL) at the University of California, Berkeley, Dr. Alex Zettl and Dr. Qin Zhou were researching graphene for audio applications. About a year later, a proof-of-concept graphene earphone, consisting of a graphene diaphragm sandwiched between electrodes to create the electrical field, was demonstrated and received a lot of press coverage. Now the LBNL technology has been exclusively licensed by GraphAudio to be developed into commercialized audio products.

GraphAudio is developing a complete implementation of a true electrostatic driver where the pure graphene diaphragm functions as part of the "motor." The in-canal earphones consisted of a graphene diaphragm sandwiched between electrodes that created the electrical field. When this field oscillates due to the audio signal, it causes the graphene to vibrate in a physical analogy to the audio electrical signal and this generates sound. It's essentially an electrostatic speaker; but instead of a metalized polymer film diaphragm, graphene is used. The graphene diaphragms are very thin and light with a small spring constant so that the air itself damps its motion. The air-damped graphene converts almost all of its energy into sound and so is potentially extremely efficient. Scholarly work has continued but a number of practical challenges remain before graphene can emerge as a viable—and game changing—alternative for established transducer technologies.

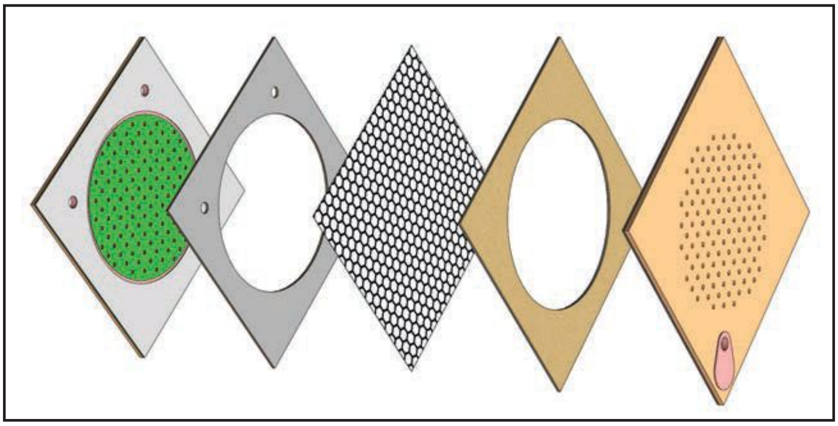
As mentioned, in 2016, GraphAudio exclusively licensed the graphene audio work and patents from The Lawrence Berkeley National Labs at UC Berkeley. With both the rights to the audio work and IP as well as the original research team of Zettl and Zhou, they began to hire other specialists and continue the path to commercialization. Business Co-Founders are Fred Goldring, CEO, and Frederick Wells, Chief Business Development Officer. Lorance (Lonnie) Wilson, CTO and VP Engineering, has much experience in semiconductor processing and commercialization of new technologies at Fairchild, AMD, and Intel, and leads the development efforts.

GraphAudio's goal is to develop a new generation of graphene-based micro audio componentry that potentially will outperform the current generation and open new capabilities. GraphAudio's patent-pending true graphene transducer delivers an electrostatic micro speaker that promises high-resolution audio with high sensitivity and ultra-low distortion.

The obvious targets for graphene-based audio components range from speakers (e.g., tweeters

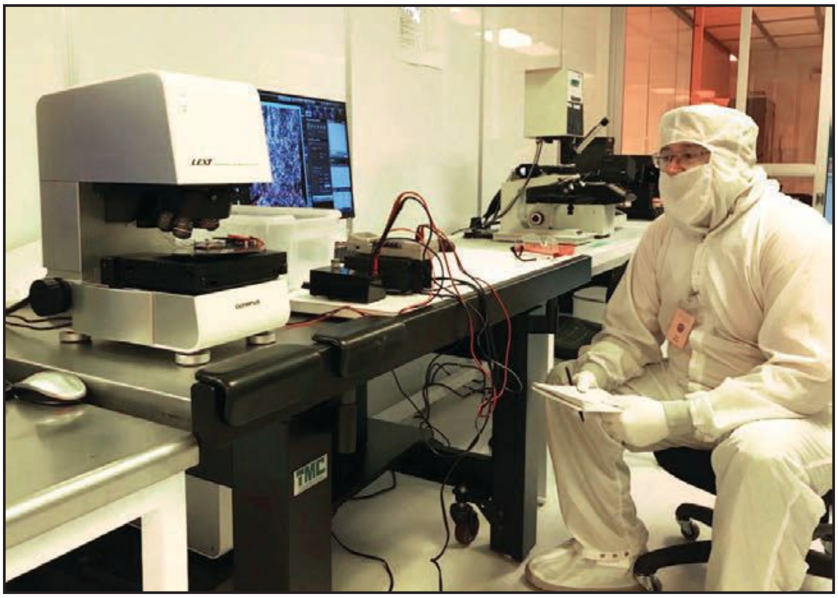


A transducer speaker schematic—in the center is the Graphene diaphragm.



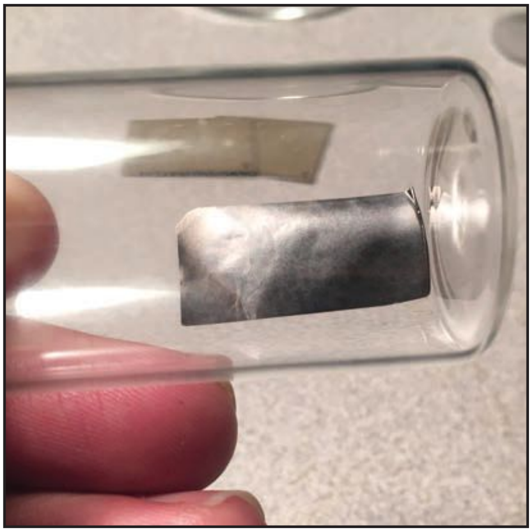
Single die exploded view stacked: Electrode 1, Spacer 1, Graphene, Spacer 2 and Electrode 2. The pure graphene transducer design functions as a speaker or a microphone, or both, but the diaphragm tensioning, electrode spacing, and bias voltage, among other parameters (e.g., sensitivity vs. Xmax excursion, electrode attach technique, etc.) still must be optimized for full functionality.

GraphAudio's Harry Chou works at Chemical Vapor Deposition (CVD) system to deposit Graphene film.



Dr. Burt Fowler (right) uses interferometry to characterize an active device.


Dr. Burt Fowler (right) and Lorance "Lonnie" Wilson are in the lab with a plasma etch tool used for processing.


The Graphene foil is used as a Seed Layer material on which GraphAudio "grows" the graphene. This sample has been coated with graphene and is ready to process.

and microspeakers) to earphones/headphones and microphones. The promise for audio enthusiasts is that graphene's high strength allows for the relatively large, free-standing diaphragms necessary for effective low-frequency response, assuming traditional planar dipole configuration. Due to the intrinsically low moving mass and high Young's modulus, achieving high-resolution audio is a given. The symmetrical push-pull electrostatic drive has been the core technology of the finest audiophile headphones and speakers and studio microphones. (Most readers are familiar with the reputation of the Sennheiser Orpheus and Stax electrostatic headphones, Neumann studio condenser mics, and a number of legendary electrostatic speakers.)

Going Smaller

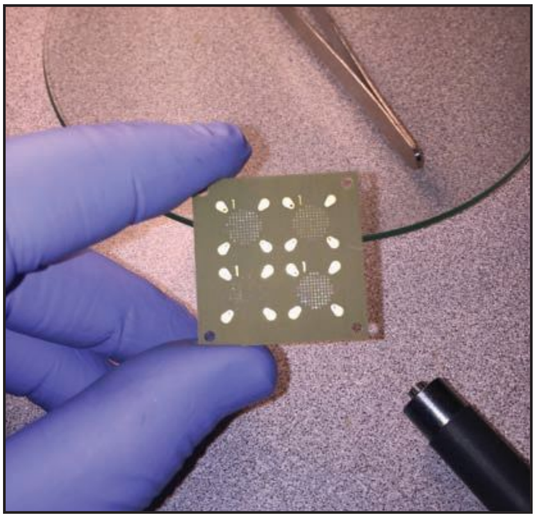
The development work being done by GraphAudio is integrating pure graphene diaphragms for higher sensitivity with much better form factors than current electrostatic implementations using polymer film (metalized or otherwise conductive coating diaphragms).

The ability to power graphene earphones and speakers using conventional mobile battery power expands their application from just the boutique end of the market. Batteries (or supercapacitors) for the DC bias, work for graphene since they source only voltage and virtually no current. Since the power is tiny, there is no need for high current and small batteries suffice. GraphAudio's innovative work on the drive electronics are highly efficient as well.

In theory, graphene transducers can be economically produced in high volume utilizing automated fabrication techniques. With any new technology, moving from laboratory into the real world is the challenge. Just understanding and learning how to control the fabrication processes for optimization of the usual parameters are the hurdles GraphAudio is just beginning to get under control. The measurement and full characterization of the transducer is where much of their focus lies, including optimization of amplifier parameters, bias voltage, as well as transducer excursion, response, tensioning and sensitivity trade-offs. The low-hanging fruit are microphones, tweeters, earphones, and headphones due to the low excursion requirements—but winning the lottery means microspeakers.

Zettl, GraphAudio' Co-founder, explained that the appeal of pure graphene for earphones and headphones is that the per-area air damping coefficient significantly decreases when the size of the diaphragm falls below the sound wavelength. For small speakers, a thinner and lower mass density diaphragm is required to continue the dominance of air damping. Such a diaphragm is difficult to realize. If conventional materials such as metalized Mylar are made too thin, they invariably fatigue and break or lose their optimum tensioning. Graphene is an ideal building material for small, efficient, high-quality broadband audio speakers because it satisfies all the above criteria. It is electrically conducting, has extremely small mass density, and

GraphAudio staff uses this Chemical Vapor Deposition (CVD) tool for Graphene development.



Here is a finished run with four 8 mm transducers, which will be diced for testing.

GraphAudio is exploring graphene-based acoustic transmitters and receivers for all applications, including professional microphones. GraphAudio's Mic1 is its initial proof-of-concept studio mic. Preliminary graphene microphone work from the UC Berkeley technology spin-off was reported in scientific journals in 2015. In the GraphAudio Mic1, a large graphene diaphragm has been fabricated and integrated into an omnidirectional condenser microphone capsule.

can be configured to have very small effective spring constant.

The pure graphene transducer design functions as speaker or microphone, or both, but the diaphragm tensioning, electrode spacing, and bias voltage, among other parameters such as sensitivity vs. Xmax excursion, electrode attach technique, and so forth still must be optimized for full functionality.

Wilson, Co-founder and CTO, comments on GraphAudio's progress: "We've made great strides with the device so far and I'm very proud of our team. It's been a real pleasure working with such a high-level interdisciplinary group. The process we've developed produces robust and stable transducers

Harry Chou at JJ Pickle Center (UT Austin) and Dr. Hu Long in the Zettl Lab (Cal) are both working on CVD reactors.

and we are currently optimizing the device package for performance prior to engineering release. We are starting to work with early adopter customer/partners to help pull us into commercialization soon. It's really fun to see that as a business, we seem to be experiencing the same push/pull dynamic as our device."

Goldring, Co-founder and CEO, expressed his passion for the project, "Graphene transducers are the holy grail of audio and our goal is to become the standard for microspeakers and microphones inside every connected device on the planet. With more and more people today (and the vast majority of young people) listening to their music on mobile devices and in-ear buds rather than large bookshelf and floor speakers, our graphene transducers will finally bring high-res audio economically to the masses. GraphAudio's transducers will introduce the first quantum leap in consumer audio technology since the advent of the moving coil, magnetic, dynamic speaker in 1921. For an entire generation which has been weaned on compressed music heard only through marginal earphones, our graphene transducer technology will be a real 'ear opener'."

GraphAudio envisions arrays of these transducers enabling control of directivity and coverage, but also audiophile applications for full range dipole speakers. Certain product categories will be available for licensing while others will be reserved for GraphAudio's own products.

Resources

R.-E. Gaskell, "GrapheneQ: Graphene Composites for Improved Sound Quality and Increased Efficiency in Portable Devices," *Loudspeaker Industry Sourcebook 2017*.

P. Gaskell, R.-E. Gaskell, J. W. Hong, and T. Szkopek, "Graphene Oxide Based Materials as Acoustic Transducers: A Ribbon Microphone Application Case Study," Audio Engineering Society Convention Paper, the 137th Convention, October 2014.

——"New Graphene Cones from ORA Sound," *Voice Coil*, April 2017.

"GrapheneQ Acoustically Optimized Graphene Materials for Improved Audio Drivers," ORA White Paper, www.ora-sound.com.

Kickstarter, www.kickstarter.com/projects/413314819/ora-the-worlds-first-graphene-headphones

Sources

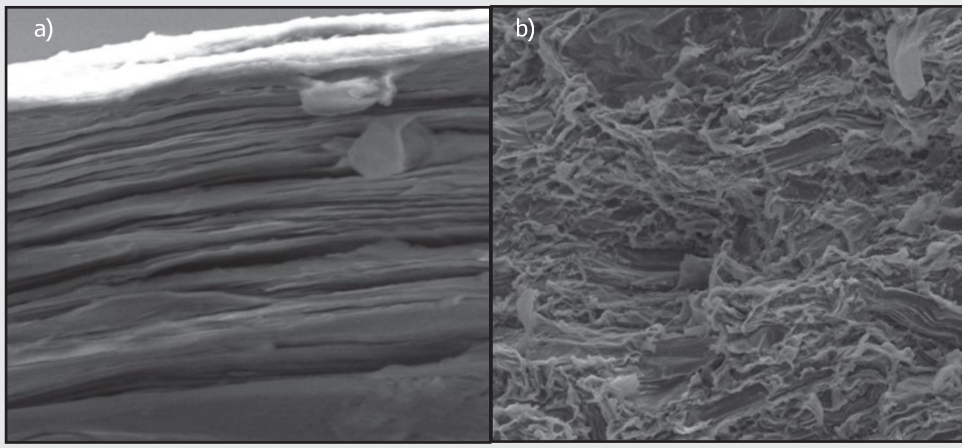
GrapheneQ
ORA Graphene Audio, Inc. | www.ora-sound.com

Patent-pending graphene-based transducer
GraphAudio | www.graphaudio.com

ORA's GrapheneQ Loudspeaker Technology

ORA's GrapheneQ technology is a new class of loudspeaker diaphragm material that is formed with the newly isolated super-material, Graphene. Graphene is the strongest and lightest material known to man. Graphene was first isolated in 2004 and has brought a rush of research but, until now, no consumer products have been able to make use of this incredible material in a way that yields significant improvements. ORA's GrapheneQ is the first Graphene based material that can provide significant performance improvements at a cost that makes it a viable solution for consumer products.

ORA's GrapheneQ is comprised mostly of Graphene. It is a laminate material formed with thousands of layers of single atomic layers of carbon that have been bonded together with ORA's proprietary blend of cross-linking agents. By adjusting the cross-linkers, ORA's engineers can tune the balance of the critical mechanical properties of the material. These properties are the Density (Weight) Young's Modulus (Stiffness) and Damping (Loss Factor). The relative values of these parameters in a material will determine the material's performance in a loudspeaker membrane application. Traditional materials typically have good performance for two out of three of these mechanical parameters. GrapheneQ has excellent performance in all three, reducing the amount of compromise a loudspeaker designer must make in a design and improving the efficiency, frequency response, time domain response, and distortion performance of a loudspeaker.

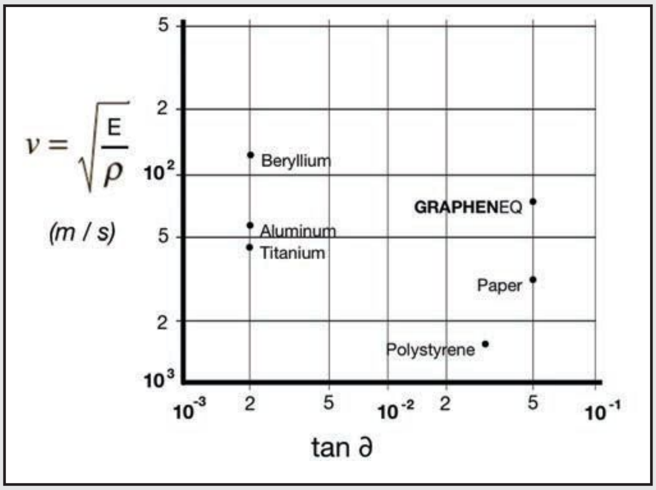


The edge view shows electron microscope images of GrapheneQ's laminar structure (a) vs. a typical Graphite-polymer material (b). This laminar structure provides the stiffness that makes GrapheneQ so ideal for loudspeaker applications.

A major hurdle that needed to be overcome was the forming of GrapheneQ materials into cone and dome shapes for standard woofer and tweeter designs. Beginning with a flat sheet, graphene's stiffness does not allow it to be stretched or formed into cones, domes, or other geometries with a non-zero Gaussian curvature. Early attempts to form loudspeaker cones and dust caps led to less than perfect results. Meanwhile, the deposition technique has evolved to allow for the production of a 40 mm headphone driver and a 3" loudspeaker. The ORA Sound team developed a technique that, when combined with a patented method from Northwestern University, allows GrapheneQ to be directly formed into loudspeaker geometries. The results have been impressive. The nano-scale of the starting material means the only limitation to forming the material is the resolution of the mold that is used in the process. The graphene flakes that form the laminate material, GrapheneQ, self-assemble parallel to the mold's surface creating an isotropic material in the desired shape. Any surface finish is achievable depending on the characteristics of the mold.

The size and thickness of GrapheneQ membranes are also widely variable. There is no theoretical limit to how large or small a membrane that can be made. There is also a wide range of possible thicknesses. ORA Sound has made materials as thin as 10 μm and as thick as 300 μm.

ORA's core technology has provided some exciting results and received a lot of attention from consumer audio OEMs as well as from cellphone manufacturers, hearing aid manufacturers, and many other industries that rely on loudspeakers in their products. Since ORA Sound was formed, its research team focused on optimizing three variations of GrapheneQ for three specific applications. ORA Sound now has a low-density material targeted toward full-range cones, a very stiff material



Comparison of GrapheneQ with other materials relative to internal loss and sound velocity

	E (GPa)	ρ (g/cm³)	Poisson's Ratio	v_s(m/s)	tan ∂	k_T (W/mK)	$FOM_\alpha = \frac{v_s}{\rho} = \sqrt{\frac{E}{\rho^3}}$
Titanium	113	4.5	0.34	5011	0.002	16	1.11
Aluminum	70	2.7	0.36	5092	0.002	240	1.89
Mylar	3	1.39	0.4	1469	0.035	0.15	1.06
Beryllium	300	1.85	0.02	12734	0.002	216	6.88
Paper	2.5	0.6	0.2	2041	0.034	--	3.4
CVD Diamond	105	3.5	0.1	5477	0.01	1600	1.56
GrapheneQ "B"	50	1.1	0.18	6742	0.05	212	6.13
GrapheneQ TL	130	0.8	0.18	12748	0.086	1500	15.93

Here are the material parameters for several common loudspeaker membrane materials. GrapheneQ "B" is ORA's current most popular material. GrapheneQ TL is the theoretical limit of ORA's Graphene material. ORA's material scientists are continuing to improve this technology with the goal of reaching these upper limits in a formable loudspeaker membrane material.

From left to right are shown a 2" GrapheneQ cone, a 40 mm GrapheneQ headphone driver membrane, and a cell phone microspeaker made with GrapheneQ.

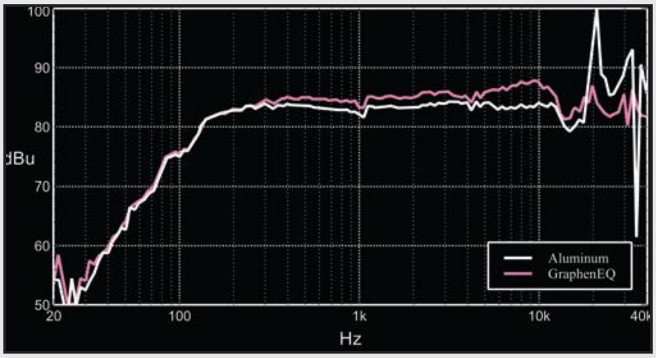
for tweeters, and is working on a material with increased thermal conductivity for microspeakers.

As Dr. Robert-Eric Gaskell, ORA's Co-founder and CEO explains, "Currently our GrapheneQ material provides the stiffness of aluminum with the damping of paper at a density that is less than typical polymers. With further work we are confident that we can approach the stiffness of Beryllium but with lower density and significantly better damping."

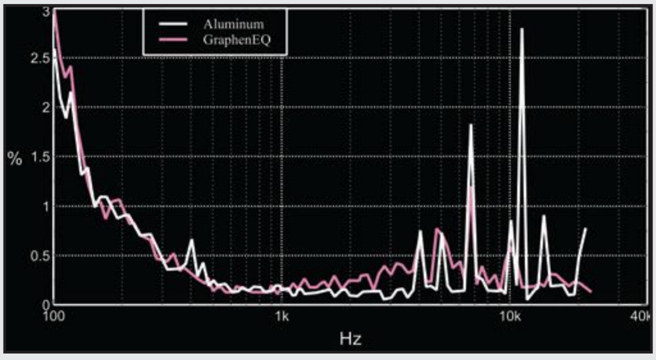
"GrapheneQ's loss coefficient is higher than even the most high-tech paper pulp materials available from the world's top loudspeaker membrane manufacturers, with one order of magnitude improvement in Young's modulus," adds Murilo Alvares, Head of Acoustics at ORA. "Also, the sound output, when compared to paper pulp, can be improved by up to 12 dB, (higher numbers when the diaphragm mass is a significant part of the moving mass)."

In short, GrapheneQ provides:

- More sound output (up to 6 dB)
- Smaller, thinner loudspeakers
- Extended frequency response
- Significantly faster transient rise time and settling time
- Fewer design trade-offs for Loudspeakers of any size
- Audible sound quality improvement



Frequency response of a 2" Aluminum cone (white) vs. a 2" GrapheneQ cone (pink). The GrapheneQ cone has approximately 3 dB more output through most of the bandwidth and a smoother response due, primarily, to the far superior damping of the GrapheneQ.



The THD of an Aluminum cone (white) vs. a GrapheneQ cone (pink) shows the GrapheneQ cone produces significantly less distortion than the Aluminum cone.

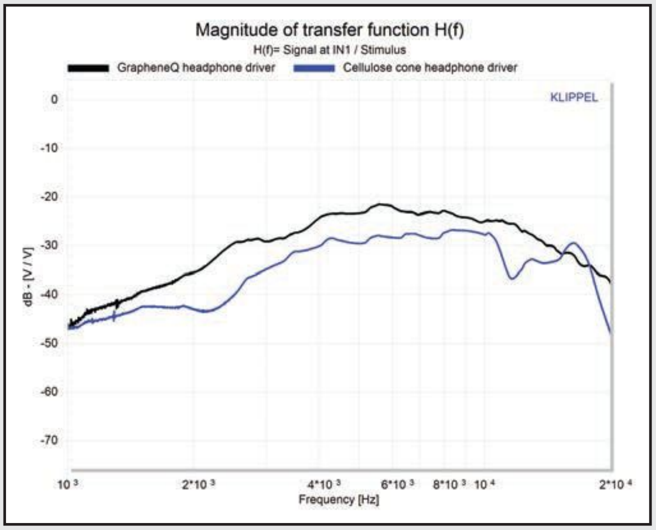
What's Next?

With less than two years of development, ORA Sound has grown its graphene loudspeaker technology from a single, hand-assembled prototype to a reliable and tunable process, and the company started to look at manufacturing. To create a manageable test-field for its technology, ORA decided to follow the most popular route these days, and after a successful presentation at CES 2017, in June 2017 it launched a crowdfunding campaign on Kickstarter for ORA Graphene Headphones—a World's First.

The company had already done a small run of headphones for demonstration and presentations. The purpose of this run of headphones was to help the transition into large-scale manufacturing of GrapheneQ as well as to show prospective investors the potential of this new material.

The ORA Graphene Headphones were optimized for its 40 mm GrapheneQ driver, enabling listeners an opportunity to experience the sound quality that the technology could provide. Measurements of Young's Modulus and density are informative,

The 40 mm cellulose headphone driver (left) and a prototype 40 mm GrapheneQ headphone driver (right).



This is the unbaffled (no enclosure) Magnitude response of a 40 mm cellulose headphone driver (blue) vs. a 40 mm GrapheneQ headphone driver (black). The GrapheneQ headphone driver has a smoother frequency response and approximately 6 dB more output through most of the bandwidth.

In June 2017, ORA Sound launched a successful crowdfunding campaign for the ORA Graphene Headphones, optimized for its 40 mm GrapheneQ driver. The first units should be shipping in March 2018.

but there is nothing more convincing than being able to hear the technology in person.

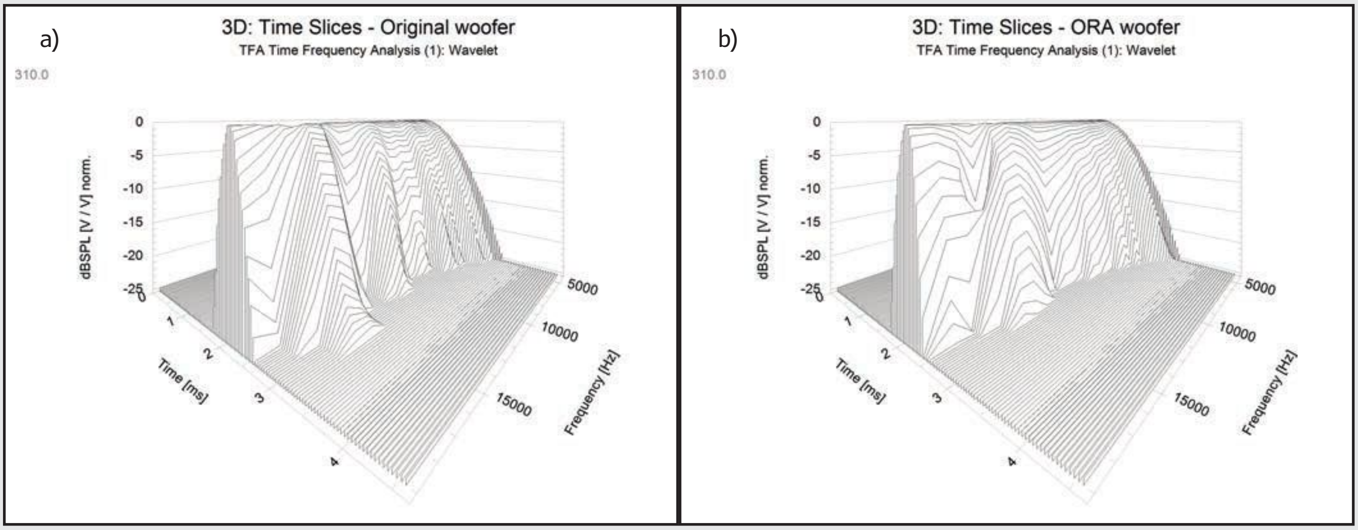
The Kickstarter campaign seemed to confirm the company's excitement with this first concept, and ORA received enthusiastic support from 2,055 backers, amassing CA$774,768 during the campaign, more than five times its original pledged goal of CA$135,000.

ORA GQ Headphones are supposed to be shipping March 2018, the company has already reported significant progress in its manufacturing process, and updated the original design significantly—including adding all the features that were proposed during the crowdfunding campaign—making it one of the first companies to create a commercially viable application for graphene.

The ORA GQ Headphones—available on Kickstarter for the early price of $199 (retail price is expected to be $499)—come equipped with ORA's GrapheneQ membranes, providing improved efficiency and up-to a 70% increase in battery life over standard wireless headphones. To demonstrate the benefits in terms of better power handling, the headphones were designed with both wired and Bluetooth wireless operation. For wireless operation, ORA expects to be able to provide 26 plus hours of playback time on a single charge, having adopted Qualcomm's CSR8675 chip, with integrated support for aptX, aptX HD, and Qualcomm TrueWireless stereo, in the production model.

The company also reported adding features such as a gesture control track-pad, a digital MEMS microphone, breathable lambskin leather, and an ear-shaped design optimized for sound quality and isolated comfort.

In the meantime, ORA Sound continues to engage with OEMs and ODMs that are interested in testing the material and becoming early adopters of ORA Sound's GrapheneQ technology.



These are waterfall plots for a 40 mm cellulose headphone driver (a) and a 40 mm GrapheneQ headphone driver (b). The GrapheneQ headphone driver shows significantly less resonance and settles much faster than the cellulose driver.

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Consider the Source

Line Source Speakers vs. Point Source Speakers in Media Rooms

This article discusses the differences between Line Source and Point Source speakers, detailing what applications best suit each technology, and how differences in speaker technology impact placement, design, and sound clarity.

Photo by Derek Novaes

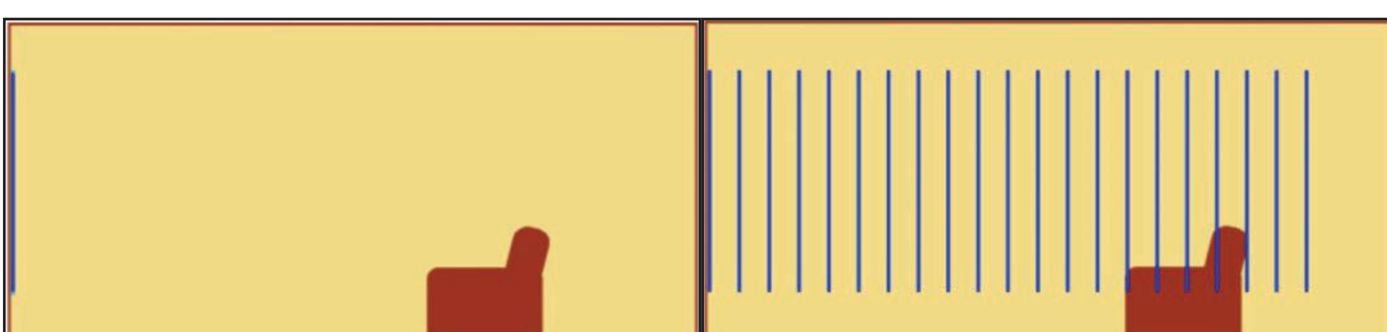
By
Luc Guillaume
(Managing Director, Wisdom Audio)

It's a universal fact that speakers don't all work or sound the same. And for anyone craving an audio aficionado's dream setup and venturing into research and testimonial mode, it can quickly become a highly subjective and overwhelming topic. A look at the market will reveal that most speakers available are point source. But there is another category of speakers that don't get a lot of airtime, line source, which can add to the confusion.

What is the difference between these two speakers? What's the best fit for a home theater or a large open media room? The answer comes down to physics and math. The difference between a line source and a point source speaker is in the way the sound propagates, or moves, in the room. Understanding how the sound is going to propagate in the room is a critical factor of ensuring superior audio clarity and an exceptional listening experience.

Overview of Point Source Speakers

Imagine a stone being dropped into a pond, creating circles that get bigger and bigger as they

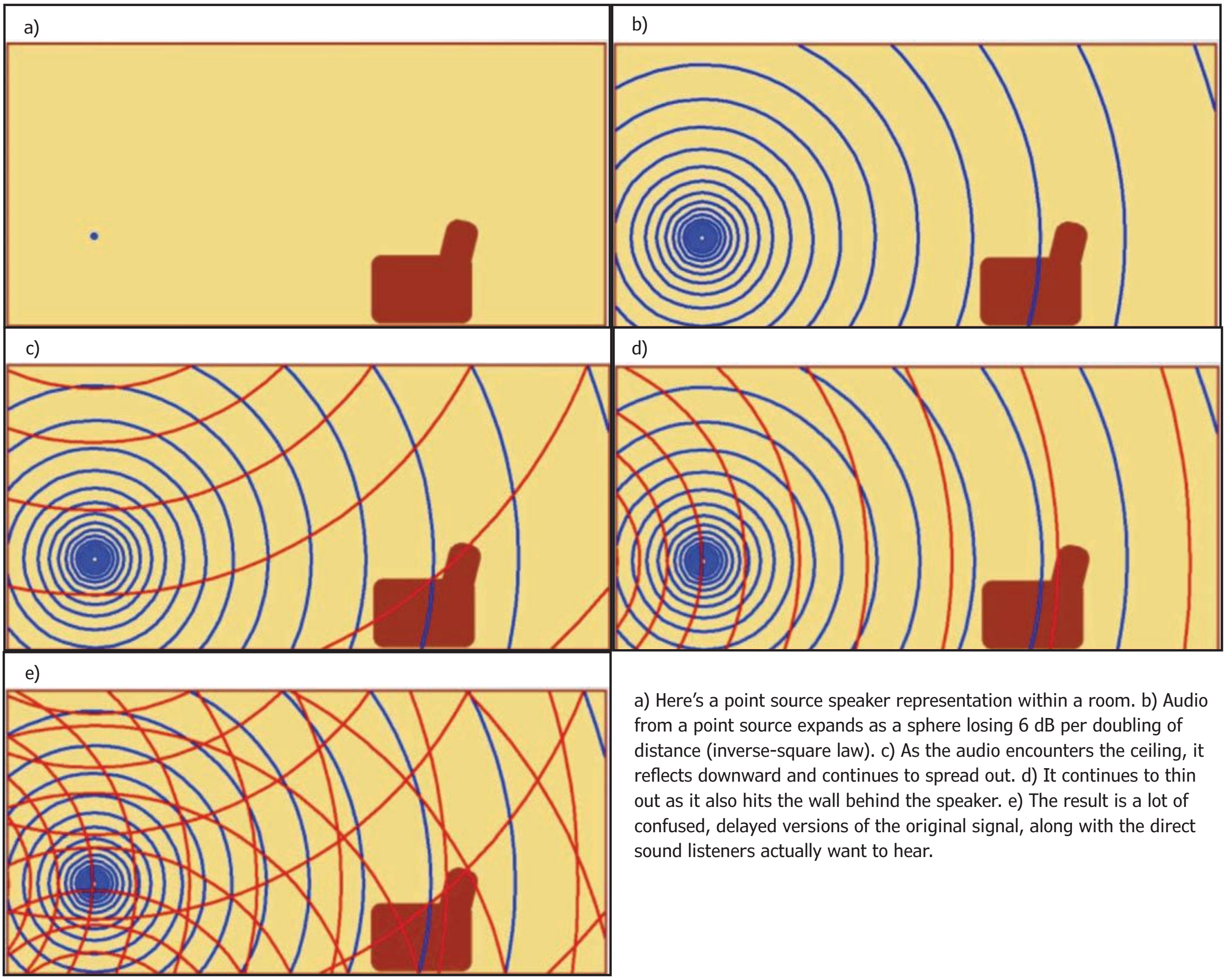


Here's a line source speaker on the wall. A line source provides more direct sound vs. reflected sound because the audio expands like a cylinder, transferring energy with only a 3 dB of loss per doubling of distance. Line sources have half the propagation loss of point sources, providing better row-to-row consistency and more power over distance. For example, if the ceiling and floor area represent 48% of the total in room boundary, and with line source audio directed in a cylinder, that's 48% of total boundary no longer activated. This equates to hearing more of the audio directly from the speaker and less of the room as well as a wider sweet spot because there are many fewer reflections.

radiate outward from the source. That is an example that's often used to describe how waves (sound and otherwise) move. In a 3D world, that circle would be a sphere. The waves of sonic energy from a point source speaker expand in all directions in an ever-enlarging sphere. The sound thins out quickly as the sphere grows, expanding both vertically and horizontally.

In an infinite open space (or even a very large outdoor space such as an outdoor festival), the sphere expands, losing power as it goes (more on that in a minute) but the listener always knows exactly where the sound originates. In a real-world, confined space (e.g., a media room), every time a part of that sphere encounters a surface—a wall, a floor or ceiling, furniture, etc.—the surface reflects part of the sphere back in the opposite direction from which it came. This sets up a series of sound waves, both direct and reflected, and can give listener a sense of being enveloped by sound. But that effect comes at a price in both decreased clarity and uneven volume throughout the space. In a dedicated home theater room, carpeting and acoustic treatment or panels can help control and dampen the reflection at the highest frequencies (10 kHz and above) and enable the listener to hear more information from the speaker. Carpeting and treatments tend to blend in well with the dark environment of these spaces. In a modern living or media space with an open concept design—where there tends to be harder surfaces such as windows and tile or wood floors—acoustic treatments may not be a practical or an aesthetically appropriate solution. Without being able to control the sound reflections via architectural treatment of surfaces, it degrades the listening experience.

Then there's a little law of physics called the inverse-square law, which dictates that total energy



a) Here's a point source speaker representation within a room. b) Audio from a point source expands as a sphere losing 6 dB per doubling of distance (inverse-square law). c) As the audio encounters the ceiling, it reflects downward and continues to spread out. d) It continues to thin out as it also hits the wall behind the speaker. e) The result is a lot of confused, delayed versions of the original signal, along with the direct sound listeners actually want to hear.

from a point source (e.g., a traditional loudspeaker) drops by 6 dB per doubling of distance from the source. In the world of physics, a 3 dB drop represents the energy being cut in half. In the real world, it takes a drop of about 8 to 10 dB before the listener perceives that the volume has dropped by half. So, let's do the math. If you measure the output of a traditional point source system from 1 m and get a reading of 90 dB, that drops to 84 dB at 2 m, 78 dB at 4 m, 72 dB at 8 m, etc. So, the listener who is about 6 m from the source not only perceives that overall level is about half of what the listener at 2 m perceives, but clarity suffers as well as you move away from the source and listeners hear more and more reflections and less direct sound.

In a modern media room environment, the ideal loudspeaker would enable tight control over dispersion patterns to eliminate or at least diminish the decrease in clarity caused by the listener hearing both direct and reflected sound information.

Overview of Line Source Speakers

To understand how line source speakers function, let's go back to that pond used in our earlier example, but this time let's imagine that instead of a stone, we have a wooden plank a few feet wide that's on the edge of the water. When the wood is pushed, it creates a wave the same width as the plank and carries that energy forward. In this example, the piece of wood would be a line source speaker, which propagates sound in the room in the same manner. A line source results in a pattern that is less of a sphere and more of a cylinder. This has two advantages, the first being a significant decrease in reflected energy, which results in greater clarity. The second advantage is that a line source is also bound by the inverse-square law, but the numbers are different—with only a 3 dB drop per doubling of distance. So, using the same example as before, a line source that's playing at 90 dB at 1 m will read at 87 dB at 2 m, 84 dB at 4 m and 81 dB at 8 m. The practical effect is that the line source speaker can play at a lower volume at the source and still have enough energy to provide a top-notch listening experience to listeners further from the source.

A line source speaker can be vertically mounted on or in the wall at a high enough elevation to ensure an optimal listening position for someone sitting or standing in the room. Because the cylinder of sound results in far fewer reflections, the ceiling and floor virtually disappear as reflective boundaries. This makes line source ideal for open-architecture designs. Because the sound is concentrated where the listener's ears are, it does not "thin out" nearly as quickly, and you have surprisingly uniform sound pressure level (SPL) throughout the listening space. People sitting near the speakers and far away will hear close to the same volume.

To hear the TV in the kitchen in a modern, open-concept home, a point source speaker must play loud enough to carry the energy over the distance. It is going to be louder closer to the speaker (e.g., someone sitting on the couch) and less loud to someone in the kitchen. With a line source speaker, because the energy decreases slowly, the audio will carry much farther without having to increase the volume. A person can walk around the entire space and feel that it's not so much louder or softer in any area of the room.

The Geometry and Physics of Sound

In addition to how each speaker propagates sound in the room, the geometry of the space plays a significant role in the listening experience. By measuring and calculating the value of each boundary with the space, listeners can determine their representative percentage. For example, in a space that is 25′ × 20′ × 11′, the floor and ceiling would be 50% of the total boundary while the sidewalls would make up the other 50%. In a space that is longer than it is wide (60′ × 40′ × 11′), as with many modern entertainment spaces, the ceilings and floor represent almost 70% of the space, while walls are around 30%. Since line sources push audio down a line, it essentially eliminates ceiling and floor

Equivalent 4 m Sensitivity

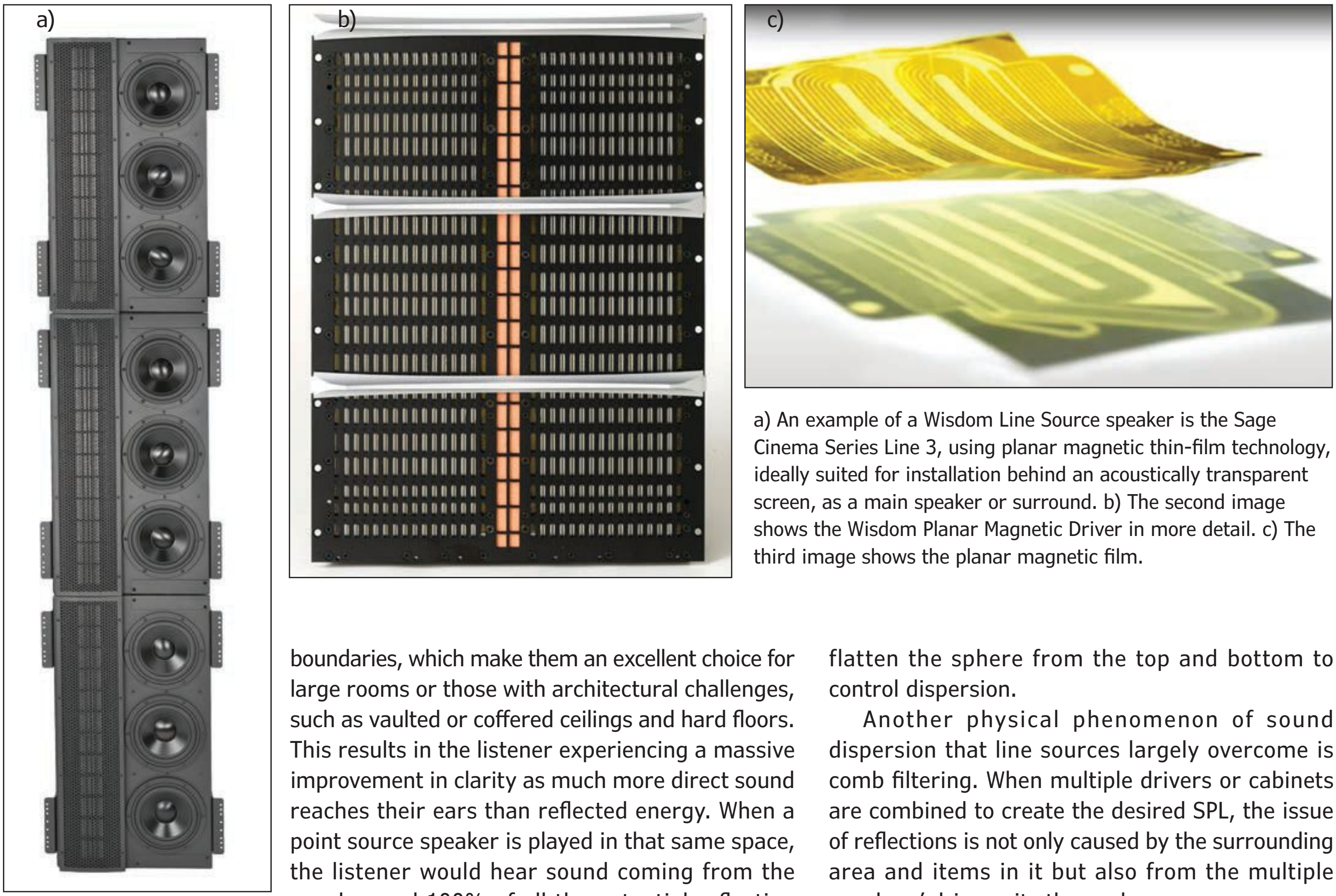
Listening distance in meters	1 m	4 m
Point source dB SPL at 1 W	101 dB SPL	89 dB SPL
Line source dB SPL at 1 W	95 dB SPL	89 dB SPL

Line sources suffer from half as much "propagation loss" as point sources, regardless of whether these point source speakers are horns or domes. This means that at a reasonable listening distance of 4 m (slightly over 13′), a line source will be only 6 dB down in volume from their 1 m sound pressure level (SPL). By contrast, the point source will be 12 dB down. To make a fair comparison of 1 m sensitivities at this reasonable listening distance, you need to add 6 dB to the line source's measurement. As an example, the 95 dB SPL 1 m sensitivity of the Wisdom Audio Sage Series L75 line source is equivalent to having 101 dB SPL in a "normal" point source speaker.

Need High SPL?

Listening distance in meters	1 m	4 m
Point source dB SPL at 400 W	127 dB SPL	115 dB SPL
Line source dB SPL at 400 W	121 dB SPL	115 dB SPL

This chart shows the same speakers, but with the equivalent of 400 W (26 dBW). This yields 115 dB at the listening area 4 m back from the speakers. In the case of the line sources, people who are closer to the speakers are less likely to damage their hearing.



a) An example of a Wisdom Line Source speaker is the Sage Cinema Series Line 3, using planar magnetic thin-film technology, ideally suited for installation behind an acoustically transparent screen, as a main speaker or surround. b) The second image shows the Wisdom Planar Magnetic Driver in more detail. c) The third image shows the planar magnetic film.

boundaries, which make them an excellent choice for large rooms or those with architectural challenges, such as vaulted or coffered ceilings and hard floors. This results in the listener experiencing a massive improvement in clarity as much more direct sound reaches their ears than reflected energy. When a point source speaker is played in that same space, the listener would hear sound coming from the speaker and 100% of all the potential reflection within the space.

Let's also take this opportunity to address the difference between a true line source, such as the planar magnetic drivers made by Wisdom Audio, and more common line array setups. A line array system is not a true line source, but rather a way of combining many point-source drivers in a manner that can mimic some of the characteristics of a true line source. The best way to visualize it is that the top and bottom elements of the array provide "steering" energy. In other words, they basically flatten the sphere from the top and bottom to control dispersion.

An example of a Wisdom Audio Point Source speaker is the Sage Cinema Series Point 3, which features a rotating high-frequency drive unit for vertical or horizonal placement.

Another physical phenomenon of sound dispersion that line sources largely overcome is comb filtering. When multiple drivers or cabinets are combined to create the desired SPL, the issue of reflections is not only caused by the surrounding area and items in it but also from the multiple speakers' drive units themselves.

Let's go back to our pond example again, and this time we'll drop two stones about a foot apart from each other. The way the waves from each stone smash into each other creates what is known as an "interference pattern." Depending on if the waves crash into each other when they are in a "peak" or a "valley" will cause the energy to either double or for the waves to cancel each other out. In practical application, it means that some frequencies are cancelled out resulting in a wave form that looks something like a comb (hence the name) and a perceived "thinness" of the sound. Speaker designers as well as those who design entire systems using point source speakers go to great lengths involving both speaker placement as well as DSP to minimize the effect of comb filtering. Because a true line source has all the sonic energy emanating from a single source, comb filter is all but eliminated.

Surround Sound and Immersive Audio

Within a traditional surround sound media room setup using point source speakers, the left surround and the right surround play at the same level and in balance to achieve the perfect listening seat, called the “sweet spot” or the “money seat.” Everything aligns there, and dealers often work hard to achieve that balance, using a microphone to gather audio measurements. A listener might be able to have the same sound quality when he or she moves one seat left or right or in front, but the farther the listener moves toward one side, the more the left or right channel is the only speaker heard. Line source speakers, however, increase the number of “money seats” to a wider area, making them an ideal option in the surround channel positions. The sound is more consistent from seat to seat, allowing everyone to have the same shared audio experience.

Finally, the last channel to consider are the height channels—in the ceiling, responsible for adding an immersive audio layer to the experience. Here, line source speakers wouldn’t be a practical solution because they would have to be installed into the entire ceiling to cover everyone. An ideal immersive audio setup mixes point source and line source speakers, with line source providing Layer 1 (LCR + Surrounds), and point source providing the height channels. Since line source speakers do not interact with the ceiling or floor the point source height channels are able to radiate energy down over the listening area with greater clarity.

Combining Sources for Optimal Audio

Combining both line and point sources into a single, coherent system is what Wisdom Audio has been doing since 1996. All the company’s products from the top-of-the-line Wisdom series featuring very large planar drivers in either a free-standing or wall-mounted configuration and moving through the Sage, Insight, and Superbar lines combine planar line sources with custom designed and manufactured point source drivers to reinforce the low end.

Why? While a line source planar driver creates a coherent and clear sound source, the planar driver must get fairly large (such as in the Wisdom series) to reproduce lower frequencies. Combining planar drivers with point source subwoofers, is literally the best of both worlds in an audio sense.

Consider the Environment to Determine the Source

Which speaker is best? It depends on the environment. Line source speakers are ideal for spaces with marble, ceramic, or wood floors; reflective or high ceilings; large, open spaces; highly reverberant room, and multi-row media rooms or cinemas. In these applications, line sources enable listeners to hear more audio directly from the speaker and less of the room. Every seat becomes a coveted money seat with each listener hearing nearly identical volume and clarity. Point source speakers span a range of prices to fit any budget, making them a popular option for many dedicated home theater rooms where radiating sound reflections can be controlled by carpet and acoustic treatments that fit within a plush, dark cinema design. They also are the perfect complement for listeners looking to add that immersive capstone to their theater, receiving the benefits of line source speakers in the LCR channels and point source in the height channels.

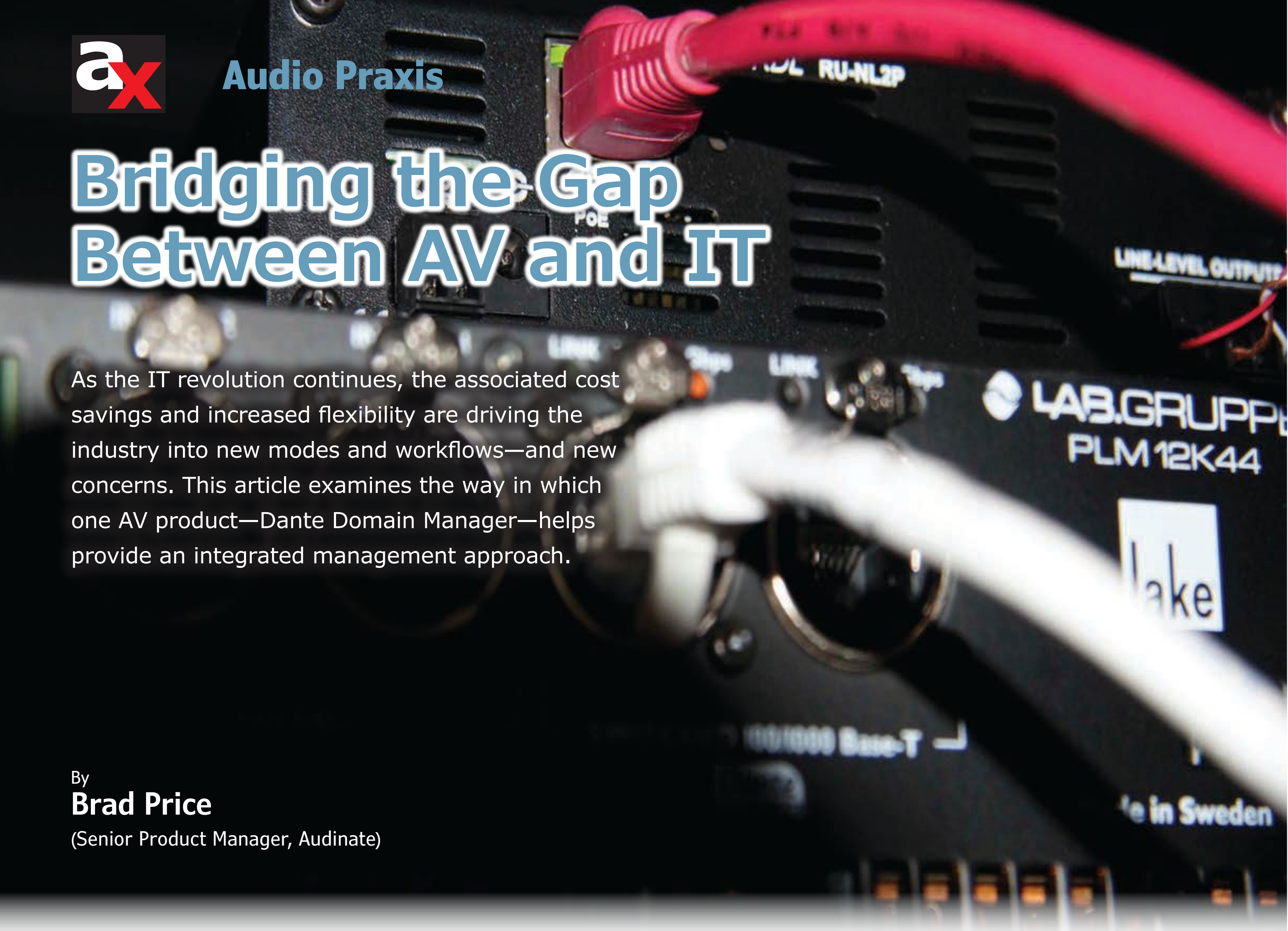
About the Author

Luc Guillaume, managing director, Wisdom Audio, has extensive experience in the audio industry, which he leverages to drive Wisdom Audio’s product development, new business opportunities, and connection to customers and markets. Luc holds degrees in economics and audio engineering and entered the audio industry in 1994 as an assistant recording engineer. He spent six years on the recording side and then joined JBL, working on its Synthesis product range outside of the US. After seven years at JBL, he served as VP of sales of Screen Research before joining Switzerland-based integrator SmartHome, ultimately becoming its CEO.

Resource

Wisdom Audio, http://wisdomaudio.com

Bridging the Gap Between AV and IT



As the IT revolution continues, the associated cost savings and increased flexibility are driving the industry into new modes and workflows—and new concerns. This article examines the way in which one AV product—Dante Domain Manager—helps provide an integrated management approach.

By
Brad Price
(Senior Product Manager, Audinate)

For much of the world, the IT revolution is old news—a decades-long transition from paper to screens, from letters that require days of travel to emails in a fraction of a second. With incredible leaps in speed and capability, the same technology that connects a PC to a printer now carries hundreds of channels of real-time, uncompressed audio alongside all the other data that an AV system requires. The cost savings and increased flexibility are driving the industry into new modes and workflows—and new concerns.

The Amphenol adapters from analog audio to one RJ45 Dante input are shown in a molded housing. They are ideal for powered speakers.

Unsurprisingly, with this transition comes the need for new tools and models of use. In highly configurable systems that are touched by many users, how does one maintain functionality? How are users kept from making mistakes that can impact entire groups? How can a system be quickly checked for proper operation? Or restored after a calamity? Fortunately, the IT world has long ago developed robust models for managing networks, providing the AV newcomers with an ample roadmap.

Integration with Legacy Infrastructure

Things may seem to change rapidly, but in fact, many older solutions are bound to remain with us for some time to come due to the investments made or simply because they still work well. While legacy infrastructure may seem to hinder a migration to IP, there are many solutions that seek to bridge this gap.

In addition to option cards from manufacturers that may often be used to add network functionality to their products, there are countless adaptors and format converters on the market that allow older analog or digital products to become at least partially "network enabled." These range from small devices such as the Amphe-Dante single-channel Dante-to-analog convertor (useful for powered loudspeakers) to larger products that can process up to 64 channels of point-to-point MADI data and convert it to Dante networked audio.

Building New AV Networks from Scratch

When building an AV network from scratch, one has the opportunity to choose from a variety of audio over IP (AoIP) gear, and to design the network to specifically accommodate AoIP needs. Planning ahead will result in fewer headaches and better performance from the start.

Fortunately, AoIP networks are not exotic or unusual. They are identical to ordinary data networks and follow all the same familiar rules, the only difference being adjustments that optimize real-time data transmission. When designing a network, consider the use cases that could result in the highest bandwidth use and make certain that connections between devices and switches are adequate to handle that case. This is true for both audio and data networks, but especially consequential when dealing with real-time media.

Another manner in which AoIP solutions differ from common network traffic is with regard to multicast. Multicast is a mode of IP transmission that sends identical data to many receivers at once, and it is often tightly regulated so as to decrease the probability of unwanted traffic congestion. In real-time media applications, multicast forms an essential part of the system that enables many devices to be kept in tight time synchronization, and so must be permitted. This is not a problem, but rather an adjustment that must be made to switches on the network.

Security should be considered at the project's start, not as a bandage after the fact. Security takes several forms—one is to ensure that devices are not left exposed on external networks such as the Internet. Another is to limit the actions that users may take on the system to prevent accidental or unwanted changes. The former is best addressed by a careful consideration of network topology, uses, and security on individual devices. The latter requires a directory system that manages users and devices (e.g., Microsoft's Active Directory, open-source LDAP, and Audinate's Dante Domain Manager for Dante AoIP networks).

AV in an IT World

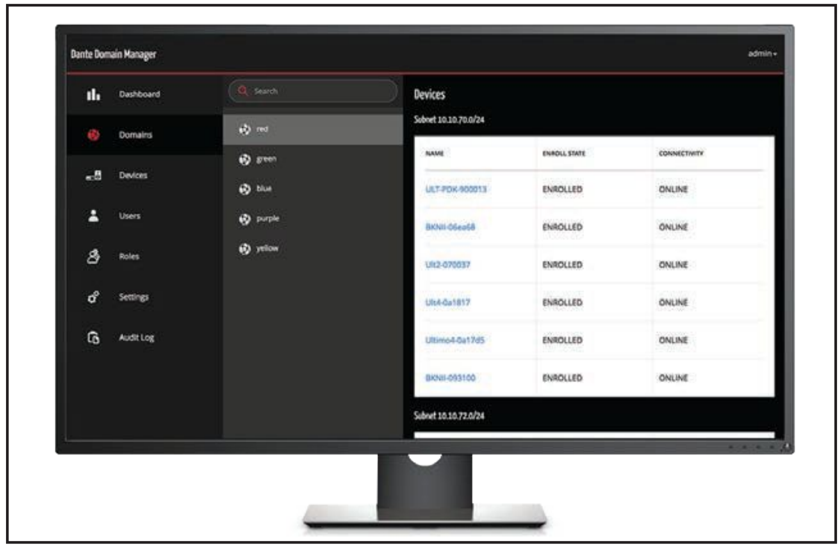
Until recently, AV professionals did not know much about IT, and vice versa. Uncertainty breeds fear,

Dante Domain Manager gives AV administrators true IT-level device management. A dashboard view provides critical information at a glance.

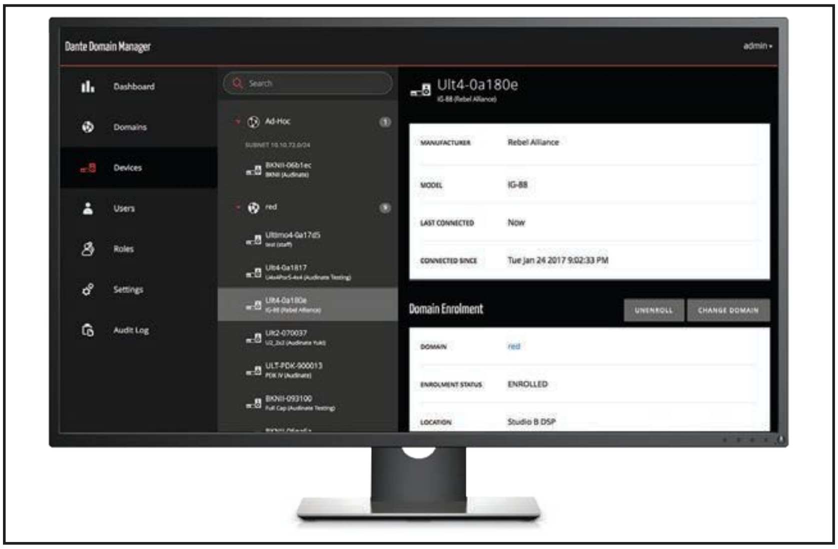
and as a result, both sides were hesitant to venture far into the other's territory. AV professionals feared that IT would be complex and inscrutable, and that performance might not match the best of analog and point-to-point systems. IT professionals are unfamiliar with AV gear and systems, and feared that AV traffic might negatively impact the critical systems they are charged with maintaining (see the section above regarding multicast traffic).

The good news is that AoIP does not have to be difficult. For IT, it means becoming familiar with new devices and requests, but the products all behave in entirely predictable ways. For AV, the expertise in real-world use of these systems remains as valuable as ever—only the method of connection is changing. Mics are still mics and loudspeakers are still loudspeakers.

IT will ask AV to be as security-minded as they are,



Devices are easily added to or removed from domains with an intuitive interface.

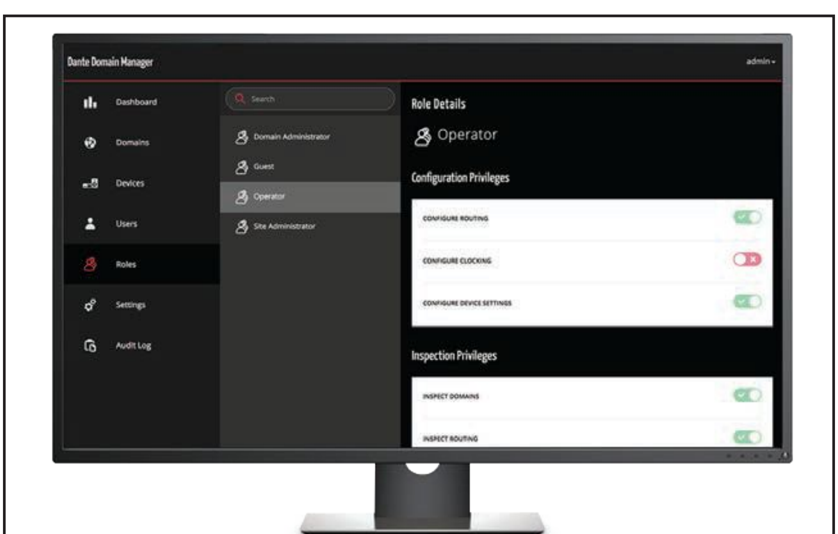


Dante Domain Manager allows users to see details about any connected Dante device.

and for good reasons born out of long experience. AV networks can easily operate under the same secure conditions as any regular network, and tools such as Dante Domain Manager extend the metaphor of user permissions and roles to the AV devices directly.

Defined Roles

Most organizations define roles for employees, so that each person knows the boundaries of their work and responsibilities. In IT, roles are enforced by rule sets that ensure that people and devices may only access items to which they have explicit access in the context of their role. This maintains both the security and functionality of the network, and prevents users from making dangerous or unwanted changes.



Pre-set roles are easily assigned to users on a global or domain level.

Typical roles might include domain administrators, local administrators, and users with various levels of access to files and resources. The importance of role enforcement in IT is evidenced by the wide array of robust tools available for the job. Systems such as Microsoft's Active Directory form the backbone of many networks, assigning permissions and capabilities to every user and machine. These systems allow computers and users to be arranged into useful groups that share resources, while preventing access outside those groups.

For the most part, AV systems have not required or been capable of supporting such schemes, as they were composed of individual devices with point-to-point connections. There was no one "place" in which the knowledge of these connections might reside, other than in the imaginations of those using the system.

AoIP systems are fundamentally different. The entire network of connections is easily seen and managed in a single software interface. With that new power comes the responsibility to manage the system, ensuring that only authorized people make changes, and that these changes be logged for any necessary review. This is precisely the gap addressed by Dante Domain Manager.

Dante Domain Manager's Inspiration and Application

Dante Domain Manager was developed to bridge the gap between IP and AV network management. It integrates with existing IT authentication systems (e.g., Active Directory and LDAP), so AV managers do not need to create parallel sets of users. Dante Domain Manager then enables managers to assign specific roles and responsibilities to each user, so that control over the different areas of a Dante network can be limited to only qualified people.

Similar to the IT systems that inspired it, Dante Domain Manager enables groups of Dante-enabled devices to be grouped into "domains." Devices within a domain may be connected together by users with specific permissions, while devices outside each domain remain inaccessible. This means that users can easily be assigned to manage specific areas or zones of an AV system without risk to others.

Dante Domain Manager enables AV managers to create functional groups of devices and user assignments independent of the underlying network architecture. Previously, AV network boundaries and de facto permissions were defined by low-level technical parameters (e.g., a network LAN or subnet). Dante Domain Manager gives AV managers the freedom to manage devices and users in ways that maximize utility and security, without artificial constraint.

System Administration

Most of the advanced administrative features in Dante Domain Manager are automated. For example, the complex goal of creating clock domains that span multiple subnets is handled in a single button click, ensuring better and more consistent results than any manual configuration. Dante Domain Manager also monitors audio devices and allows for the installation of firmware updates on those devices as they become available.

User management is accomplished directly in the Dante Domain Manager web interface, where users may be added, subtracted, or be assigned different permissions. Settings allow new devices to be automatically added for later configuration, and for new users to be given any desired role without manual intervention.

Fundamental Network Considerations

At a more mechanical level of design, a network intended to carry AV traffic must be implemented with appropriate performance levels in mind. This means being aware of, and working around, potential bottlenecks and bandwidth constraints that arise from network topography and switch configurations. It means AV and IT working together to ensure that components such as switches are correctly configured to handle things universally required by AV networking (e.g., multicast traffic for clocking or QoS settings that prioritize time-sensitive traffic).

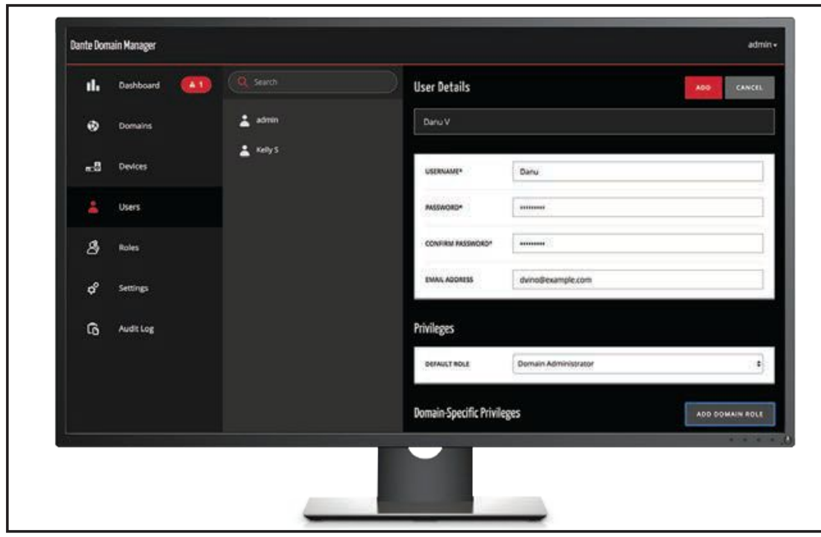
Dante Domain Manager can alert users to certain types of network failures, especially those that result in clocking problems. This can be useful in determining whether a given choice of topology or gear is appropriate and working correctly.

Users

In most IT environments, user accounts are already managed by a system such as LDAP or Microsoft Active Directory. AV management systems such as Dante Domain Manager take advantage of this fact and integrate with these existing systems, making setup of AV permissions quick and intuitive.

For administrators, this means that users of the AV network can be quickly identified and assigned roles over the different domains of AV devices—even automatically. Roles cover every need, from a top-level administrator to a "view only" member for each domain created.

For users, Dante Domain Manager authentication is direct and unambiguous, using the same familiar names and passwords used to gain general access to the computer and the network. Once authenticated, users see only the Dante domains the administrator has allowed, using the familiar interface of Dante Controller.

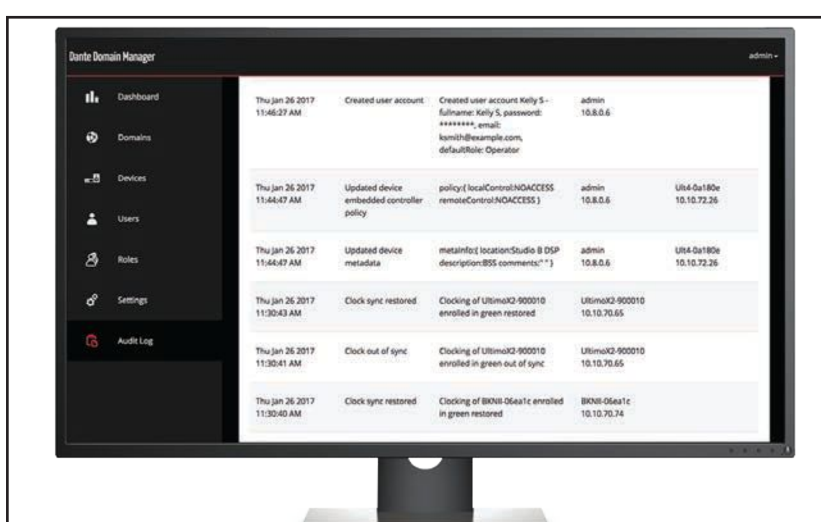


Robust user authentication and role assignment ensures that only the right people can operate or alter a system.

Permissions and Training

The permissions available within Dante Domain Manager align with the Dante system itself, and so are concerned only with changes that affect the transport of audio. There are four roles available:

- System Administrators have the ability to create, delete, and manage all domains in the system. Only a System Administrator may edit the permissions and roles of other users.
- Domain Administrators have total control over devices in selected domains, but cannot create, delete, or edit other domains.
- Operators may control channel subscriptions within a designated domain, but cannot make other changes to devices.
- Guests may view the subscriptions in a domain, but cannot make any changes.



A complete audit log allows administrators to observe all actions taken on their system. Who, what, and when are all captured and searchable.

These roles may be assigned on a per-domain basis, or applied uniformly across all domains for any given user. For example, a user may be a Domain Administrator for some domains, but only a Guest for others.

Training for use of a product such as Dante Domain Manager is primarily useful for System and Domain administrators, who need to configure the system for their particular environment. Other users need only be aware that restrictions are in place. Anyone familiar with the basics of network directory setup (e.g., LDAP or Active Directory) will find this tool immediately familiar.

Security

Security means several different things in IT systems, depending upon the type of problem one is seeking to solve.

Hacking: Devices connected to the Internet should be secured against outside access to their underlying systems. There have been several notable breaches of systems that exploited the weak configurations of IoT devices, a problem that is best addressed by manufacturers of these devices.

User actions: Many problems on networks are not the result of malevolent actions, but human error. Users make changes accidentally or without understanding consequences.

Permissions systems address the latter, providing a set of rules for available actions in order to prevent unwanted or unauthorized changes. As with all of these systems, Dante Domain Manager keeps copious logs of user activity and can provide alerts to administrators, allowing for rapid discovery and resolution of issues.

Conclusion

As AoIP becomes further integrated with existing multi-purpose networks, robust tools for management and security will become key to acceptance. No IT department in charge of an office building, radio station, or concert hall will wish to compromise or diminish the levels of management to which they have become accustomed, and they will expect AoIP vendors to match this point of view. Dante Domain Manager is the first system to address this need, bringing AV fully into the IT world.

About the Author

Brad Price is the Senior Product Manager at Audinate and has an extensive background in audio engineering, music performance, and software product development. He works with the development team to create software for Dante Audio Networking that brings value to audio professionals across a wide range of industry categories.

Introduction to Dante Audio Networking

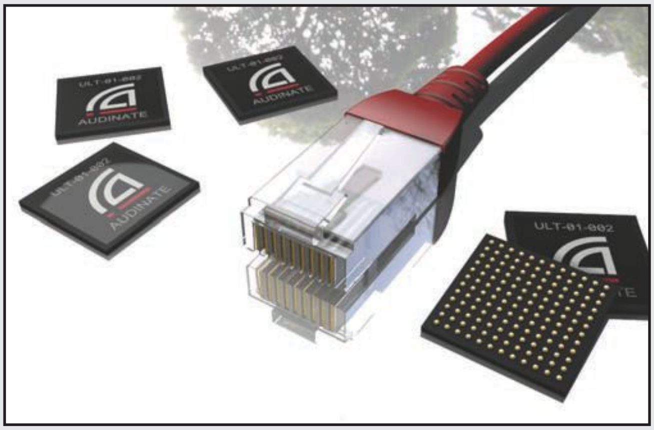
Dante is by now a familiar term in the AV industry, with more than 1,100 compatible products in the marketplace from more than 160 manufacturers. But what is Dante, exactly?

Dante is a standards-based audio over IP (AoIP) solution made by Audinate for audio equipment manufacturers. Dante encompasses network transport and discovery protocols, system-wide clocking, hardware designs, control APIs, and a suite of software products that manage and expand audio networks. The central promise of Dante is interoperability. All Dante-enabled products can connect to one another using the same methods and tools, regardless of the brand.

Dante replaces point-to-point physical connections with logical ones, greatly reducing the time required to set up or change a system. All devices and connections are managed with easy-to-use software from any point on the network. All connections are made with inexpensive, lightweight CAT5 or CAT6 cable connected to nearly any network switch.

A Dante audio network's performance rivals or bests legacy modes of analog or digital connectivity with low, deterministic latency (typically 1 ms or less), sub-microsecond synchronization, nearly non-existent jitter, and support for up to 512 bidirectional channels of audio per device on a typical gigabit network.

Because Dante is built upon network standards, it is compatible with readily available network hardware and runs alongside non-audio traffic with ease. IT administrators and AV integrators alike find Dante easy to understand and manage.



Dante Software Tools

With Audinate software, any Windows or Mac computer can manage a Dante network or participate as an audio device on the network. This allows for unprecedented flexibility when setting up an installation or recording an event, as it eliminates the need for additional hardware I/O.

Dante Controller is the free "must have" management tool for Dante networks. Version 4.0 provides compatibility with the newest features added to the Dante ecosystem, allowing users to:

- Manage channel subscriptions between all devices
- Set device and channel labels and names
- Set device parameters (e.g., sample rate, latency, etc.)
- Monitor clock stability and network latency for troubleshooting

Version 4.0 brings support for Dante Domain Manager and Device Lock, enabling users to secure, monitor, and administer Dante networks.

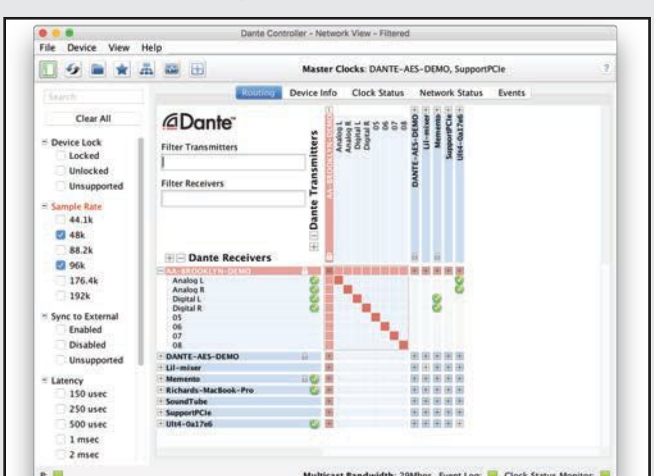
Dante Virtual Soundcard enables any Windows or Mac computer to send and receive up to 64 bidirectional channels of audio directly to and from a Dante network. Use your favorite audio applications to record events, process audio in-line, or playout music and effects.

Version 4.0 supports Dante Domain Manager, allowing computers to be securely managed as audio devices.

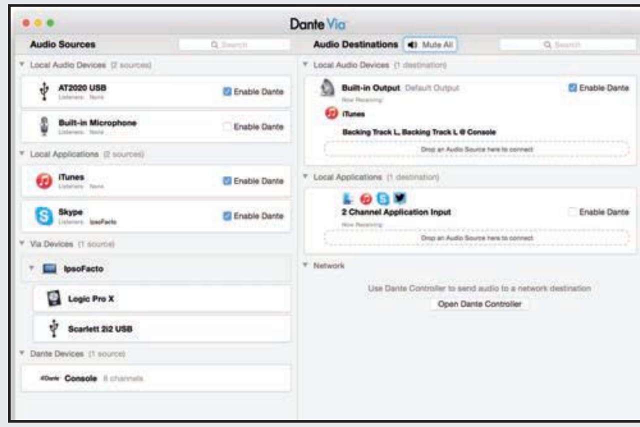
Dante Via lets customers use any computer-connected audio device or application on a Dante network. USB interfaces simply appear as "channels" of the PC or Mac in Dante Controller, and individual audio applications can be routed to different locations with ease.

Dante Domain Manager is a comprehensive network management solution for Dante AV systems. With Dante Domain Manager, multiple secure Dante networks—domains—can share physical infrastructure across subnets with no interference or clocking conflicts, allowing incredible flexibility and scalability in system design. It provides robust security for IT departments and AV managers, including user authentication and control, role administration and integration with Active Directory.

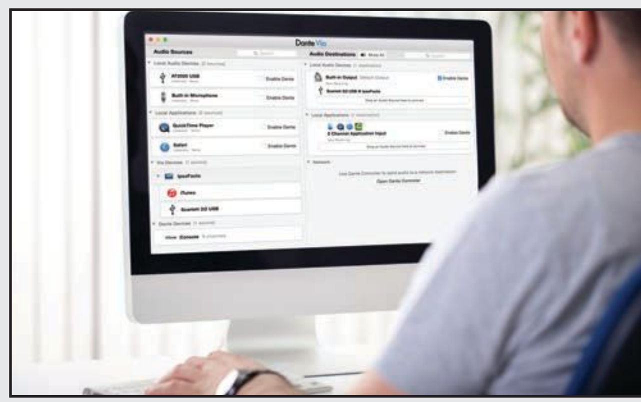
Dante Domain Manager V 1.0 is slated for release in early 2018.



Dante Controller software



With Dante Via, users can share audio from applications and USB devices, create computer-only audio networks, or connect to any Dante system. It works with Windows and Macintosh computers and supports 16-bit, 24-bit, and 32-bit audio at 44.1 kHz to 192 kHz sampling frequencies.



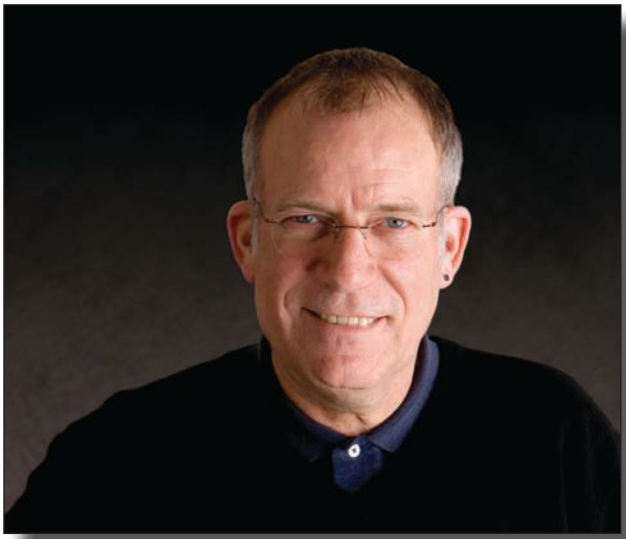
Dante Via connects all existing computer-based audio components, including USB, Thunderbolt, and PCIe devices as well as audio applications to any Dante audio network.

Dante Virtual Soundcard turns any computer into a high-performance Dante-powered recording workstation. Users can instantly connect to record, process, and play out up to 64 × 64 channels using any audio application and any combination of Dante-enabled devices and software. Dante Virtual Soundcard also allows any application (e.g., iTunes) to be used for background music, theater sound, and effects, sending audio from VoIP calls like Skype to an external audio system and using any DAW system for virtual sound checks, with up to 64 channels of playout. As with Dante Via, Dante Virtual Soundcard uses the Ethernet port already existing in the computer—no special cables or connectors needed.

Lifelong Entrepreneur and Innovator Continues to Seek New Challenges

An Interview with Tom DeVesto

By
Shannon Becker
(United States)



Tom DeVesto, an entrepreneur, audio designer, and visionary with more than 40 years' experience pioneering high-end audio products founded Como Audio in 2016.

SHANNON BECKER: Tell us a little about your background, and the companies you founded.

TOM DEVESTO: I have spent the last 40 years designing and introducing quality consumer electronics brands for consumers worldwide. Music is and will always be in my DNA. I went to Western Carolina University in Cullowhee, N.C., in the late 1960s when the Vietnam War was in full swing. I was a Political Science/English major and served in the Navy. Rock and roll was shaping the music world and I started building sound systems that ended up touring with rock bands at concerts and bluegrass festivals.

My most recent company is Como Audio, which I founded in 2016. Prior to that I was involved in the launch of several companies and brands at Advent, Boston Acoustics, Kloss Video, Cambridge SoundWorks, and Tivoli Audio. I consider myself to be an entrepreneur, audio designer, and visionary with a history of innovative, never-before tried marketing, manufacturing, and sales principles. I introduced the concept of selling factory-direct to the consumer with Cambridge SoundWorks and created a website (www.hifi.com) well in advance of the Amazon days. At the time, the skeptics said that you couldn't sell stereo equipment direct to the consumer since they had to listen to it in-store before purchasing. I knew then that stereo systems sound differently at home than they do in a store's listening room. So we made it so easy to purchase and return, if necessary, offering a 100% consumer satisfaction guarantee. We had audio techs available to customers to help them set up their systems, and reviewed room layouts for best sound positioning. We even sent our audio technicians to customers' homes to install car stereos and speakers. Customers no longer had

Solo is a hi-fi audio system that enables users to access all of their music content with the press of a button.

to drop their cars off at auto shops. This method of selling soon became the industry standard. Cambridge SoundWorks was sold to Creative Labs of Singapore and I went on to form Tivoli Audio with another unique concept for listening to music.

This time I re-introduced the concept of AM/FM radio as a form of home audio entertainment and the concept caught on. The equipment I designed with Hall of Fame audio engineer Henry Kloss featured analog knobs for fine tuning so that the most hard-to-tune-in stations could be easily found. The initial radio, aptly called Model One, was introduced in furniture-grade woods and four color ways at a retail cost of $99. At the time, no one was listening to the radio except for maybe at a backyard ball game and certainly no one was looking for a high-end, home décor table model with a retro look. The sound, design, colors, and wood finishes were the selling points and the Model One, which began as a single product on the web (tivoliaudio.com), evolved into millions of sales worldwide across 50 countries. A small country like Norway sold more than 700,000 Tivoli radios while I was CEO at the company. I ran the company for 15 years, introduced several award-winning products and Tivoli became a highly desirable iconic brand.

Tivoli Audio was sold to a private equity group in 2015 and after a one-year, non-compete, I founded my current company, Como Audio, in 2016.

Como Audio was founded with one singular goal in mind—and that is to make all of the music content accessible on one device, simply with the press of a button. I set out to make listening to music from an iPhone, Spotify, Bluetooth, the Internet, or FM Radio easier with the multi-room, hi-fi audio systems that let you access all of your music content with the press of a button. With Solo and Duetto, there is no need for an app or remote control to listen to music from all content sources. So with a minimal amount of space and advanced technology, Solo and Duetto make accessing music content easy and simple. No longer will telephone calls have to be interrupted because music is playing from a phone. Because today music is everywhere, with Solo and Duetto, the music is accessible at any time and in any room.

Solo has a custom digital signal processor (DSP) with a digital amplifier for its great sound. It comes with six pre-sets for your favorite music sources, a 2.8" color display for Artist/Song meta

Amico, the newest high-fidelity design from Como Audio, was brought to market with all of the same features as the Solo in a portable, battery-powered, and lightweight system featuring great sound with eight hours of playback.

Musica, shown here in Hickory, looks as good as it sounds and comes in Italian-designed, furniture-grade finishes with retro anodized or painted metal knobs and control panel.

data and available album art, four-layer voice coil, 3″ woofer, and 60-W RMS amplifier with 3/4″ tweeter, dual independent alarms, Internet Radio accessing 20,000 stations including Podcasts, multi-room capable, Bluetooth with aptX, is Google Cast/Amazon Dot-ready, NFC, DLNA, four high-res inputs, DAB/DAB+ available for outside North America, and has an optical input to connect audio from a TV. Solo will wirelessly stream music from your computer and comes with a remote control unit, as well as free iOS and Android "Como Control" app. Solo comes in furniture-grade wood finishes, as well as a high-gloss lacquer white. Duetto has everything that Solo has except it comes in a slightly larger model with a 3.2 color display for Meta data, with more drivers to fill a larger room with greater sound. Each product will receive updates through the Internet.

One year later "Amico," the newest high-fidelity design was brought to market with all of the same features as the Solo in a portable, battery-powered, and lightweight system featuring great sound with eight hours of playback. Also newly introduced is "Musica" accessing music content from Internet radio, Spotify Connect, Tidal, Deezer, Napster, and Amazon Music, FM RDS, Bluetooth with aptX and features AAC (Advanced Audio Coding) for better sound quality, and NFC for pairing devices. Musica also streams CDs to other Como Audio products throughout the home. Musica looks as good as it sounds and comes in Italian-designed, furniture-grade finishes of Hickory, Walnut, Piano Black, and Piano White with retro anodized or painted metal knobs and control panel.

SHANNON: You have a lot of titles—Designer, Innovator, Audio Expert, Manufacturer and Founder—which one do you find most gratifying?

TOM: I enjoyed designing products that fill a void in the marketplace. In all of my companies, I was instrumental in introducing new concepts to the marketplace for listening to great music. At Cambridge SoundWorks I encouraged the consumer to purchase stereo equipment direct from the manufacturer and set up the components in their homes. At Tivoli Audio, I introduced the consumer to a retro form of home entertainment in the table radio. And at Como Audio, I encouraged consumers to enjoy all of their music content from one design that was easy to operate with buttons and not apps. In other words, phones should be used for making phone calls and sending messages and not listening to music as a source it is not designed for.

SHANNON: Can you please discuss your design process and how it has evolved over the years.

TOM: The design process always starts with me asking what would I like to see in a music system and what is trending now. The concept for the new Como Audio products were based on what I called "passing the guest room" test. Everyone I knew had a Sonos speaker in their home to listen to music.

Tom DeVesto's first product was the Novabeam Model Two for Kloss Video, a company owned by audio inventor Henry Kloss.

I wasn't that enamored with the sound quality, but more importantly if there were guests staying at my home in the guest room, they could not listen to music without the Sonos app. I think music should be accessible at any time and from any room in the house, so it started me thinking how I could approach this void in the marketplace. That is how I came up with Solo and Duetto—the all-in-one music systems that can turn on and off with the press of a button. No music lover wants to listen to music through a smartphone or other device, nor do they want to rely on apps to play their music.

My overall goal has always been and will continue to be listening to great sounding music on devices that are simple and easy to use that also look as good as they sound. I brought audio to a home décor market by making products that rely on good design principles as well as the latest in audio technology. My design team is based in Como, Italy, and they are instrumental in selecting finishes and colors that are beautiful for any home décor.

SHANNON: What was your first audio project? Is it still in use?

TOM: My prior companies and the brands that they launched are still popular today and continue to be sold to customers worldwide. My first product was the Novabeam Model Two for Kloss Video, a company owned by audio inventor Henry Kloss. The Novabeam was the first projection TV that created a home theater experience on a big screen.

SHANNON: You recently received one of the 2017 Consumer Technology Association (CTA) Innovative Entrepreneur Awards. What did the award signify to you?

TOM: The Consumer Technology Association (CTA) awarded me "Small Business Executive of the Year" at its sixth annual Innovation Entrepreneur Awards. I was thankful to the CTA for this significant honor as a lifelong entrepreneur and innovator to the consumer electronics industry. This has been my life's work and it is an honor to be recognized by the CTA and its distinguished panel of judges.

SHANNON: What comes next for Como Audio and Tom DeVesto?

TOM: Como Audio will continue to bring innovative products to market that make listening to music a joy and appeal to all music lovers. I like to think about what is missing in the marketplace and fill that void with the next great product. We cross over all demographics from teens to seniors and all musical listening tastes, but we always try to be the next product that everyone else will copy and say "why didn't we think of that."

As for me, in addition to designing products for the audio marketplace, my passion also lies with listening to great music, boating, and spending time with my family.

You Can DIY!

High-Performance All-Valve Phonograph Preamplifier

Bruce Heran shares his high-end sound project, which has gone through several evolutions before attaining his stamp of approval. Builders will need some advanced skills to take on an all-valve phonograph preamplifier.

By
Bruce Heran
(United States)


Photo 1: This is the rear view of the phonograph preamplifier.

About eight years ago I was challenged to build a phonograph preamplifier. It took many tries to get one that even sounded reasonable and many more tries to get one that was hum and noise free. The design detailed in this article is the most recent and likely final one in the process. It is similar in architecture to its numerous predecessors but has many refinements (see **Photo 1**).

This preamplifier can provide excellent performance, but the design assumes the builder will have more than entry-level experience with high gain valve construction projects. It is not possible in the space of a short article to cover all the details involved with the layout and wiring particularly with respect to circuit grounding.

Author's Note: This project utilizes electrical voltages that potentially could cause fatal injuries if not properly handled. The author presents the project in good faith but cannot be responsible for results that might arise from building the project.

The Design

My design goals from the start were to use only tubes in the active circuitry, have close compliance with the Recording Industry Association of America (RIAA) equalization curve, a high level of overload capability, and most importantly to me, it had to be quiet. There are many phonograph preamplifier designs but as far as I know none using the architecture that this one does. The individual portions of the architecture are not proprietary, but there does not seem to be anyone who has combined them the way I did (see **Figure 1**).

Both main gain stages are "totem pole" designs. Depending on how you define the operating points, they could be called either series regulated push pull (SRPP) or actively loaded common cathode stages. I call them SRPP as it is, in my opinion, closer to how they operate. Others may differ on this detail and in the long run it doesn't affect the performance. This type of gain stage was selected for a number of reasons. The primary one is I was familiar with how it behaved when I use it as a driver stage in power amplifiers. There are numerous references on the specifics of this type of gain stage and the particulars are beyond the scope of this article.

This preamplifier exhibits a considerable amount of time and effort to achieve the initial goals. The first stage has several functions. It provides the load for a type moving magnet (MM) or moving iron (MI) cartridge. This is set at 47 kΩ, but without causing any problems can be adjusted over a wide range to suit cartridges requiring different values. The

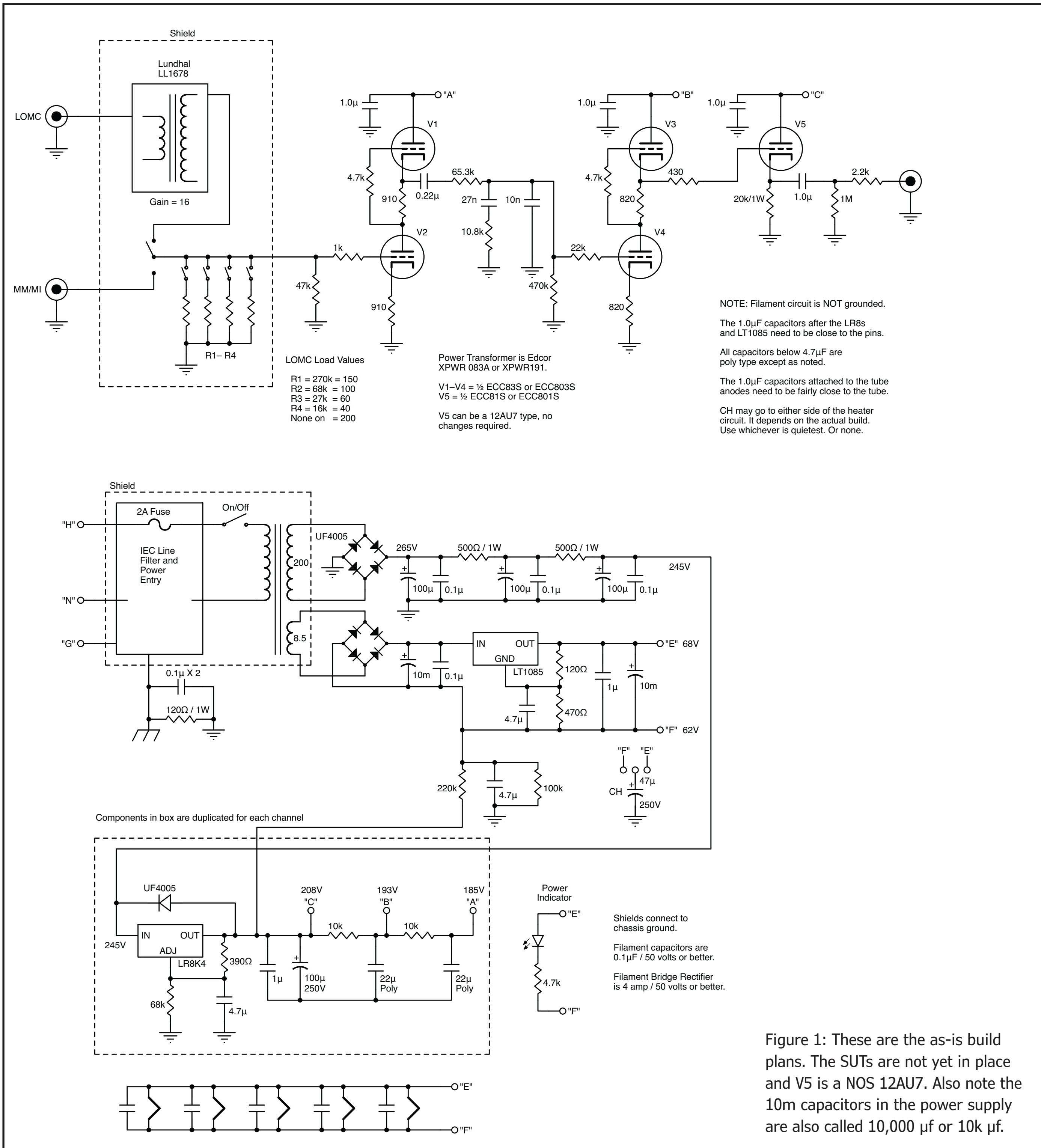


Figure 1: These are the as-is build plans. The SUTs are not yet in place and V5 is a NOS 12AU7. Also note the 10m capacitors in the power supply are also called 10,000 µf or 10k µf.

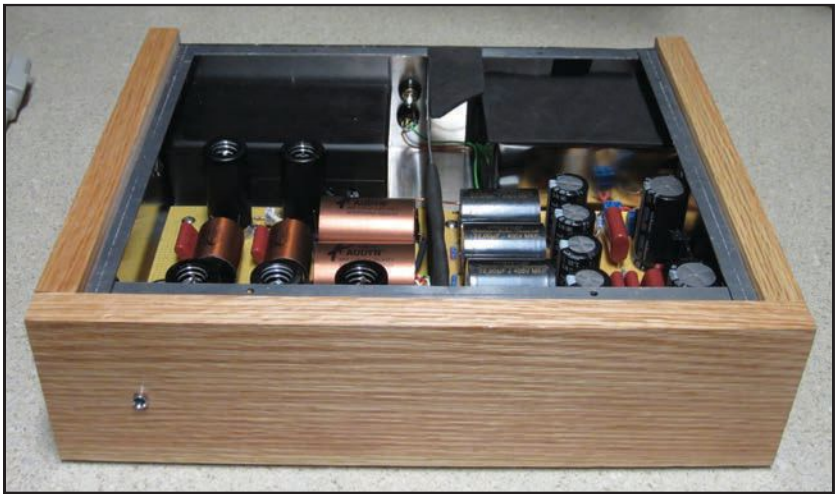


Photo 2: Front view with top cover removed and internal shields in place

stage provides about 33 dB of gain. This is a rather modest amount and was selected to insure that there is virtually no way any normal cartridge could overdrive it and cause distortion. Finally, the stage provides a stable impedance to the equalization network that follows it. This is important as the equalization is entirely passive and the impedance feeding it enters into the equation.

I indicate specific tubes for use in this stage that will match the requirements. Other tubes can be used, but a particular resistor value may need to be adjusted. The resistor is the one following the first stage and is in line to the RIAA part of the network. The value of the resistor plus the anode impedance of the tube used must equal 89 kΩ. The second stage receives the signal from the equalization network and amplifies it approximately 33 dB. It feeds a directly coupled cathode follower stage. That stage translates the relatively high output impedance of the earlier stage to a much lower one that will be able to drive the cables and eventually a line stage preamplifier for the sound system.

The output is sufficiently low in impedance and is capable of driving many passive volume controls and line stages as well. This tube can be either a 12AT7/ECC81/ecc801 or a 12AU7/ECC82/ECC802. The performance in the circuit is quite similar. The preamplifier in the project uses a new-old stock (NOS) Sylvania 12AU7 tube. The choice is up to the builder. The final value for the operating point for each gain stage is approximately 0.7 mA. The cathode follower stage operates at approximately 8 mA.

The coupling capacitors I used are all Audyn True Copper capacitors. They seem to be slightly more detailed than any others that I used in the earlier versions of the preamplifier. They are physically quite large and surprisingly somewhat sensitive to hum pickup. The outer foil layer appears to be capacitively coupled to the lead on the right end when you can read the lettering in correct orientation. Putting a finger on the interstage capacitor causes an immediate increase in hum by a significant factor. This could be an issue in some builds. If you have a favorite brand, they can used as well.

Construction Details

I built the preamplifier upside down in the chassis using the base plate (that was louvered) as the top. I build most of my gear, including this preamplifier, in a modular manner as it is considerably easier to trouble shoot or modify (see **Photo 2**). Many interconnections are made with "Euro" style terminal connectors. Hard wiring can certainly be used as can a single board. Internally, I used Mu metal shields between the power supply board and the active circuit board. Mu metal was also used as a second covering over the existing steel transformer cover. In this configuration, there is no hum at any volume setting when this preamplifier is used in my main system. Not using the extra shielding is possible, particularly with a larger chassis, but the amount of degradation in signal-to-noise performance is unpredictable.

I used separate LR8K4 high-voltage regulators for each channel to set the voltage to the third stage at 208 V. However, a range of 204 V to 215 V is satisfactory. The high voltage filter system has two CRC stages in front of the LR8s. There are two more CRC stages after the LR8 as well. The final two sections of the filter use 22 µF Jantzen Cross Cap capacitors. All other sections used electrolytic capacitors with poly bypass capacitors. A 4.7 µF poly was used on the adjust pins of the LR8 regulators. This does slightly hamper the regulating effect but since all stages are Class A, this is not an issue. It does, however, increase the filtering effect much as a capacitor multiplier circuit would.

Since noise is the primary concern, this was deemed a good trade-off. The heater circuit is regulated by a LT1085 three terminal regulator. The heater circuit is not grounded, but raised by approximately 60 VDC to protect the tubes. In this type of circuit, there is a danger of exceeding the voltage ratings of the tubes between the heaters and cathodes. The DC potential on the heater circuit prevents that from occurring. It also seems to have a slight effect on lowering the overall noise floor of the preamplifier. It appears to scavenge some stray electrons that don't make their way to the anodes. There is another capacitor in the heater circuit that may or may not be beneficial. It depends on the actual build. It is a 47 μF electrolytic capacitor that goes from the heater circuit to signal ground. When beneficial it can reduce wideband noise in the preamplifier by about 2 dB.

Unfortunately, it seems unpredictable as to which side of the heater circuit to use or indeed if either side will work. In six builds of this design and similar preamplifiers, it worked in four of them and it was a 50-50 split on which side of the circuit worked better. What I do now is put three jumper pins on the board with the center one going to the capacitor and the end pins going to each leg of the heater circuit. This way I can test the capacitor easily by using a small jumper similar to those found in many computers to short out each choice in turn.

The RIAA network that I used is slightly different from the calculated one but provides the correct response when measured on an oscilloscope doing a Bode plot. Response was down 3 dB (into 47 kΩ) at 10 Hz and 30 kHz. Between 30 Hz and 15 kHz (the

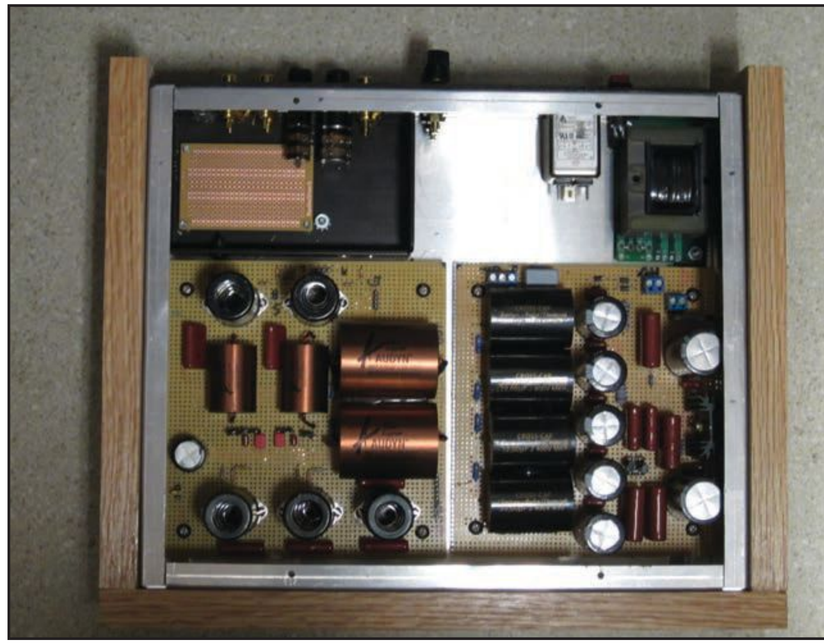


Photo 3: Here is the layout of the major components.

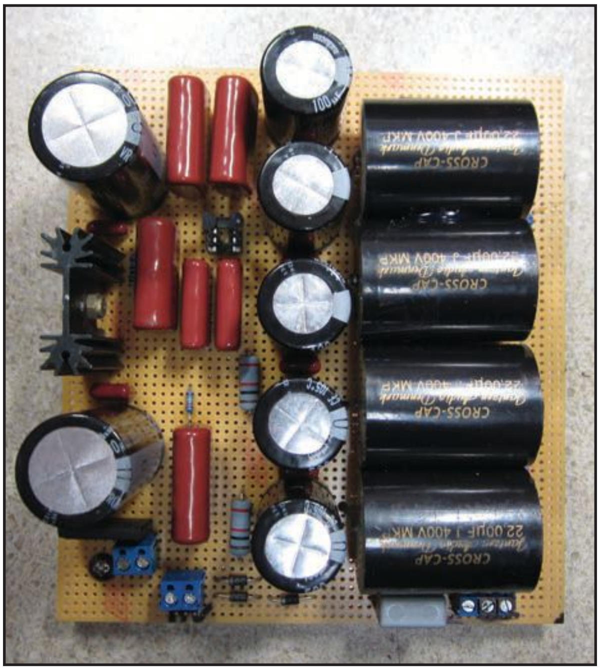


Photo 4: This is the power supply board for the phonograph preamplifier.

standard for RIAA), it deviated by less than 0.25 dB. Deviation at 20 Hz and 20 kHz was under 1 dB. I was unable to measure overload. Using a reverse RIAA filter on the input, I was able to drive the preamplifier to 700% of the nominal value without noticing any abnormalities on the scope. This would be roughly equal to a cartridge that delivered 35 mV (a normal one is typically rated at 5 mV).

Wideband signal-to-noise measurements were limited by the noise floor in my work shop. Above 200 Hz, the preamp noise floor measured -90 dBV at the output with no input signal. Below 200 Hz, the noise floor in my shop is fairly high and typically at about -75 dBV. Measurements of the preamplifier with shorted inputs were at that level. When actually in the system, there was no residual hum at any volume level so based on comparisons with other known preamplifiers the signal-to-noise range for the sub 200 Hz portion of the spectrum is estimated to be about -85 dBV.

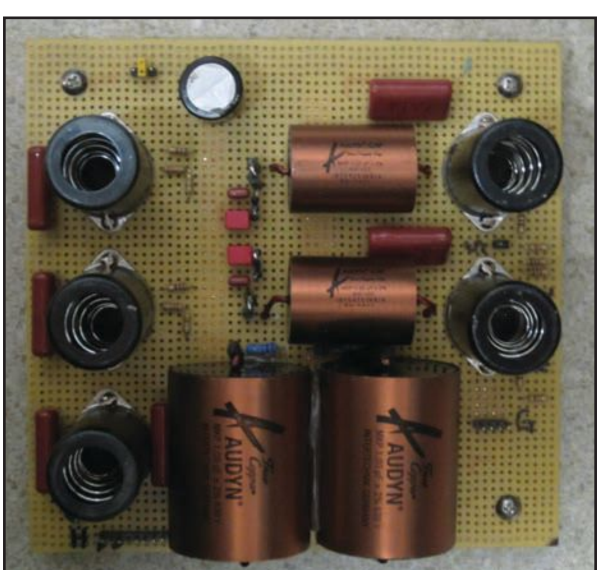


Photo 4: This is the main circuit board.

Layout Considerations

Layout is fairly straightforward with the power supply and active circuitry on separate boards (see **Photo 3**). Component layout of the power supply section is not critical and can be tailored to match the case. An external supply could be used if the last three filter stages are actually near the active circuitry area (see **Photo 4**).

The best layout I found for the active portion of the preamp is a linear one. The input goes directly to the first stage with very short leads. The equalization is on the board between stages one and two. Both channels are arranged parallel on the board. I used a large central ground between the two channels that went to a star type of configuration using the input jack ground connection as its termination.

The chassis is isolated from the ground of the circuit (see **Photo 5**). A single connection between the two is via a type X2 capacitor with a resistor in parallel with it. With the usual three wire AC main power input this provides the needed safety protection and effective electromagnetic interference (EMI) shielding.

My experience is that steel chassis seem to provide better EMI shielding than aluminum ones. I did not use boutique resistors or wire in the project. I did as mentioned earlier use high-quality coupling capacitors. You could also use premium valves, however, I found the JJ tubes to be quite excellent in this preamplifier.

Pre-Operation Verification

After construction, I recommend that you check the voltages at the midpoint of the first two stages. It should be approximately one half of the B+ applied to the tube. A variation of about 10% to 20% is satisfactory. Greater than that would indicate either a wiring error or a tube with poorly matched sections.

The voltage from the heater circuit to ground should also be checked. A range of 55 V to 80 V is acceptable. If it is outside that range, then it will indicate a wiring error and will not provide the necessary protection for the valves. If the voltages are satisfactory, then the preamp can be used in the audio system.

Operation

The preamp should be able to match most MM/MI cartridges. If you have one that requires a load different from the normal 47 kΩ, you can either change the resistor in the first stage or if the value needed is lower than 47 kΩ, you can make an adapter for the input that places another resistor in parallel with the 47 kΩ. You will need to calculate that value as a parallel combination.

An alternative is to put various resistors and indeed small (100 pf to 220 pf) capacitors inside the preamp with DIP switches that select them. This provides an easy way to change values. When operating, the preamp should not have audible hum or noise. A number of individuals that listened to some of the earlier versions initially thought the preamps were dead in the box because they were so quiet. A good build should be able to duplicate that level of performance.

How It Sounds

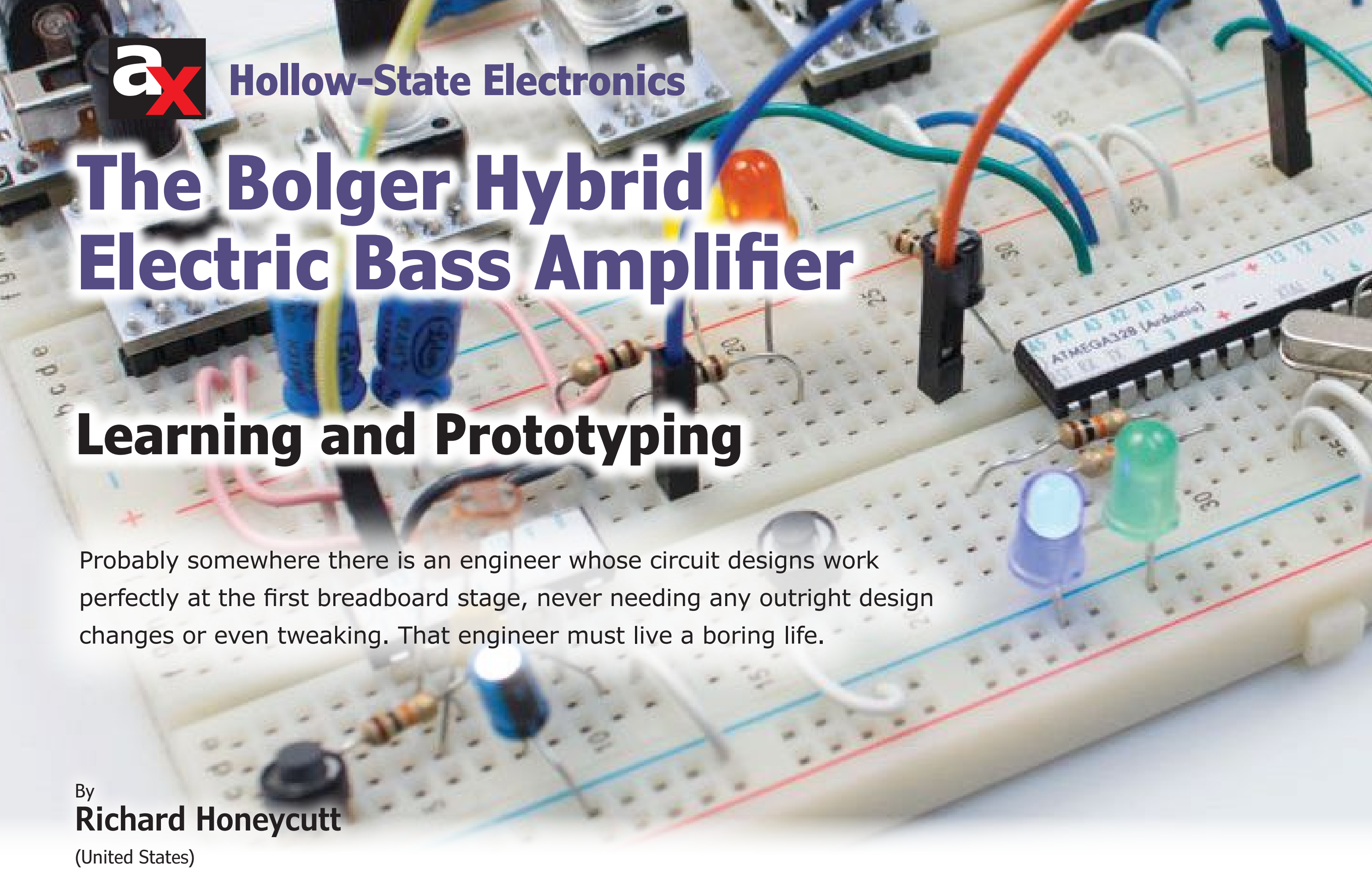
Everyone has personal thoughts on what is good or bad in sound. I find this preamp to have the best attributes of valve phono preamps by being really easy to listen to. Common descriptions do not describe the sound, but clear, sweet, and musical seem close. Being very quiet, it also wanders into solid-state territory. It is not as quiet as the one I use for reference purposes (-106 dB), but then it is not as clinical in sound as the solid state one either.

I use it with the following cartridges: Dynavector Karat23RS-MR, Hana EL, AT33PTG/II (with step up transformers), Ortophon OM30, and Grado Reference Platinum. All sound excellent in their distinct ways. I also use it with four different step up transformers: Lundahl 1678, Lundahl 9226, Silk Audio SAC, and Softone PLT-1. The latter two are not commonly known but offer excellent performance at a reasonable cost. My present one of choice is the Softone.

I feel the preamplifier is a worthy high-end sound project and will serve most listeners quite well. As I mentioned at the beginning of the article, this project benefits from a more than novice level of skill. Good listening, Bruce. ax

About the Author

Bruce Heran lives in Sierra Vista, AZ, and is an avid DIYer. He has been involved in electronics for 58 years. He built his first project —a one-transistor amplifier—when he was 13. He has a BS in Biology and studied electrical engineering for two years. Electronics has always been his hobby. In May 2011, he retired from his work as a project manager of a logistics contract with the Federal government. Now he is co-owner of Oddwatt Audio, an electronics company that specializes in vacuum-tube audio components. He is vice president of design and support, and all of the company's products are kits and assembled audio equipment based on his personal designs. He is an avid collector of vinyl records and vintage turntables.



Hollow-State Electronics

The Bolger Hybrid Electric Bass Amplifier

Learning and Prototyping

Probably somewhere there is an engineer whose circuit designs work perfectly at the first breadboard stage, never needing any outright design changes or even tweaking. That engineer must live a boring life.

By
Richard Honeycutt
(United States)

The reason we breadboard and prototype circuits before producing our final creation is that we make errors, sometimes because of things we don't know and sometimes in spite of things we do know. In the former case, we have an opportunity to learn, and that's part of what makes an electronics hobby so much fun!

In the November and December 2017 Hollow-State Electronics articles, we started the design for a hybrid practice amplifier for electric bass. Now is the time to see how well our design worked out, and whether it presented any learning opportunities.

In order to keep size and weight down, we elected to use a prefab IC-based tone/volume control module and a Class-D stereo power amplifier powered by a 24 V switching power supply. To provide a gentler onset of clipping when the amplifier is driven to its maximum, our plan was to use two tube stages—a common-cathode one and a cathode-follower—to gently limit the signal before it gets to the ICs.

The Circuit Rationale

Figure 1 shows the circuit as it had evolved by the December 2017 article, "Circuit Design for

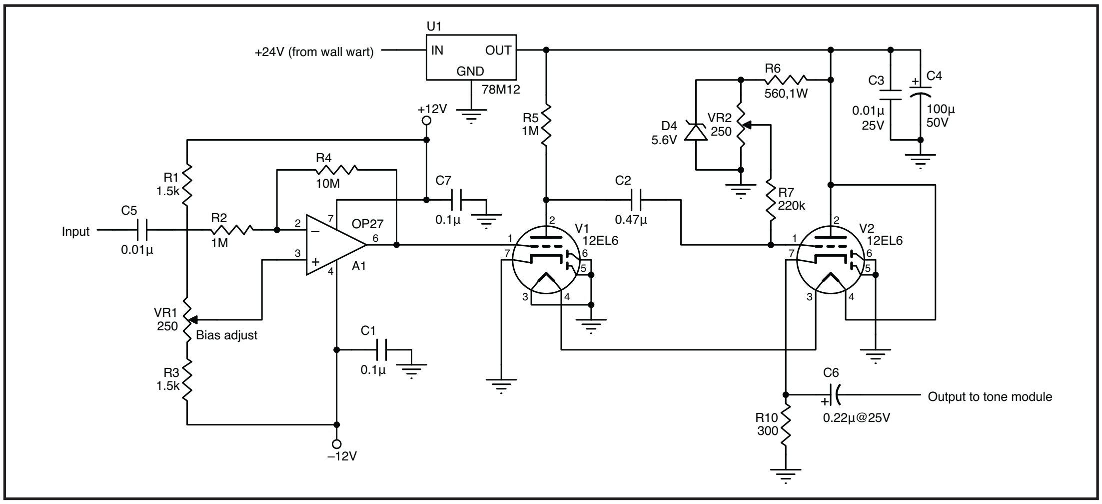


Figure 1: This circuit is the starting point for our prototyping adventure.

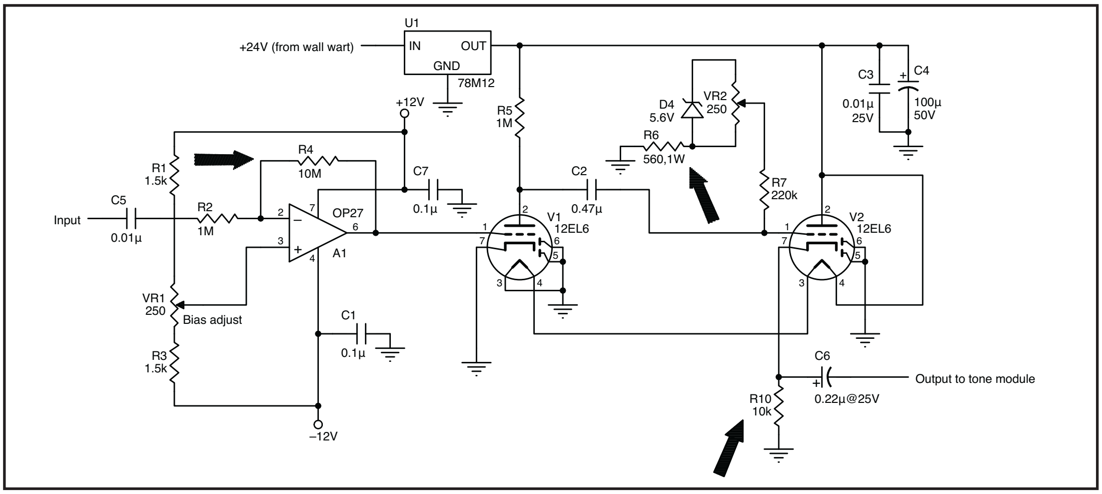


Figure 2: The cathode-follower and its biasing circuit had to be modified.

a Small Bass Amplifier." The tubes I chose were 12EL6 dual-diode-triodes (with the diodes not used). The 12EL6 is a "space-charge" tube designed for operation with a low B+ voltage. I selected these tubes because a 12 V B+ was available by simply down-regulating from the 24 V switching supply via a 78M12 three-terminal regulator. Thus, the cost and weight of a high-voltage B+ supply could be avoided.

From previous error...uh... learning opportunities in working with low-B+ tube circuits, I was aware that the input impedance of a 12EL6 amplifier stage would not be high enough to work well with an electric guitar or bass pickup. Also, there would not be enough gain from one common-cathode 12EL6 stage to amplify the signal to the nominal one volt or so needed by the tone module. Thus, I added a high-impedance stage using an OP27 op-amp, designing for an input resistance of 1 MΩ. This stage also enabled me to create a variable biasing circuit for the 12EL6, making it easier to control the level at which the tube stages limit the signal. The +12 and -12 V supplies are taken from leads soldered to the output terminals of the positive and negative three-terminal regulator ICs that are part of the tone/volume module.

As a side note, space-charge tubes have a lot of grid current, so the grid resistor must be quite low in order not to bias the grid too negatively. Cathode biasing is definitely out of the question. Pull-up biasing or voltage-divider biasing will work (Think: biasing a bipolar transistor), but using a potentiometer to provide a variable DC voltage at the noninverting input of the OP27 enables one to direct-couple the op-amp output to the triode grid and conveniently adjust the bias.

So as not to unduly load the common-cathode stage by the 47 kΩ input resistance of the tone module, a cathode-follower stage was incorporated

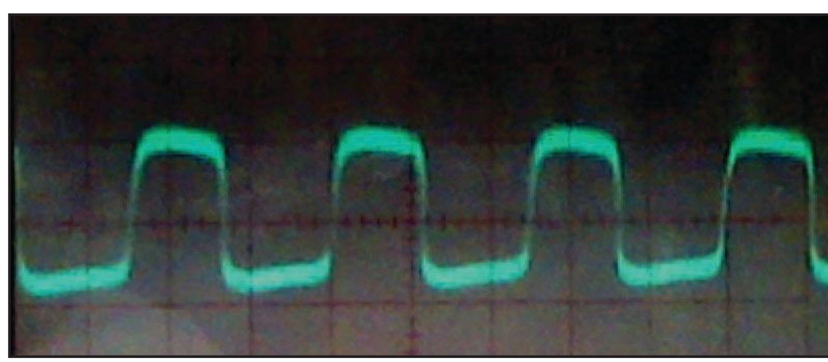


Figure 3: This waveform was produced by the circuit of Figure 2 when limiting.

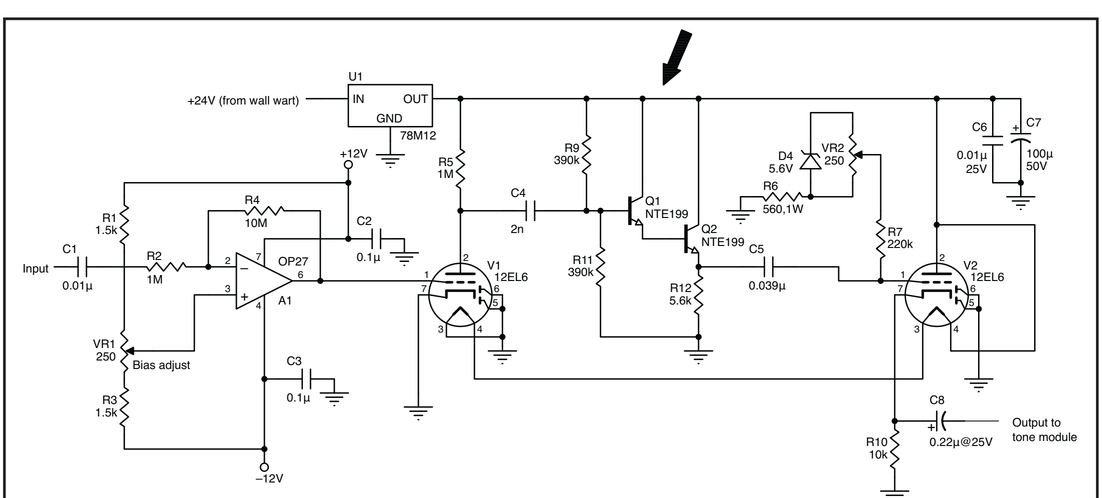


Figure 4: Another learning opportunity led to this variation of the circuit.

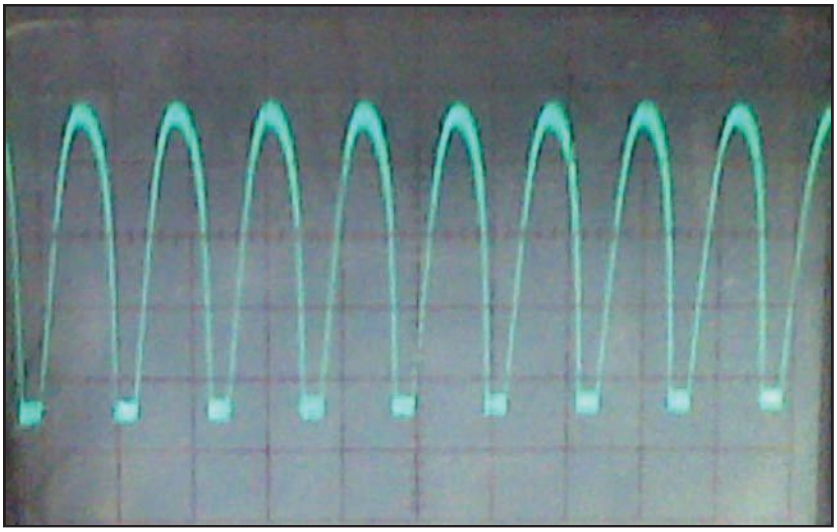


Figure 5: The final circuit has a waveform indicating predominantly even harmonics when limiting.

into the original design. Adjustable pull-up biasing enabled me to tweak the grid bias voltage so as to somewhat control the asymmetry of the output wave when the tube stages are limiting.

Circuit Modifications

When I built this circuit, I found that the original design for the cathode-follower and its adjustable biasing circuit limited the output voltage to too low a level, so I increased the cathode resistor to 10 kΩ and interchanged the positions of R6 and VR2 to provide a better bias range. Also, the onset of limiting occurred at too low a voltage, so I reduced the gain of the OP27 by changing R4 to 220 kΩ. The new circuit is shown in **Figure 2**. The limited waveform (see **Figure 3**) is nearly symmetrical—perhaps too symmetrical in that it would have a predominance of odd harmonics rather than the nice "tubey" even-order harmonics we'd like. Also, this waveform did not provide enough voltage to drive the tone module.

A bit of 'scoping around revealed that the first stage provided good gain and almost enough output voltage when it was not loaded by the input of the cathode-follower. In most hollow-state designs, we rely on a cathode-follower to provide an input resistance in the megohm range and to drive a load in the kilohm range, with a voltage gain of almost unity. But space charge triodes have such a low input resistance that they seriously decrease the gain of the preceding stage. I decided to insert a Darlington BJT emitter-follower stage to drive the cathode-follower without hurting the gain of the prior triode stage.

Figure 4 shows the resulting circuit. Having a slew of TIS 97 low-noise NPN transistors around, I used two of these. With a minimum beta of 300, the Darlington pair gives us an equivalent beta of 90,000, providing an input resistance of almost 1.95 MΩ. (The TIS97 is now obsolete, but NTE199s will work just as well; they have a beta of 800.) The output resistance of about 22 Ω can easily drive the cathode-follower stage. Both triode stages limit below the maximum linear range of the emitter-follower stage, so we don't have to worry about solid-state distortion. **Figure 5** shows the waveform from the cathode-follower when limiting. The asymmetrical limiting indicates the presence of more even-order than odd-order harmonics, which will make for a somewhat blanketed sound when the amplifier is driven hard, rather than a razzy, harsh sound.

Tweaking the Components

The new hybrid limiter stage provided exactly the limiting signature I wanted, but the decreased gain in the OP27 resulted in the output to the tone module being too low. Therefore, I changed the feedback resistors for the first op-amp stage of the tone module from 100 kΩ to 330 kΩ. As I discussed in my December 2017 Hollow-State Electronics article, changes were necessary in the capacitors that set the frequencies at which the bass and treble

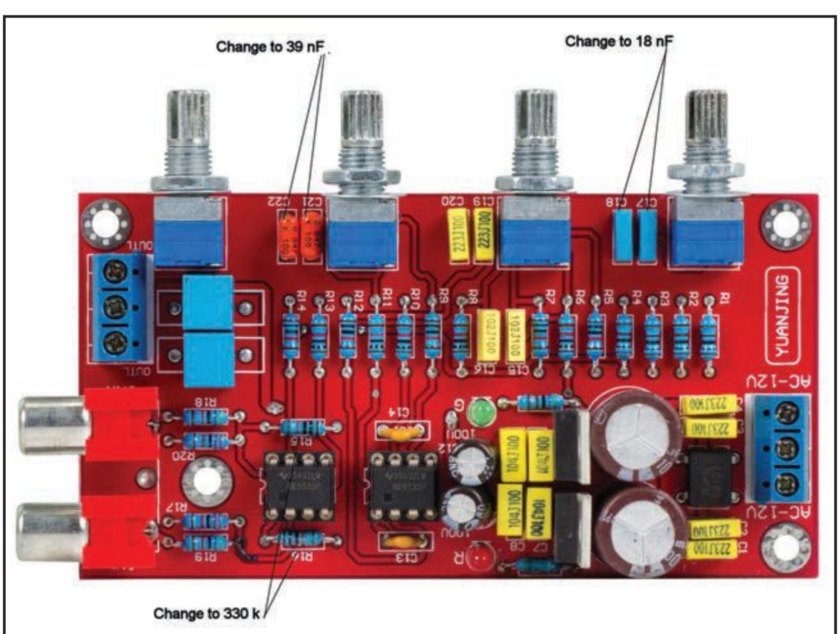


Photo 1: Six components must be changed on the Yuan Jing tone/volume control module in order to provide the desired operating characteristics.

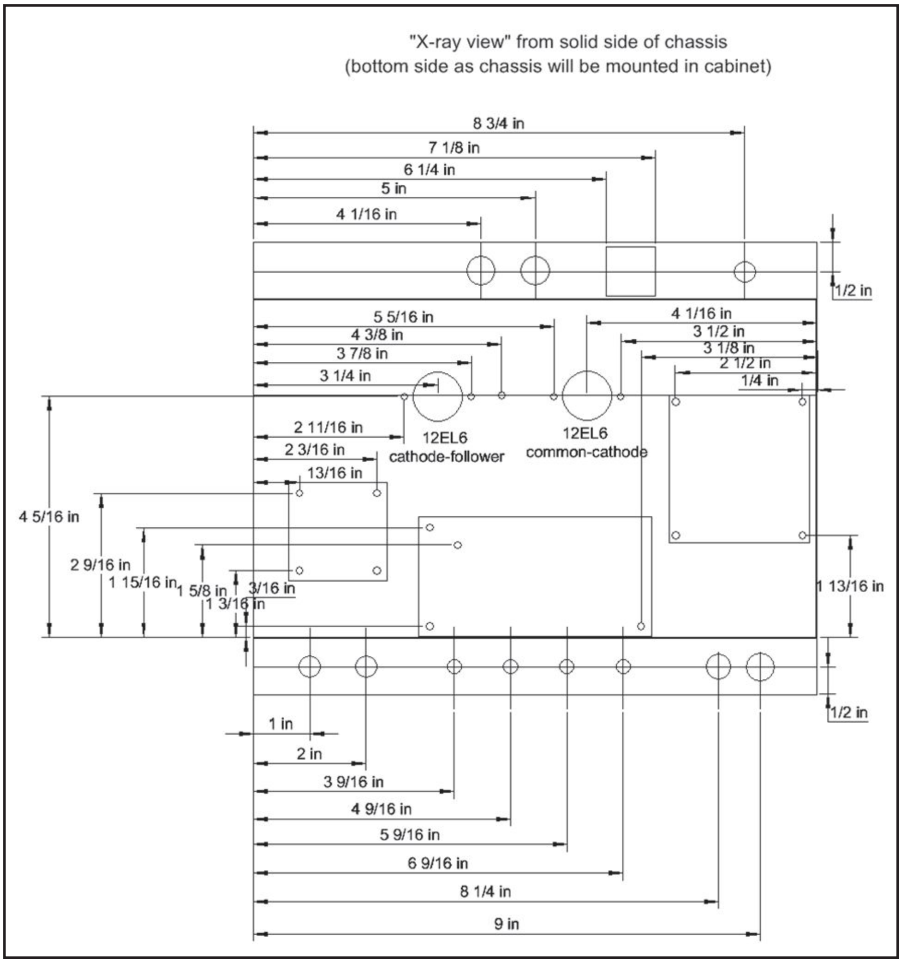


Figure 6: This is the parts and module layout as shown from the closed or solid side of the chassis.

controls act. I discovered that, unfortunately, the schematic that I received for the tone module contained several errors, so I changed the component values again. **Photo 1** shows the component replacements needed to tailor the Yuan Jing volume/tone control module to our purposes. It is important to note that the lead diameter of the through-hole components on the module are very small (under 0.6 mm), so the replacement resistors and capacitors must also have very small-diameter leads.

Chassis Preparation and Layout

Prototyping also involves chassis preparation and layout. I chose a Hammond 10″ × 6″ × 1″ aluminum chassis, with the parts and modules mounted as shown in **Figure 6** and **Figure 7**. The chassis will be mounted "bottom up" in the amplifier cabinet, meaning that the solid side of the chassis will be down. This is the view shown in Figure 6. Figure 7 shows the dimensioned chassis for machining, as viewed from the open side, so this is the view the builder will see when assembling the amplifier electronics. The 24 V CT power transformer for the tone module and the 24 V switching power supply will be mounted outside the chassis.

Figure 8 shows the power circuit, which simply switches the AC power to the primary of the 24 V CT transformer (for the Yuan Jing module) and to the receptacle into which the 24 V switching power supply is plugged. There are also a fuse and a neon pilot lamp.

With the electronics prototype complete, we only have to finish designing and building the cabinet to house the amplifier and speakers. Wait for it!

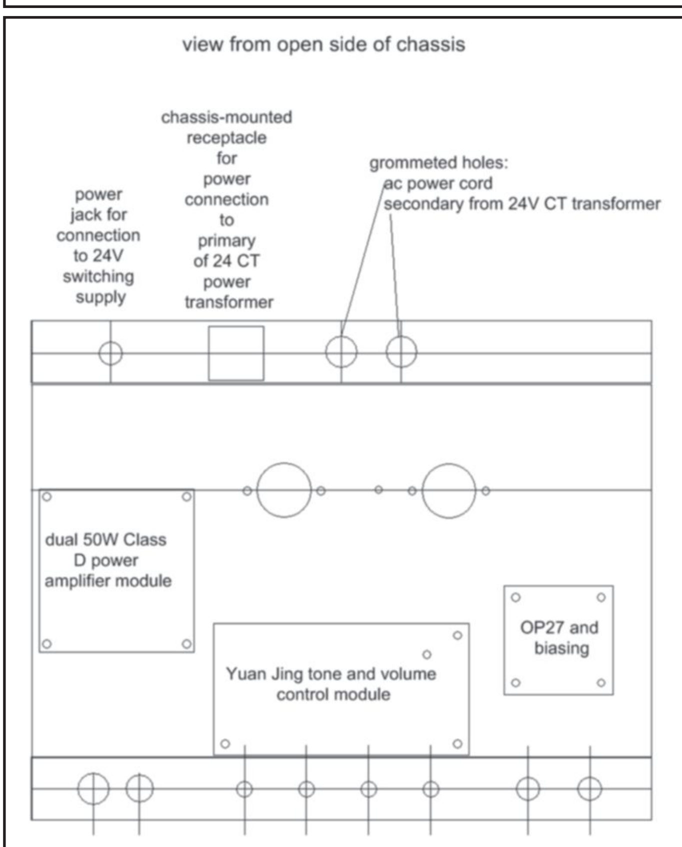


Figure 7: The parts and module layout looks like this, as viewed from the open side of the chassis.

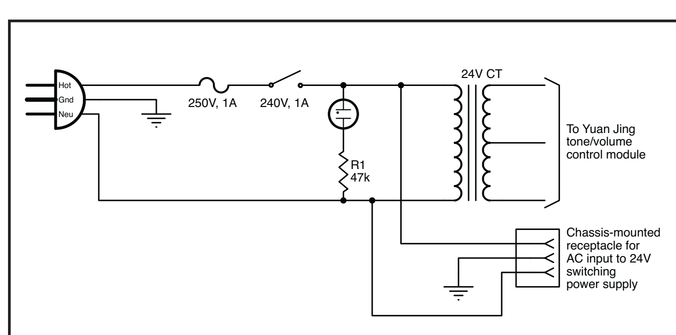


Figure 8: The power circuit is very simple.

Industry Calendar

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

February 2018
February 17–18
CanJam NYC
New York Marriott Marquis, New York, NY
www.canjamglobal.com/nyc2018

March 2018
February 26–March 1
GSMA Mobile World Congress
Fira Gran Via, Barcelona, Spain
www.mobileworldcongress.com

March 14–16
electronica China
17th International Trade Fair for Electronic Components, Systems, and Applications
China Import & Export Fair Complex, Guangzhou, China
Shanghai New International Expo Centre, China
www.electronica-china.com

March 23–25
Salon Audio Montréal/Montréal Audio Fest
Hotel Bonaventure, Montréal, Canada
www.montrealaudiofest.org

March 24–25
CANJAM Singapore
Pan Pacific Singapore, Singapore
www.canjamglobal.com/singapore2018

April 2018
April 7–8
CanJam SoCal 2018
The JW Marriott Los Angeles, L.A. LIVE, Los Angeles, CA
www.head-fi.org
www.canjamglobal.com/SOCAL2018

April 9–12
NAB Show 2018
Convention: April 7-12
Las Vegas Convention Center (LVCC), Las Vegas, NV
www.nabshow.com

April 10-13
Prolight+Sound Frankfurt
April 11-14
Musikmesse
Frankfurt Messe Exhibition Center, Frankfurt Germany
www.musikmesse.com
www.prolight-sound.com
www.messefrankfurt.com

April 13-15
Audio Expo North America (AXPONA)
Renaissance Schaumburg Hotel & Convention Center, Schaumburg, IL
www.axpona.com

May 2018
May 3–5
Consumer Electronics China (CE China) 2018
Shenzhen, China
www.ifa-berlin.com
http://www.cechina-ifa.com/en

May 4–6
Lone Star Audio Fest 2017
The Embassy Suites Dallas Park Central, Dallas, TX
www.lonestaraudiofest.com

May 7–11
175th Meeting of the Acoustical Society of America, Minneapolis, MN
acousticalsociety.org

May 10–13
HIGH END 2018
MOC Munich-Germany
www.highendsociety.de

May 23–26
144th AES Convention Milan 2018
Exhibits: May 24–26
NH Hotel Milano Congress Centre, Milan, Italy
www.aes.org/events/144

June 2018
June 2–8
InfoComm
Las Vegas Convention Center (LVCC), Las Vegas, NV
Show: June 6–8
www.infocommshow.org

June 8–10
Los Angeles Audio Show
Hotel TBC, Los Angeles, CA
www.laaudioshow.com

June 13–15
CES Asia
Shanghai, China
www.cesasia.cn

June 20–22
2018 AES International Conference on Music Induced Hearing Disorders
Chicago, IL
www.aes.org/conferences/2018/hearing

June 28–30
2018 AES International Conference on Audio Archiving, Preservation & Restoration
US Library of Congress National Audio-Visual Conservation Center, Culpeper, VA
www.aes.org/conferences/2018/archiving

July 2018
July 21–22
CANJAM London
Park Plaza Westminster Bridge, London, UK
www.canjamglobal.com/london2018

July 27–29
California Audio Show
Hilton Oakland International Airport, 1 Hegenberger Rd., Oakland, CA
www.caaudioshow.com

July 28–30
Summer NAMM
Music City Center, Nashville, TN
www.namm.org/summer/2018

August 2018
August 7–9 (pre-event August 6)
2018 AES International Conference on Spatial Reproduction—Aesthetics and Science
Tokyo, Japan
www.aes.org/conferences/2018/spatial

August 10–12
2018 Hong Kong High End Audio Visual Show
Hong Kong Convention & Exhibition Centre (HKCEC)
www.audiotechnique.com/avshow/13978
www.audiotechnique.com/av-show

August 20–22
2018 AES Conference on Audio for Virtual and Augmented Reality
DigiPen Institute of Technology, Redmond, WA
www.aes.org/conferences/2018/avar

August 27–30
SET EXPO 2018
International Technology Fair on Equipment and Services for Television Engineering, Broadcasting, and Telecommunications
Expo Norte, São Paulo, Brazil
www.setexpo.com.br

September 2018
August 31–September 5
IFA Berlin (The International Funkausstellung)
www.ifa-berlin.com

September 4–8
CEDIA 2018
San Diego Convention Center, San Diego, CA
http://expo.cedia.net

September 12–14
#MWCA18 GSMA Mobile World Congress Americas
Los Angeles Convention Center (LACC), Los Angeles, CA
www.mwcamericas.com
www.gsma.com | www.ctia.org

September 13–18
IBC 2018
RAI Amsterdam, The Netherlands
Exhibition: September 14–18
www.ibc.org

September 13–15
Prolight+Sound NAMM Russia
Sokolniki Exhibition and Convention Centre, Moscow, Russia
www.namm-musikmesse.ru | www.prolight-namm.ru

September 16–18
PLASA Show London
London, Olympia, UK
www.plasashow.com

September 23–25
Prolight+Sound Middle East
Dubai International Convention and Exhibition Centre, Dubai, United Arab Emirates
www.messefrankfurt.com
www.prolightsoundme.com

October 2018

October 5–7
Rocky Mountain Audio Fest (RMAF)
Marriott Denver Tech Center, Denver, CO
www.audiofest.net
CANJAM RMAF
www.canjamglobal.com/rmaf2018

October 10–13
HKTDC Hong Kong Electronics Fair (Autumn Edition)
Global Sources Electronics Show
electronicAsia |
www.hktdc.com/hkelectronicsfairae

October 16–18
The Loudspeaker Sourcing Show 2018
Nansha Grand Hotel in Guangzhou, China
www.loudspeakersourcingshow.com

October 17–20
AES New York 2018
145th International Convention
Program: October 17-20
Exhibition: October 17-19
www.aes.org/events/145
NAB Show New York 2018
October 17-18, 2018
Jacob K. Javits Convention Center, 655 W 34th St., New York, NY
www.nabshowny.com

November 2018

November 2–4
Capital Audiofest
Hilton Hotel at Twinbrook Metro, 1750 Rockville Pike, Rockville, MD
www.capitalaudiofest.com

November 3–4
Shanghai Headphone Festival
Shanghai Marriott Hotel City Centre, Shanghai, China
www.canjamglobal.com/shanghai2018

November 9–11
New York Audio Show
Park Lane Hotel, Central Park South, New York, NY
www.chestergroup.org/newyorkaudioshow/2017

November 12–18
Live Design International (LDI) Show
Las Vegas Convention Center, Las Vegas, NV
Exhibition: November 16–18
www.ldishow.com

November 13–11
electronica
Messe Munchen
www.electronica.de

November 14–16
Inter BEE
Inter BEE 2018 | 54th International Broadcast Equipment Exhibition
Makuhari Messe, 2-1, Nakase, Mihama-ku, Chiba City, Chiba Prefecture 261-0023, Japan
www.inter-bee.com

December 2018

No Events Scheduled