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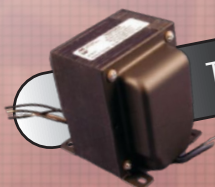
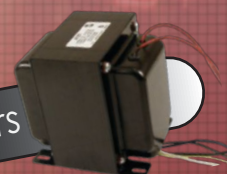


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Unpredictable Consequences

Consequences of the ongoing “trade wars” between the US and China are having a dramatic impact on imports and exports between the two countries. Back in March 2018, the US released a list of \$50 billion USD of imports from China and announced new tariffs that would be implemented. In July, a 25% tariff on \$34 billion USD of Chinese goods began, with another \$16 billion tariff to be implemented at a later date. On July 10, the United States Trade Representative gave a list of \$200 billion of Chinese imports that will potentially face an additional 10% tariff. More recently, on August 1, President Trump proposed to increase the 10% tariff on a list of \$200 billion dollars of Chinese imports to 25%.

While the US administration calls it “a dispute” justified by macroeconomic policies of trade imbalances and tries to associate the tariffs with defensible causes such as the protection of IP property, the totally unpredictable and unintended consequences are first and foremost becoming visible in the very same industries that the policies claim to protect. All audio manufacturers, not just those based in the US, rely on electronic components parts sourced from China and other regions of the world. All audio companies increasingly depend on exports around the world—many US companies have important subsidiaries in China—and the entire consumer electronics industry depends upon the global dynamics.

Of course, these completely improvised measures are starting to have visible consequences for the audio industry, with US-based companies and businesses now starting to suffer. In just one week, *audioXpress* received messages of appeal from multiple US audio companies that are already projecting severe disruptions to their business because, although they manufacture products in the US, they rely heavily on electronic components from China and there aren't any other sources for those components outside of China. As of July 6, the cost of materials and components from China has risen by 25%. Imposing a 25% tax on electronic components from China affects a large percentage of material costs, which will ultimately result in further price increases, making US-based products much less competitive.

This is just the start, with many other industry players already showing signs of retraction and adopting a cautious attitude on their short- and medium-term investment plans, including many key multinational companies, which are the main engines for most of today's innovation.

Basically, everyone is holding their breath in the hope this doesn't escalate further and that the tariffs are quickly removed.

Loud Sounds

On a lighter and more optimistic note, I received news of an interesting recent study on “Sound's differences between age groups,” promoted by researchers at the Western Research Chair in Cognitive Neuroscience (<http://news.westernu.ca/2018/08/study-sound-differences-age-groups/>). The researchers examined responses to different sounds in the auditory cortex of participants in their 20s and 60s. What they found was differences in responses to soft and loud sounds, and that our brains become more sensitive to sounds as we age—an already recognized cause of hearing challenges over a lifetime. But what the study also revealed is that when young adults are in a loud environment (e.g., a rock concert), their brains become less sensitive to relatively quiet sounds. This enables the listener to hear the relevant sounds (e.g., a guitar riff) well, without being distracted by irrelevant sounds. However, as a person ages, researchers found that older listeners become over-sensitive to sounds, hearing both quiet and loud sounds without the ability to ignore or tune out irrelevant auditory information. Without the ability to reduce sensitivity to irrelevant sounds, the individual experiences hearing challenges.

This also means that younger adults are less discriminative to less important sounds, (akin to the perceptual coding of highly compressed digital music files) whereas older individuals still stay sensitive to these relatively quiet sounds—making them over-sensitive to a loud restaurant environment, and also more demanding on the quality of the music sources they hear and the quality of the equipment they listen to. So, this means eventually millennials will also convert to high-end audio as they get older—if they can still hear at all.

I'm extrapolating, but it certainly makes sense. Just give them time.

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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2	0	04:37:54	00:02:00		IMAGE.DAT	USQD09900002	Hot Shots	Hot Shots	
3	0	10:23:04	00:02:00		IMAGE.DAT	USQD09900003	Light my Fire	Hot Shots	
4	0	10:25:04	03:41:23		IMAGE.DAT	USQD09900004	Fire Burning	Hot Shots	
5	0	14:06:27	05:48:46		IMAGE.DAT	USQD09900005	Ashes	Hot Shots	
6	0	19:58:73	06:46:06		IMAGE.DAT	USQD09900006	Cool	Hot Shots	
7	0	26:45:04	04:32:36		IMAGE.DAT	USQD09900007	Smoke	Hot Shots	
8	0	31:19:40	00:02:00		IMAGE.DAT	USQD09900008	Burning Fire	Hot Shots	
9	0	31:21:40	08:24:51		IMAGE.DAT	USQD09900009	Hot Shots	Hot Shots	



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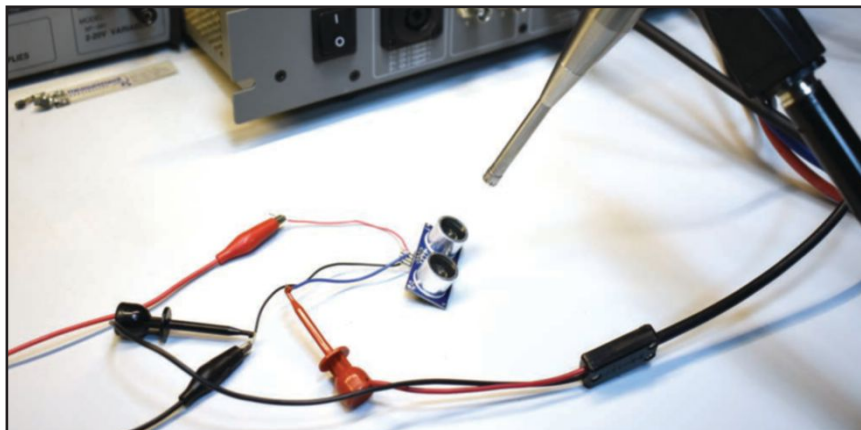
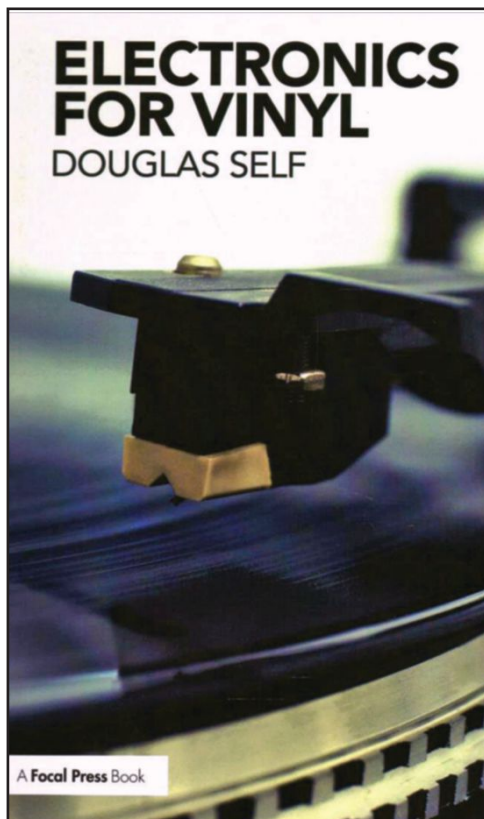
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An In-Depth Exploration of *Electronics for Vinyl* (Part 1)

Diving Into the Book

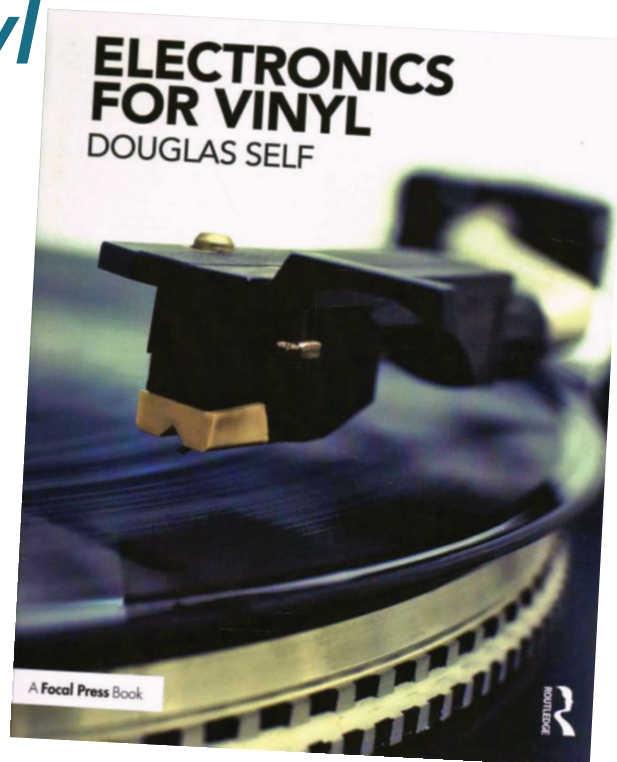
A two-part article series for a book review? That's what happens when a well-known author on turntables and phono preamps, delves into the extensive work of Douglas Self to dissect his recently published book *Electronics for Vinyl*. Gary Galo provides an extra chapter.

By
Gary Galo

(United States)

Douglas Self is a well-known engineer and author specializing in audio equipment design. His books include *Audio Power Amplifier Design* (6th Edition, Focal Press, 2013), *Small Signal Audio Design* (2nd Edition, Routledge, 2014), and *Self on Audio – The Collected Articles of Douglas Self* (3rd Edition, Routledge, 2015), and *The Design of Active Crossovers* (2nd Edition, 2018). He has written numerous articles for a variety of periodicals. A more thorough overview of his activities can be found on his website, www.douglas-self.com.

Electronics for Vinyl is Self's latest book on audio. In his Preface, Self lays out some of his own design philosophies and viewpoints. He notes that "I have no time for faith-based audio, so there will be no discussion of esoteric components with high price tags that actually achieve nothing... Passive RIAA equalization is not recommended; it is thoroughly deprecated with good reason." He is also a strong believer in double-blind testing, which he claims always shows that "...if you couldn't measure it, you couldn't hear it. Very often if you could measure it, you still couldn't hear it." There are many credible designers in the audio industry who would dispute these



Douglas Self's *Electronics for Vinyl* is more than 300 pages in length, and covers phono preamps and equalization, filtering, and a host of related topics. The designs reflect the author's preference for the 5532 and the 5534 op-amps.

views, and I would remind readers that Richard Heyser once wrote that if it measures good and sounds bad, you're not measuring the right things (or words to that effect). In the Preface, Self also notes that, of the four RIAA feedback configurations, "...I have shown that one of them requires significantly smaller values of expensive precision capacitors than the others. This new finding is presented in detail in this book, along with related techniques of optimizing resistor values to get convenient capacitor values."

Amplifier Topologies

Self begins the first chapter with an overview of basic amplifier topologies, dividing amplifiers into four categories: voltage amplifiers, transconductance amplifiers, current amplifiers, and transimpedance amplifiers. Along the way, he covers optimizing headroom and lowering noise. Self's discussion of the various types of noise encountered in audio circuits is thorough, and includes the various "colors" of noise (white, pink, and six others), as well as Johnson noise, shot noise, 1/f noise, and popcorn noise. At the end of this chapter, he has a short section on using multiple amplifiers to increase drive capability. The topology he shows involves taking the output of an op-amp gain stage and feeding it to a unity-gain

Electronics for Vinyl

by Douglas Self

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follower, then combining the outputs of the follower and the gain stage to double the output current (Figure 1.14, p. 23). Burr-Brown published a similar circuit in its applications bulletin “Double the Output Current to a Load with the Dual OPA2604 Op Amp,” written by Morgan Monks, and Monarchy Audio uses this technique to drive the MOSFET output stage of its SM-70-series power amplifiers with an OPA-2604 dual op-amp. Self doesn’t discuss the use of inside-the-loop buffers (e.g., the BUF04, the BUF634, and the LME49600).

An entire chapter is devoted to passive components. Self fully recognizes that passive components often do not perform like their theoretical models, and discusses various types of distortions and non-linearities in both resistors and capacitors. Series and parallel resistors and parallel capacitors are often needed to produce optimum calculated RIAA values, and Self’s discussions of the best methods for doing so are quite extensive. He prefers paralleling nearly equal resistor values for lowest noise; similarly, paralleling nearly-equal capacitor values will improve the overall tolerance of the parallel network, compared to trimming a large value with a small one.

Self discusses various types of capacitor anomalies, including leakage, equivalent series resistance (ESR), dielectric absorption, and non-linearity. Despite these problems, especially in electrolytics, Self believes that at the low signal levels found in audio applications, capacitor non-linearity is not detectable. He states that “Much nonsense had been talked about mysterious coupling capacitor properties, but it is all absolute and total nonsense.” He also believes that there is “no sense” in bypassing electrolytic coupling capacitors with smaller non-electrolytic types.

Needless to say, his views on these issues are at odds with those held by many in the audio community. The benefits of bypassing electrolytic capacitors with smaller, film types were discussed by Walt Jung and Dick Marsh as far back as 1980, in their *Audio* magazine series “Picking Capacitors.” Comments throughout his book indicate a disdain for the philosophies of many high-end designers. He also expresses the view, backed up with measurements, that electrolytic capacitors should never, “under any circumstances,” be used to set time constants in audio due, in part, to poor component tolerances. This point is very well-taken, and I doubt that many designers would argue.

Op-Amp Characteristics

“Opamps and Their Properties” is exactly what the title implies. Self discusses the important op-amp

characteristics, including noise, slew rate, common-mode range, input offset voltage, bias current, and various types of distortion. For comparisons, Self has picked 14 well-known IC op-amps, offering tables that compare these specifications, as well as cost. Self notes—correctly, in my view—that common-mode distortion is one of the most important, yet widely overlooked, aspects of op-amp performance. He discusses common-mode distortion in both bipolar and JFET-input op-amps, and focuses on several specific devices, discussing their strengths and weaknesses. The bi-polar devices he picked include the NE5532/5534 (dual/single), the LM4562 (dual), the AD797 (single), and the OP27 (single). The JFET devices covered include the TL072 (dual) and the OPA2134 (dual).

Readers of Self’s other books and articles know that he still prefers the dual 5532 and the single 5534—designs that go back to the 1970s—to nearly all of the more recent devices. Although other op-amps may appear to have better noise specifications than the 5532 or the 5534, Self argues that in real-world implementations the 5532 and/or the 5534 are superior. For the LM4562 he notes: “The input noise voltage is typically 2.7 nV/√Hz, which is substantially lower than the 4 nV/√Hz of the 5532...However, with MM cartridges the greater current noise means that the overall noise is higher than that of the 5532 or the 5534.” Self admits that the 5532 suffers from common-mode distortion, which should preclude its use in the unity-gain follower used in his current-boosting scheme. He notes that the LM4562 is “much more resistant to common-mode distortion than the 5532,” but he doesn’t give any op-amp recommendations for the current-boosting circuit.

Self states that it has taken nearly 30 years for a better audio op-amp than the 5532 to appear. “The LM4562 is superior in just about every parameter except for its higher current noise. This leads to an EIN with the standard cartridge of -117.9 dB, a significant 4.6 dB noisier than the humble 5534A.” But, is 4.6 dB significant when we’re comparing noise figures hovering around -120 dB? He also says that the LM4562 has “a much higher price.” Perhaps that’s true in Europe, but Digi-Key currently sells the 8-pin DIP version for \$3.11—cost-effective unless you’re a real audio cheapskate. His priorities seem focused in two areas—cost and noise, even if the latter is at the expense of other parameters.

Regarding the AD797, he notes that at 0.9 nV/√Hz, it’s a “remarkably quiet device in terms of voltage noise, but current noise is correspondingly high due to the high collector currents of the input devices, and so it is noisy when used with MM cartridges.” He

does find it more suitable to low-impedance moving-coil cartridges, where current noise will be much lower (Jensen recommends the AD797 for moving-coil amplification following a step-up transformer). He describes the distortion-cancelling feature of the AD797 as “ingenious” and provides a graph to show its effectiveness. Yet, his comments on the audibility of differences between op-amps at the end of this chapter would indicate that this feature has no audible benefit. He considers the cost of the AD797—which he claims is about 40 times that of the 5523 per op-amp section—to be exorbitant. But, his cost analysis is off the mark by a factor of two. Digi-Key currently sells the 8-pin DIP version of the single AD797 for \$11.09 US, quantity 1, compared to roughly \$1.00 for the dual 5532. This is closer to a factor of 20, not 40.

The OPA2134 was designed by Burr-Brown before its acquisition by Texas Instruments (TI). Self notes that “The manufacturer claims it has superior sound quality due to its JFET input stage. Regrettably, but not surprisingly, no evidence is given to back up this claim.” In the datasheet, TI boasts “ultra-low distortion” of 0.00008% and a slew rate of 20 V/μs as evidence. Despite the numbers, I have never been all that impressed with the OPA2134 or its single version, the OPA134. In listening tests I conducted years ago, I concluded that it was a good device, sonically, but certainly not one of the best. But, I reached that conclusion by actually playing music through the device and listening, a practice Self ridicules. Self doesn’t mention any of the LME-series devices that TI acquired from National (e.g., the excellent LME49860, which is comparably priced to the LM4562).

Again, Self expresses his disdain for those he calls “Subjectivists who have contrived to convince

themselves that they can tell op-amps apart by listening to music played through them. This makes me laugh because there is probably no music on the planet that has not passed through a hundred or more 5532s on its way to the consumer.” I’m sorry, but this is simply untrue. Yes, most pop and rock recordings made from the late 1970s onward probably passed through a slew (no pun intended) of 5532 op-amps. Most of these recordings were made with complex mixing consoles, using multi-track recorders, and subject to all sorts of analog signal processing and track bouncing before the final mix-down to stereo tape. By 1980, the 5532 had become ubiquitous in much of that equipment, and every process involved another trip through the mixing console.

But, that only represents a portion of the analog recordings issued on LP records. A large number of vinyl aficionados treasure vintage LPs made during the “golden age” of analog recording which, give or take, encompasses the mid-1950s through around 1970. These LPs were recorded on magnetic tape and cut to lacquer LP masters using tube equipment or, later, discrete solid-state equipment. Tens of thousands, or more, of these LPs were made before the 5532 was even invented.

I listen almost exclusively to classical music. Most of my reference LPs never passed through a 5532. And, I would bet that the more reputable companies pressing new vinyl records from vintage analog tapes go out of their way to avoid the “hundred or more” 5532s that Self claims were in the signal path of every recording. I’ve spent many evenings in my listening room comparing various op-amps in phono preamps, preamp line stages, and digital playback equipment. I have consistently heard differences between them. If a double-blind test fails to reveal a difference, only one conclusion can be drawn: That test failed to reveal a difference. Self obviously disagrees and devotes a lengthy section of his website to maligning those who use their ears. He’s entitled to his viewpoint, but perhaps not to the contempt in which he seems to hold anyone who has reached conclusions different from his own.

In 1991, the Audio Engineering Society (AES) made the theme of its New York convention “Audio Fact and Fantasy—Reckoning with the Realities.” At that time, the prevailing view at AES was that all amplifiers sounded the same, and anyone who thought otherwise was delusional and lacking in scientific rigor.

At that conference David Clark, a well-known proponent of A/B/X testing, gave a paper titled “Ten Years of A/B/X Testing.” During the Q&A session, he got into an exchange with Jon Risch during which Clark admitted the following: “All we come up with

About the Author

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

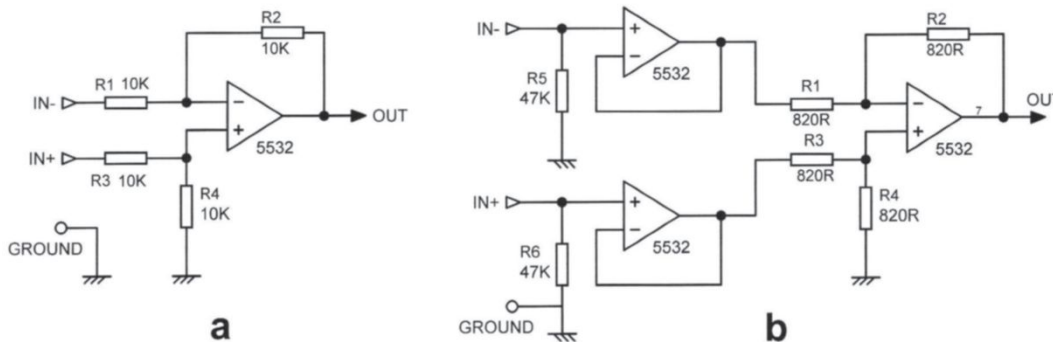


Figure 1: This is Figure 4.2 from Douglas Self's new book *Electronics for Vinyl*. The figure shows a typical balanced input stage with 10 k Ω input resistors, which can be a source of noise. His solution is to lower the value of the input resistors, buffering them with a pair of unity-gain followers to maintain high input impedance. (Printed with permission © 2018 Taylor and Francis).

on amplifier tests are negatives—that we fail to hear a difference. We cannot make the conclusion that there are no audible differences between amplifiers.” After all, you have no scientific credibility if you claim to have “proven” a negative. I wrote a detailed report on this convention, which appeared in *Audio Amateur* (January 1992), and I still have a cassette recording of that session.

Preamp Architecture

Self gives an overview of various preamp architectures, beginning with purely passive and transformer-based types, outlining the pitfalls of each approach. He clearly prefers preamplifiers with

active circuitry (I completely agree) and discusses the pros and cons of unbalanced vs. balanced circuitry, passive vs. active gain controls, the problems of gain distribution, and the integration of phono amplification into a system that also incorporates line-level sources. He claims that balanced inputs are always noisier than unbalanced inputs, “by a large margin, in terms of noise generated by the circuitry itself rather than external noise.” Self notes that: “The extra noise is due to the relatively high resistor values around the opamp, which generate Johnson noise, and also the effect of opamp current noise flowing in these resistors. Their value cannot be reduced”... (from the 10 k Ω he shows in Figure 4.2a,

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p. 82)... “without reducing the input impedances to below what is acceptable.” (For this review, we will call Figure 4.2a from Self’s book **Figure 1**).

He may be unfamiliar with THAT Corp.’s InGenius series of balanced line receivers designed by Bill Whitlock. These chips use a bootstrapping technique that raises the common-mode input impedance into the megohm region, without the noise penalty that comes from using high-value input resistors. The THAT 1200 balanced line receiver, which operates at unity gain, has $0\ \Omega$ input resistors, a differential input impedance of 48 kHz, and output noise of -106 dBu. My articles on those chips are still available on THAT’s website. It’s also fair to ask, as an example, how Benchmark Media Systems achieved an unweighted signal-to-noise ratio (SNR) of 130 dB in its superb AHB2 power amplifier, using balanced inputs.

Self’s solution is to buffer the inputs, as he shows in Figure 4.2b from his book, with unity-gain followers using 5532s—which, as he noted earlier, suffer from common-mode distortion. A unity-gain follower is one of the worst ways to use an op-amp, since the entire input signal is common mode; all op-amps suffer from common-mode distortion, to varying degrees, as Self notes earlier in the book. Why use a 5532 in this application?

He correctly notes that the output level of a phono preamp will need to be high in order to match the line-input levels from digital sources. He offers three solutions to the problem and makes a case for phono preamplifiers with variable gain. A short chapter on “Moving Magnet Inputs” weighs the plusses and minuses of active and passive RIAA equalization. Self believes that the series-feedback network is the only way to go, the only difficulty being the interacting time constants. This, of course, runs contrary to the passive or hybrid approaches favored by many high-end audio designers. He shows topologies for series (non-inverting) and shunt feedback (inverting), as well as various types of split EQ schemes using passive and hybrid passive/active techniques.

“Signals From Vinyl – Levels and Limitations” explains the operating principles of various types of phono cartridges in common use, including moving magnet, moving coil and moving iron. Self discusses various problems encountered with vinyl records: spurious signals, including noise generated by the turntable; room vibrations and low-frequency noises on the discs themselves; distortion, including that produced by the cartridge itself; inner groove distortion (which he calls “end of side” distortion); tracking error in pivoted tonearms; and stylus mistracking. Speed instability—specifically wow and flutter—are also discussed, as are signal levels encountered on vinyl

records, sensitivity of moving-magnet cartridges, along with overload problems and other amplifier limitations.

RIAA

At 42 pages, the chapter on RIAA equalization is the longest in the book. Self defines the RIAA curve with its three time constants and turnover frequencies, wisely using the nomenclature given by Stanley Lipshitz in his landmark 1979 AES paper “On RIAA Equalization Networks” (T_3 , T_4 , T_5 and F_3 , F_4 , F_5), and explains when T_1 , T_2 , and T_6 come into play. He also discusses the 1976 International Electrotechnical Commission (IEC) Amendment to the RIAA curve, which added a time constant of 7950 μ Sec, producing a high-pass filter with a corner frequency of 20.02 Hz (Lipshitz T_2). He points out the various reasons this amendment has been largely discredited, due to audible degradation of the low end, and ineffective rumble filtering. The RIAA never endorsed this, and the IEC withdrew it in 2000.

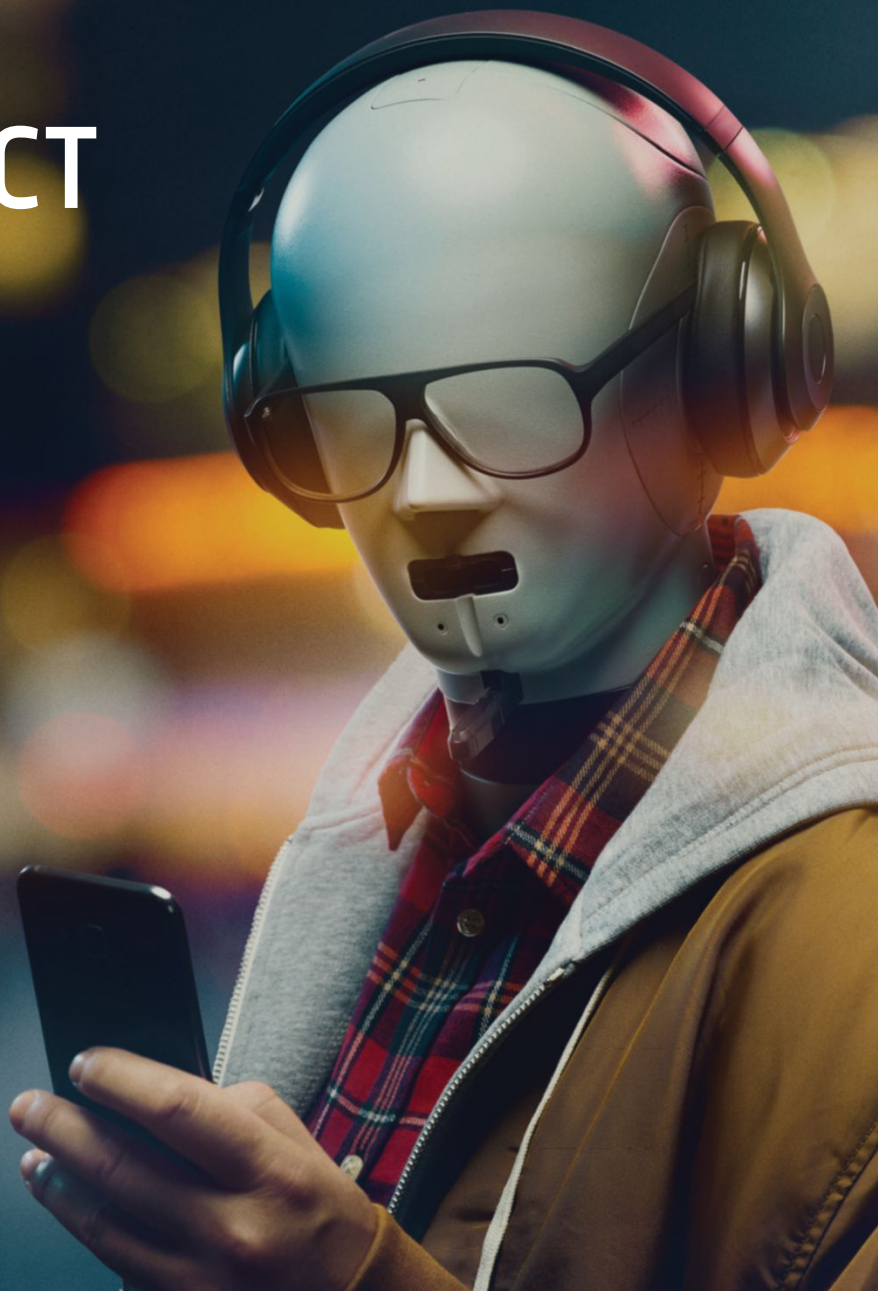
Many readers will be familiar with the mythical “Neumann” pole, an additional 3.18 μ S time constant allegedly used in later Neumann cutting heads, which changed the cutting to constant-velocity above 50 kHz. I am in agreement with Self that this is a myth. He notes that Neumann’s most popular cutter, the SAL 74B, has no such pole. Instead, ultrasonic protection is provided by a second-order low-pass filter with a corner frequency of 49.9 kHz. But, this is only one cutter made by one manufacturer, and I agree with Self that there’s no point in compensating for it because records made with other cutters are not likely to have the same characteristic.

Self notes that implementing the Lipshitz math can be daunting because some of the equations are long and complex. Sure enough! Like me, Self designed his own spreadsheets to implement the Lipshitz math, though his approach is somewhat different from my own (more on that shortly). Self describes all four feedback RIAA networks, using the standard “A” through “D” designations. He also discusses the high-frequency correction pole need for most designs (Lipshitz T_6), and implementation of the IEC Amendment, which he always makes switchable.

All of Self’s feedback RIAA designs have an electrolytic capacitor (C0) in series with the lower gain-determining resistor, which sets the low-frequency cutoff for the circuit (Lipshitz T_1) and ensures that the circuit is unity gain at DC. He strongly advises against using this capacitor to set the IEC roll-off, which is consistent with his view that electrolytic caps should never be used to set

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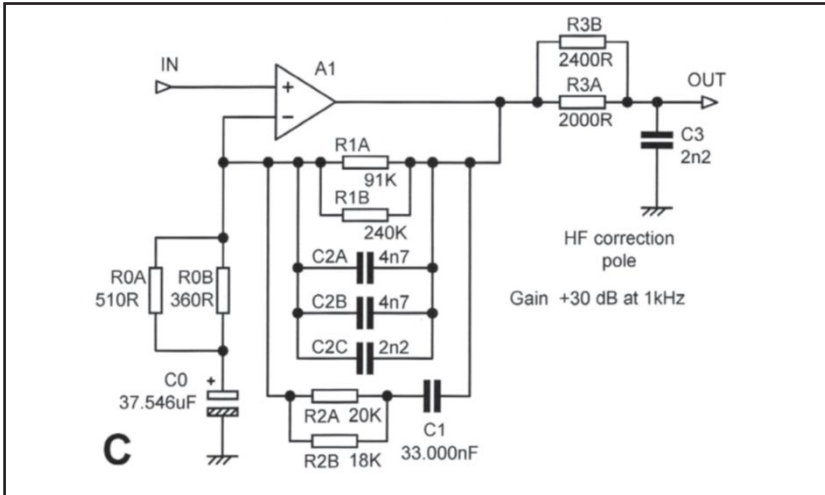


Figure 2: This is Figure 7.11 from Douglas Self's book, showing RIAA Network C optimized so a single, off-the-shelf capacitor can be used for C1. (Printed with permission © 2018 Taylor and Francis).

time constants. I agree completely. Instead, he places the switchable IEC rolloff as a separate high-pass filter at the output of the complete circuit, using a film cap. This is followed by the low-pass, high-frequency correction pole.

Self states that any phono preamp with single, off-the-shelf capacitors and resistors in the RIAA equalization network is likely to be inaccurate, which is usually true. RIAA networks just don't lend themselves to off-the-shelf values, especially capacitors. So, he spends considerable time dealing with the various issues involved in selecting RIAA capacitors, including capacitor cost, using single capacitors, and paralleling three to six caps to make the optimum values. Network C has never gotten much respect in commercial audio products, A and B being the most popular. Self shows that for a given gain and value of C0, Network C will require a smaller value for C1, the larger and more expensive of the two caps in the network.

He also shows how to optimize the networks using parallel combinations of three resistors and prefers parallel to series resistor networks because it makes PC board layouts "simpler and more compact." Self notes that "the relatively small number of E24 resistor values also means that it is difficult to pursue good nominal accuracy and effective tolerance reduction at the same time. This can be addressed by instead using three resistors in parallel." To simplify the equivalent resistor calculations, Self uses tables created by Gert Willmann. The tables are very lengthy, so he only shows a portion of the three-resistor version as an example. Willmann's complete tables are available on Self's website.

As he mentioned in the Preface, Self shows how to optimize Network C for overall parts cost that is lower than any of the other networks. He illustrates several permutations of Network C configured in a preamp with the 1 kHz gain set at 30 dB, using various combinations of parallel capacitors and resistors. He also discusses optimizing RIAA network C for different gains, and offers five pages of tables for multiple gains, and recommended parallel capacitor and resistor values. Self shows how component tolerances affect RIAA accuracy, and supplies graphs that are similar to a Monte Carlo analysis.

Lipshitz devoted a section of his paper to the issue of open-loop gain and its effect on RIAA accuracy. Self notes that there's no point in designing a super-accurate RIAA network if the active device (op-amp) doesn't have sufficient open-loop gain. Self points out that the 5534A has insufficient open-loop gain for a closed-loop gain of 40 dB at 1 kHz, if you are aiming for accurate RIAA response. His view is that, in order to accommodate a variety of cartridges and recording levels, 1 kHz gain should be set no higher than 30 dB. With 30 dB of gain, the errors due to limited open loop gain of the 5534A will be negligible. A gain of 30 dB may be fine for cartridge outputs of 5 mV or higher, but many of the better cartridges in the "high output" category have outputs in the range of 2 to 3.5 mV. I normally design my preamps with 40 dB of gain to accommodate these cartridges. Self does cover designs with higher gains, and shows how to implement variable gain into RIAA phono preamps with minimal effect on RIAA accuracy.

For those who don't have access to spreadsheets for the Lipshitz math, the data included in Self's RIAA chapter will be very helpful. In designing a spreadsheet, the question is: Where do you start? I never found the section in Lipshitz's paper titled "How to Design RIAA Networks" to be of much help. I'm not a mathematician, and needed something far more down-to-earth and practical. In the end, I had to find my own way with the Lipshitz math.

As I mention earlier, my approach to designing the spreadsheet was somewhat different from Self's. On page 109, Self notes that he begins with the desired gain at 1 kHz and the value of R0. The problem with this is that you often wind up with non-standard values for both capacitors in the network, which requires making two ideal values from parallel capacitors.

With my spreadsheets, you begin by entering a value for the larger of the two capacitors (C1). This will be a standard, off-the-shelf value, so you only need to use parallel capacitors to make C2, which is the smaller of the two caps. In three of his

optimizations for Network C, Self does come up with a single, off-the-shelf value for C1 (see **Figure 2**). Other variations have up to four parallel caps).

I hand-select my capacitors for 1% or better tolerance on a BK Precision 879B LCR meter, which has a 1% tolerance mode to assist in sorting. If I buy 10 of the 5% tolerance Wima polypropylene caps of the same value, I usually get two to four that are well within 1% tolerance, and a couple that are close to spot-on. The cost of hand selection isn't significantly higher than using three or four caps in parallel. I run simulations on every network I design, and I hand trim C2 in simulation to optimize the network for the open-loop gain of the op-amp in question. Some engineers might consider this cheating, but we have tools at our disposal that were unthinkable when Lipshitz wrote his paper back in the late 1970s. Why not use them to make our lives easier?

Regarding keeping the parts count low, I should mention that I used Network C in my Music Library Preamp more than 13 years ago (see **Figure 3**, which originally ran in *audioXpress*, March 2005). That phono preamp has a simulated RIAA accuracy of

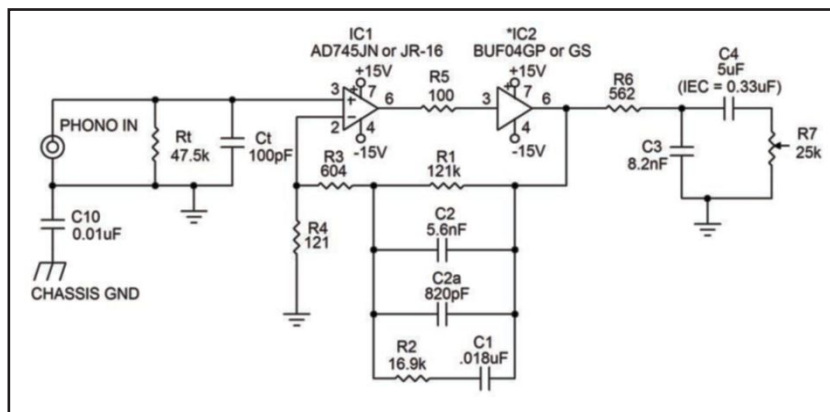


Figure 3: This is Gary Galo's Network C circuit from his Music Library Phono Preamp, published in the March 2005 issue of *audioXpress*. The preamp has 40 dB of gain and uses single, off-the-shelf values for all parts except C2. Measured RIAA accuracy is better than ± 0.05 dB, 20 Hz to 20 kHz, without the optional IEC roll-off. Noise is -86 dBu, unweighted. The circuit uses Lipshitz's R3 for amplifier stability, and a BUF04 high-current buffer inside the feedback loop.

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± 0.014 dB, and a measured accuracy of better than ± 0.050 , 20 Hz to 20 kHz, using a single cap for C1, two parallel caps for C2, and single resistors for R1 and R2. The capacitors are standard E12 values and the resistors are E96. For resistors from the E24 series, series and/or parallel combinations will be required. My goal was not minimum parts count—it just worked out that way. I also used only one capacitor for C1 in each of the preamps I described in Part 4 of my Adcom GFP-565 modification series (*audioXpress*, February 2004).

I can't imagine designing RIAA phono preamps without using SPICE simulation. On page 143, Self offers a SPICE circuit for generating an RIAA

emphasis curve—you'll need one to run RIAA preamp simulations. I also showed one in the previously mentioned articles, which is essentially the reciprocal of an RIAA de-emphasis model I got from Walt Jung more than 25 years ago. A lot of designers I know are using LTSpice. This free program was created by Linear Technology but has been offered and supported by Analog Devices since their acquisition of LT in 2016. I'm still using CircuitMaker 2000, which hasn't been sold in many years. But, it works just fine under Windows 7 and still does the things I need it to do.

Self is adamant that a magnetic cartridge should never be DC coupled to a phono preamp input. He's not aware of any published work on how cartridge distortion is affected by DC bias currents, "but I think it pretty clear that they will not improve things and may make them very much worse.

Large currents might partially demagnetize the magnet, be it moving or otherwise, ruining the cartridge." Self raises valid points here, but it probably depends on the input bias current of the op-amp and the impedance of the cartridge.

Walt Jung addressed this issue, as it relates to DC offset, in the section on RIAA phono preamps published in Analog Devices massive Op-Amp Applications. In reference to a DC-couple RIAA phono preamp he described, Jung noted: "For bipolar-input amplifiers, DC input bias current can be a potential problem when direct coupling to a cartridge, so in this circuit only a very low input bias current type is suggested. If a bipolar amplifier is used for U1, it should have an input current of < 100 nA for minimum DC offset problems (assuming a typical phono cartridge of ~ 1 k Ω ... FET amplifiers tend to have negligible bias currents but also tend to have higher voltage noise." My own view—and this is strictly my opinion—is that with FET-input amplifiers, DC-coupling is probably safe for the cartridge, but with bi-polar junction transistor inputs a high-quality capacitor is probably a wise precaution.

There are several popular op-amps that aren't unity-gain stable, including the OPA637, the AD745, and the LT1028 (and its nearly-identical cousin, the LT1115). The OPA637 and the AD745 are stable at gains of 5 or higher; the LT devices are stable at gains of 2 or higher. For amplifiers like these, Lipshitz adds a resistor in series with the RIAA feedback network, which is R3 in all of the examples in his paper.

The nature of a feedback-based RIAA preamp is that the gain decreases with increasing frequency, and at some point the gain will drop below the

Resources

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stable operating point of the op-amp. The value of R3 is calculated based on the value of the gain-determining resistor(s) in the feedback network, to ensure that the gain of the op-amp never drops below the lowest safe operating level. The 5532 is unity-gain stable and the 5534 can be made unity-gain stable by adding a small, external compensation capacitor, usually in the 10 pF to 20 pF range (5534 datasheets from various manufacturers don't offer much help in selecting this value). Because all of his designs are built around these two chips, he avoids discussing Lipshitz's R3.

A reasonably accurate RIAA network can be designed using altered time constants. These have been published in a number of sources, including Reg Williamson's 1990 article in *Audio Amateur*. For Network B, as an example, T3, T4 and T5 would be entered as 2937, 236.79 and 81.21 μ S, rather than the real values of 3180, 318 and 75 μ S. These altered numbers will work quite well, but once you throw in the resistor for amplifier stability, they're useless. A well-known designer once insisted, to me, that people used to design accurate RIAA networks before Stanley Lipshitz's paper was published. This is true, but in the old days amplification circuits, whether discrete or op-amp based, were normally unity-gain stable.

Later, some op-amps introduced for high-gain applications were designed to be stable at gains of 5 or higher. In an RIAA circuit, an amplifier stability resistor became essential. I've never seen anyone state this in print, but this is a major reason why the Lipshitz article is so important, because it provides the math for accurate design of RIAA networks with R3 in place. Even if the stability resistor is not needed, using the Lipshitz math and entering $R3 = 0$ will produce a more accurate network than one designed with the altered time constants.

It's better to get the calculations as close to ideal as possible, so that after rounding off for real-world parts—which you usually have to do even with series and parallel combinations—the end result will be excellent. Since Self used the Lipshitz math to design all of his RIAA networks, it's safe to assume that he entered "0" as the value of R3 in his spreadsheets. But, it's unfortunate that he left out any discussion of this issue, for those readers who might wish to investigate those specialized op-amps.

The DC servo integrator is another topic that Self doesn't discuss. I've had excellent results with DC servos in RIAA preamps, specifically in two designs I offered in my Adcom GFP-565 article. The two best sources on DC servo design are the section Jung wrote in Analog Devices' *1993 System Applications Guide*, and the chapter on DC servos in Bob Cordell's *Designing Audio Power Amplifiers*. A DC servo eliminates the need for an output coupling capacitor and C0 in the feedback network. Depending on your views about those capacitors, you may or may not consider this an advantage.

In the second part of this extensive book review, I will discuss some the remaining key chapters in Self's book and share my overall opinion of *Electronics for Vinyl*. 

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Modulation Distortion in Loudspeakers and Its Detectability (Part 2)

A Theoretical Approach to Evaluating the Detectability of Doppler Distortion

The September 2018 issue of *audioXpress* contained an introductory article on Doppler distortion (Dd) in loudspeakers. As reported in that article, the detectability, unpleasantness, and even the existence of Dd have been matters of discussion—and some contention—for about 75 years. In this article, we will discuss a theoretical rationale upon which a criterion for detectability of Dd can be based. In a later article, experimental results of an investigation of this theoretical construct will be evaluated, and those results will be compared with other experimental results published in the literature.

By
Richard Honeycutt
(United States)

In 1923, Vern Knudsen performed a number of rather extensive experiments, which determined among other things, the minimum pitch change which is perceptible by the human ear for an SPL of 40 dB. In 1931, E. G. Shower and R. Biddulph repeated his experiments, this time for a variety of SPLs. It was found that the minimum detectable ratio of:

$$\frac{\Delta f}{f}$$

where f is the original frequency and Δf is the change from that frequency, varies greatly with changes in level and frequency. The ear is most sensitive to small pitch changes at frequencies near 2000 Hz and at an SPL around 70 dB. The value they obtained for:

$$\frac{\Delta f}{f}$$

under these conditions is 0.0017. The ear can detect a pitch change if a 70 dB, 2000 Hz tone is shifted

by 0.17%, or 3.4 Hz to 2003.4 Hz or 1996.6 Hz. If we apply this finding to the classical formula for Doppler shift, we can obtain a criterion for theoretical evaluation of Doppler distortion.

First of all, we know that:

$$c = f\lambda \quad (1)$$

where c = the speed of sound in air,
 f = the frequency of the sound, and
 λ = the wavelength of the sound

Now the Doppler Effect is concerned with radiation from a moving source, so let us designate the formula as follows:

λ_s is the wavelength produced by a stationary source
 f_s is the frequency produced by the stationary source

Now we may write:

$$c = f_s \lambda_s \quad (2)$$



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or

$$\lambda_s = \frac{c}{f_s}$$

Similarly, we may define f_m and λ_m as the frequency and wavelength, respectively, produced by that same source when it is moving with a velocity v . Then:

$$\lambda_m = \frac{c}{f_m} \quad (3)$$

We are interested in the difference between the wavelengths produced by the stationary and the moving sources. Thus:

$$\Delta\lambda = |\lambda_s - \lambda_m| \quad (4)$$

or, from equations (2) and (3):

$$\Delta\lambda = \left| \frac{c}{f_s} - \frac{c}{f_m} \right| \quad (5a)$$

$$= \left| \frac{c(f_m - f_s)}{f_s f_m} \right| \quad (5b)$$

Now $f_s f_m$ is very nearly f_s^2 . Also, if a is the fraction (such as 0.0017, mentioned above) by which f must be multiplied to get Δf , then:

$$\Delta f = af \quad (6)$$

If we now note that the factor $(f_m - f_s)$ in equation (5b) is simply Δf , the change in frequency, we may write:

$$= \Delta\lambda = \frac{c\Delta f}{f_s^2} = \frac{caf}{f_s^2} \quad (7a)$$

$$= \frac{ca}{f_s} \quad (7b)$$

Next, we wish to derive a comparable formula for $\Delta\lambda$ from the classical Doppler formula:

$$\lambda_m = \frac{c - v}{f_s} \quad (8)$$

where, we recall, v = the velocity of the source.

Using equations (2) and (8), we may write:

$$\Delta\lambda = \lambda_s - \lambda_m = \frac{c}{f_s} - \frac{c - v}{f_s} = \frac{v}{f_s} \quad (9)$$

Equations (7) and (9) may be combined:

$$\Delta\lambda = \frac{v}{f_s} \cong \frac{ca}{f_s} \quad (10a)$$

or

$$v \cong ca \quad (10b)$$

If we now insert the values for c and a —we shall use $a = 0.0017$ for the minimum detectable Dd—we have:

$$v_{\min 70dB_{2000Hz}} \cong \left(\frac{1050\text{ft}}{\text{sec}} \right) (0.0017) = \frac{1.78\text{ft}}{\text{sec}} \quad (11)$$

This, then, is the hypothetical minimum cone velocity for detectable Dd, and we will assume (without any explicit justification) that since we are dealing with oscillatory motion, we may consider this to be the RMS velocity. (This seems to be a reasonable assumption, and, even if invalid, only involves an error of the order of 3 dB in our final results.)

The peak velocity is then:

$$v_p = \sqrt{2} V_{rms} \cong 2.52\text{ft} / \text{sec} \quad (12)$$

We may now calculate the peak cone excursion (maximum motion to either side of center equilibrium position) corresponding to this peak velocity at various frequencies, using the formula:

$$A = \frac{v_p}{\omega} = \frac{v_p}{2\pi f} \quad (13)$$

where A = amplitude of maximum cone excursion. The results are given in **Table 1**.

From standard acoustical texts such as Olson's *Acoustical Engineering*, we have the formula:

f (Hertz)	15 Hz	20 Hz	40 Hz	60 Hz	80 Hz	100 Hz
A (inches)	0.321"	0.241"	0.12"	0.08"	0.06"	0.048"

Table 1: This is the theoretical cone excursion for various frequencies for cone velocity required to produce minimum detectable Doppler distortion.

$$p(r) = \frac{vf\rho_0 s_c}{r} \quad (14)$$

where $p(r)$ = the magnitude of the RMS sound pressure produced at a distance r from a loudspeaker in an infinite baffle,
 ρ_0 = density of air $\approx 1.18 \text{ kg/m}^3$
 v = velocity of cone—here, the critical velocity for detectable Dd $\approx 1.78 \text{ ft/sec}$ or 0.543 m/sec
 r = the distance from the speaker to the point at which the SPL is measured—here we will use 3 m
 s_c = the effective surface area of the cone

(This formula is approximately valid only at frequencies at which the cone diameter is a small fraction of the wavelength; that is, only at low frequencies). Solving for a 12" speaker and a frequency of 20 Hz, we obtain:

$$p(r) = \frac{(0.543)(20)(1.18)(0.053)}{3} \approx 0.226 \text{ nt/m}^2$$

corresponding to 81.1 dB SPL

Similarly, we may compute the sound pressure levels for other frequencies and speaker diameters. The results are graphed in **Figure 1**.

Untested Values

Now, having determined these values, we must mention that they are impossible to test directly in a rigorous fashion, because what the listener hears is total modulation distortion, comprised of Amplitude-Modulation Distortion (AMD) and Dd, adding in an RMS fashion. James Moir's measured results mentioned in my previous article indicated that the contributions of AMD can likely be ignored, contrary to the expectations of many investigators of loudspeaker distortion.

We can, moreover, take advantage of the directionality of Dd. Since the production of Dd depends upon the motion of the loudspeaker cone toward and away from the listener, we may expect Dd to have a directional characteristic: it will be most pronounced directly in front of the loudspeaker and will be absent directly beside the speaker. Its value goes as a cosine function with respect to the angle from the speaker axis. Consequently, we may measure the critical level of the low or modulating frequency for minimum detectable Dd with the observer in front of the speaker, then repeat the measurement with the observer beside the speaker. A higher level of the

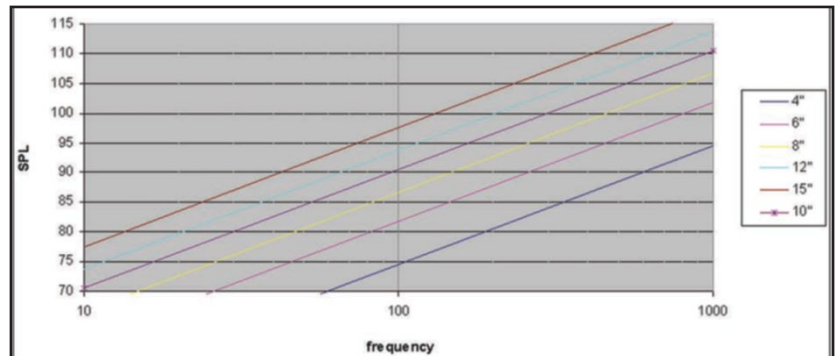


Figure 1: This is the theoretical SPL at which Dd becomes detectable for speakers of various sizes.

low frequency will be required in the latter case since, when he is beside the speaker, the observer will hear only AMD; whereas, when he is in front of the speaker, he will hear both AMD and Dd.

Suppose, for example, that for the speaker used, the amplitude of 60-Hz wave required to produce detectable AMD alone happened to be twice the amplitude required to produce detectable AMD and Dd. This means that the amplitude of AMD and Dd components required for detectability would

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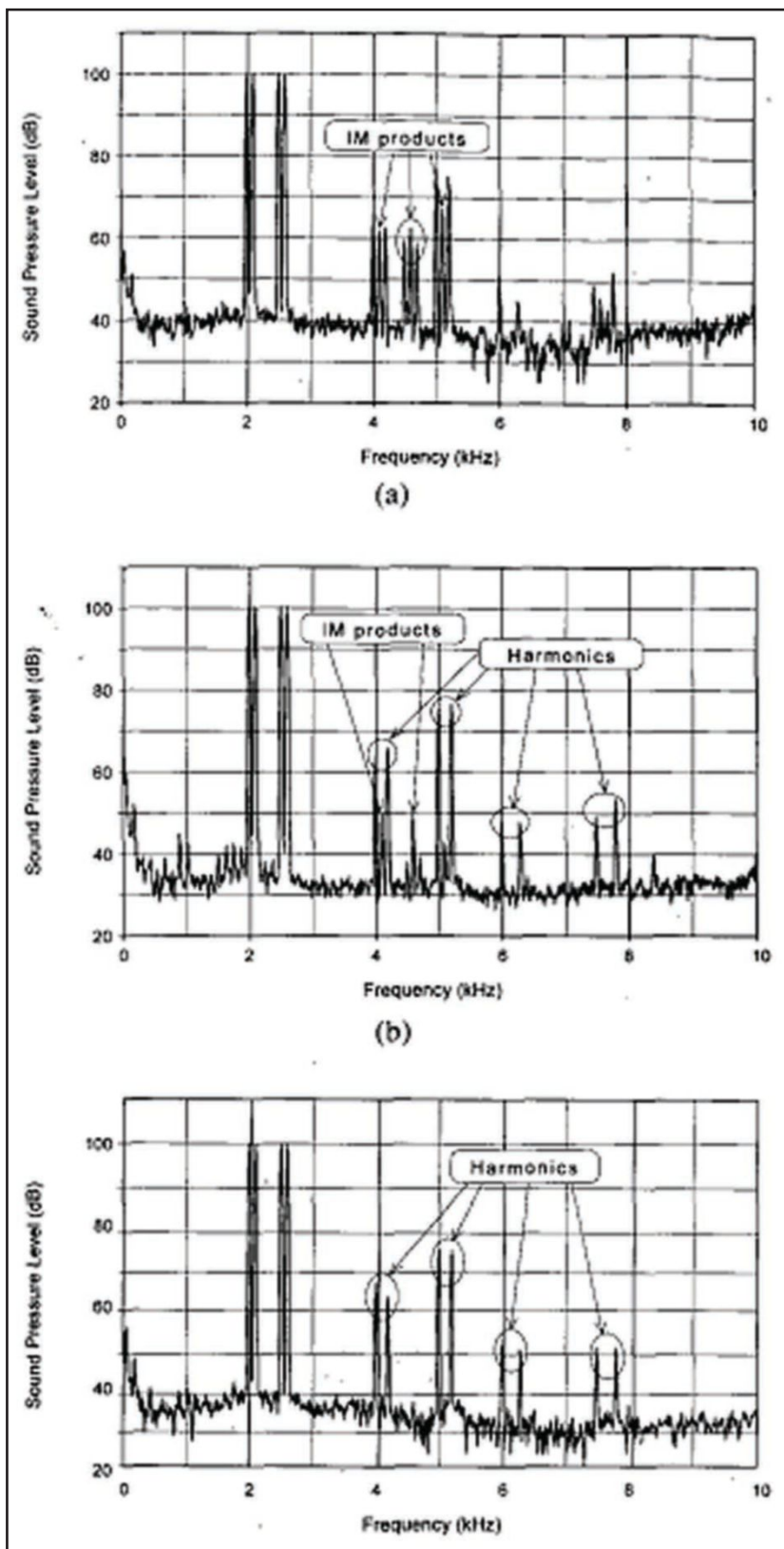


Figure 2: Here are spectra of "trombone" signals, simulated by four loudspeakers, with an SPL of 100 dB at 10 m. (a) separation between speakers = 1/2 m, (b) 1 m; (c) 3 m (Image from Czerwinski, et al, see Resources)

be twice the amplitude of the Dd components alone. Since the two add in an RMS (or vector quadrature) fashion, the amplitude of the Dd components must be 1.732 times the amplitude of the AMD components. And since both AMD and Dd are proportional to the amplitudes of the two signals, this ratio should hold for that loudspeaker at any SPL.

Now since 1.732 units of Dd add in quadrature to 1 unit of AMD to produce a barely audible two units of IMD, then 0.268 more units of Dd components would be required for audibility of Dd alone, in the absence of AMD (in a hypothetical loudspeaker, which was perfect from the AMD standpoint). We remember that Dd adds linearly to itself, not in an RMS fashion, since Dd components at a given frequency cannot be out of phase with themselves. Thus, 1.732 units of Dd + 0.268 units of Dd add to produce 2 units. This represents a necessary increase in the amplitude of the low-frequency wave of $2/1.732$ or 1.24 dB. We must, therefore, add 1.24 dB to our results in order to obtain the level of the low-frequency wave required to produce a detectable level of Doppler modulation of the higher wave, if we used a speaker which produced negligible AMD. (Note that this 1.24 dB correction factor is based upon a strictly hypothetical situation. It is mentioned only to illustrate the method of deriving a correction factor for an actual experiment.)

Factors Affecting Dd

Not only can we expect that larger speakers will produce lesser amounts of Dd at any combination

Resources

E. Czerwinski, A. Voishvillo, and A. Terekhov, "Air-Related Harmonic and Intermodulation Distortion in Large Sound Systems," *Journal of the Audio Engineering Society*, Volume 47, pp. 427-445, June 1999.

V. Knudson, "The Sensibility of the Ear to Small Differences in Intensity and Frequency," *Physics Review*, XXI, p. 84, 1923.

J. Moir: "Doppler Distortion in Loudspeakers," presented at the 46th Audio Engineering Society (AES) Convention, September 1973.

H. F. Olson, *Acoustical Engineering*, Van Nostrand and Co., pp. 558-591, Princeton, 1947.

E. G. Shower, and R. Biddulph, "Differential Pitch Sensitivity of the Ear," *Journal of the Acoustical Society of America*, Volume 3, p. 275, 1931.

of low (modulating) frequency and SPL, but we can also predict that multi-way speaker systems will also produce smaller amounts of Dd, since the speaker drivers producing mid and high frequencies are not simultaneously producing the low frequencies that require large cone excursions and velocities. However, personal communication with Paul Klipsch and other investigators indicates that even two-way speakers having concentric high-frequency horns can produce detectable Dd due to the motion of the air molecules excited by the woofer being superimposed on the motion created by the horn tweeter.

Although to my knowledge these results have not been published, Eugene Czerwinski, et al did present related results in which they detected modulation distortion in the sound produced by the trombone section of an orchestra, caused by cross-modulation of the different notes played by the players of the various parts. As illustrated in **Figure 2** and **Figure 3**, this distortion could be reduced by physically separating the trombones. The same behavior these researchers demonstrated with relation to distance between the trombones

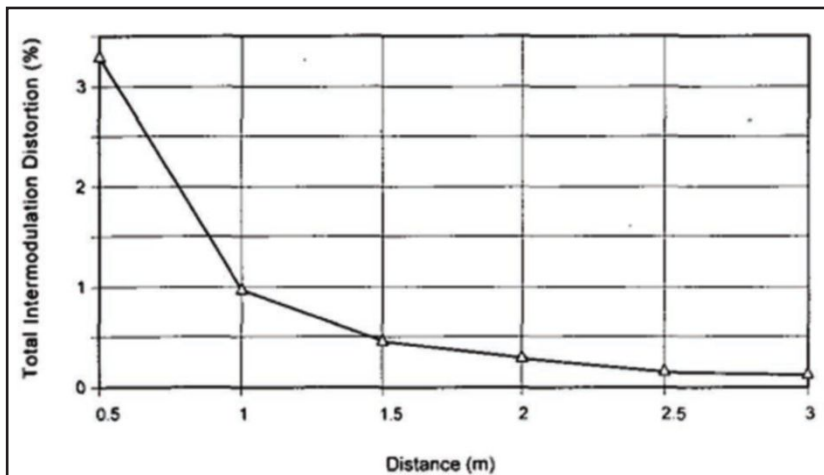


Figure 3: This graph shows the dependence of air-related IMD upon spacing between "trombone" speakers.

also applies to modulation distortion produced by interaction of the sound fields by the bass, mid, and high-frequency drivers in a multiway speaker system: larger spacing decreases air-related modulation distortion. Unfortunately, larger spacing can have malign effects upon acoustical imaging and interference within the crossover zone of frequencies. There's still no free lunch! 



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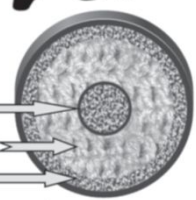
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Audio Software Doesn't Have to be Complex or Expensive to be Powerful

In this article, Pieter Stenekes, founder of Sonoris, describes how his company successfully created a range of solutions, which have been adopted by leading mastering studios and facilities around the world, and are still powering all types of deliverables in current digital audio formats.

By
Pieter Stenekes
(Founder, Sonoris)

Sonoris, a mastering audio software house, develops high-end audio products for recording and mastering professionals. Our multiplatform software ranges from a complete line of audio plug-ins to stand-alone Disk Description Protocol (DDP) solutions, all of which you'll find in mastering studios around the world.

Sonoris grew out of my love of technology and music. I have a degree in electronics and computer technology, but I'm also a semi-professional trumpet player. When it was time for college, I had to make the difficult choice between studying technology and going to conservatory for music education. Alas, I chose technology, but I was always looking for a way to combine the two.

I studied computers in a town in the north of the Netherlands, and at the end of college, I was lucky enough to get an internship with a company that specialized in high-end audio. I did a project about ultra-low-noise microphone amplification, which was great fun. From there, I spent 13 years at the Amsterdam-based electronics powerhouse Philips. The job was not music-related, but I got to work with very high-end equipment and gained priceless experience programming to high-level specifications and very strict tolerances.

During my years at Philips, I never lost my passion for music technology. In my spare time,

I not only developed audio software for myself, but I began developing plug-ins for a DAW called SAWStudio. I got good feedback from SAWStudio's loyal users and began to get requests for more and more plug-ins. Eventually, I quit my job at Philips and founded Sonoris.

That was in 2007, and the company is going strong with three main products: plug-ins, DDP software, and custom software.

Audio Tools with Unprecedented Transparency and Simplicity

From the beginning, my goal was to develop tools that are transparent, logical, and easy to use. That is, they should do the job without adding unwanted distortion or coloration of any kind, and they should come with an intuitive user interface.

At the time, simplification was a relatively new concept. There were a lot of companies developing plug-ins that were meant to look and act like popular analog outboard hardware, which, in my view, created unnecessary complexity and introduced room for error. My aim was never to emulate compressors, equalizers, and the like, with all their bells and whistles. Instead, I wanted to deliver transparent digital tools people could use alongside that hardware to enhance the music. That's just what Sonoris software has become known for:

digital tools that complement the audio hardware.

For example, many Sonoris clients use outboard hardware to shape the sound of a project and then use Sonoris software for last-minute refinement. That's because they know that if they use a Sonoris equalizer, for example, it will not ruin the color they established with the hardware. Instead, they can just do some final touch-ups to the music without doing any harm.

Filling the Void for Affordable DDP Imaging and Playback

As I broadened the plug-in range for SAWStudio, I began to get requests from mastering engineers who needed an affordable way to create error-free DDP files, which lets them transfer audio masters to CD manufacturers without burning a physical CD master. At the time, there were only a couple software platforms that allowed users to create DDP files. Those platforms were quite expensive, and therefore, only accessible to big-budget studios and factories.

My answer was the DDP Creator for Windows and Macs—a stand-alone, cross-platform DDP authoring tool I developed about eight years ago. It is much less expensive than the high-end platforms, yet it does all the same kinds of checks to ensure 100% integrity of the files. Compatible with virtually any DAW software on the market, DDP Creator lets users assemble a professional RedBook-compatible audio CD and then export it to a DDP 2.0 file set. Users love how straightforward and simple it is to use. You just drag audio files into the workspace; set the track order and adjust spacing; add metadata (e.g., title, performer, PQ codes, and ISRC); and when you're finished, click "export." That's it.

Most importantly, DDP Creator put DDP imaging within reach for smaller studios and independent mastering professionals. Now, after 10 years, online forums and magazines characterize DDP Creator as the industry standard for DDP authoring, not just for smaller studios, but for some of the industry's biggest players.

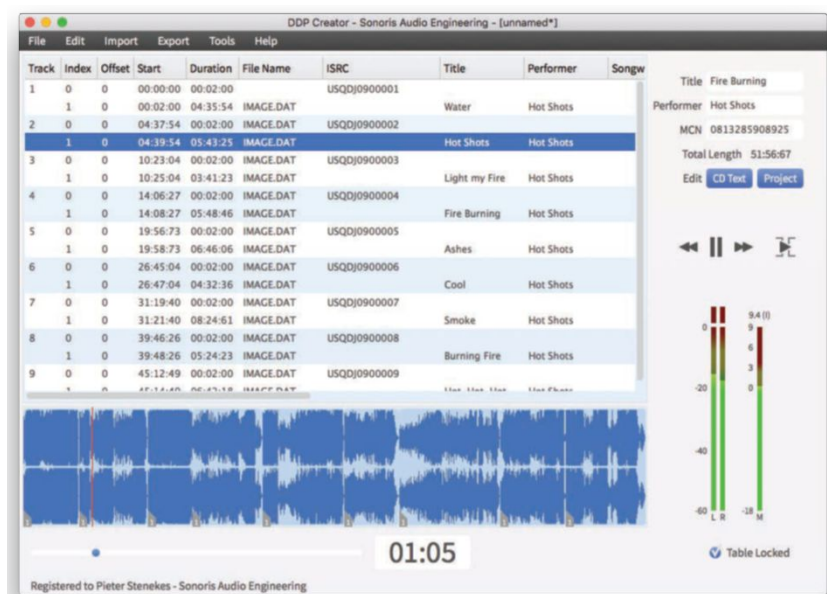
Based on the success of DDP Creator, we developed a spin-off called DDP Player, which lets studios and their clients (artists) do a final check of a mastering project before sending the DDP files to the factory. It used to be that mastering studios had to upload master files in .wav format to an online folder, along with a separate file to show spacing and the relative levels between songs. From there, artists would have to download, preview, and approve both files. Now artists can do a final check of the sound quality and spacing all at once through DDP Player without having to mess with downloads



Pieter Stenekes is Sonoris' founder and mastermind.

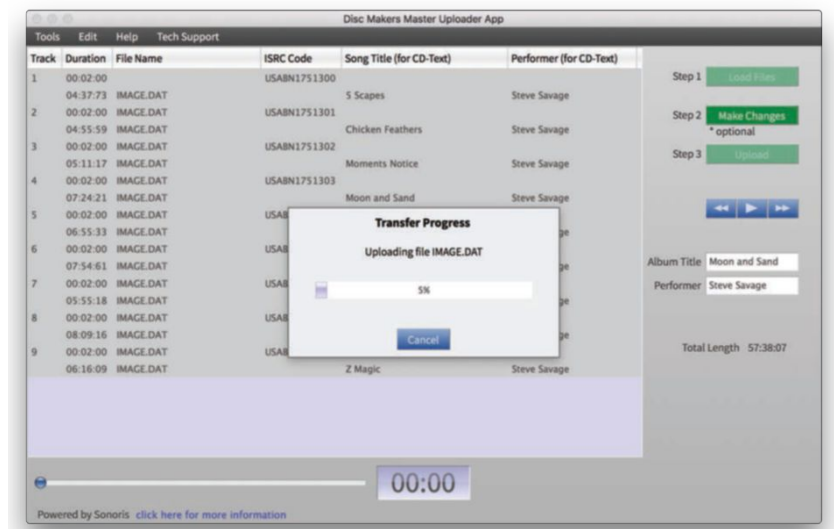
and separate files. Studios appreciate that Sonoris can customize DDP Player with the studio's branding and contact information.

Speaking of customization, Sonoris supplies OEM versions of another of its software products, the Master Uploader App, which was created for mastering professionals and CD manufacturers. Originally developed for Disc Makers, the world's largest independent music-mastering company, the Master Uploader App lets artists upload their initial audio files for mastering rather than having to ship the content on physical media. The Master Uploader App is built to spec and customized with the manufacturer's look and feel. Clients



DDP Creator is a standalone, cross-platform DDP authoring tool with high-end capabilities designed for audio engineers.

For more than 60 years, Disc Makers has seen the business evolve from pressing vinyl records to eight-track and cassette tape, and then on to current digital audio formats. The Disc Makers SoundLAB uses the Disk Description Protocol (DDP) software family from Sonoris.



With Sonoris DDP tools, mastering studios can generate an audio file and upload it online for client approval. Sonoris' tools speed delivery of final products with greater efficiency.

simply download and use the Master Uploader App to transmit a finished master directly to the manufacturer's FTP server and verify the accuracy of the upload—making concerns about faulty masters, CD handling, and shipping delays a thing of the past.

The Evolution of Mastering Influences the Evolution of Sonoris

Artists and companies such as Disc Makers have seen the business evolve through every medium—from vinyl to eight-track and cassette tape to current digital audio formats. Since I started Sonoris, the nature of music mastering and manufacturing has rapidly progressed toward digital, file-based transactions. In fact, right now I see the CD fading out in favor of streaming and compressed audio. CDs are still being pressed, but in much lower quantities. For example, today an artist might do a run of 500 CDs to have on hand during a performance, as opposed to pressing 10,000 CDs a decade ago to supply physical retail shops. I'm also seeing a resurgence of vinyl.

The DDP format is well-suited to all those media, so it's not going away anytime soon. Whether you want to send an album to iTunes or have it pressed into vinyl, you can do it with a DDP file set because all the high-quality audio is included along with all the metadata.

As the industry evolves, so too does Sonoris software. Sonoris still creates plug-ins for DAWs and does updates of DDP Creator to make sure it stays in step with the industry. For example, with vinyl making a comeback, I intend to add features to DDP Creator with vinyl in mind. However, I don't want to overload an existing product with features that don't serve its original purpose.

I listen carefully to mastering engineers, whose feedback and feature requests are invaluable in helping us develop products that best fit the mastering workflow. I have a clear vision for all Sonoris software to be transparent and intuitive, so my litmus test is always this: Is it a logical feature that adds value without bloating the software or affecting its ease of use? If not, then I won't add it to existing products. Instead, I keep it in mind for a new product.

The Master Uploader App is a perfect example. Disc Makers originally requested that the Master Uploader App functionality be added to DDP Creator, but then we agreed it would be better as a separate product. ISRC Editor is another.

Over the years, I've made a point of stressing accessibility and prompt support at Sonoris. Instead of filling out a trouble ticket and waiting for someone to deal with their issue, clients can always contact

About the Author

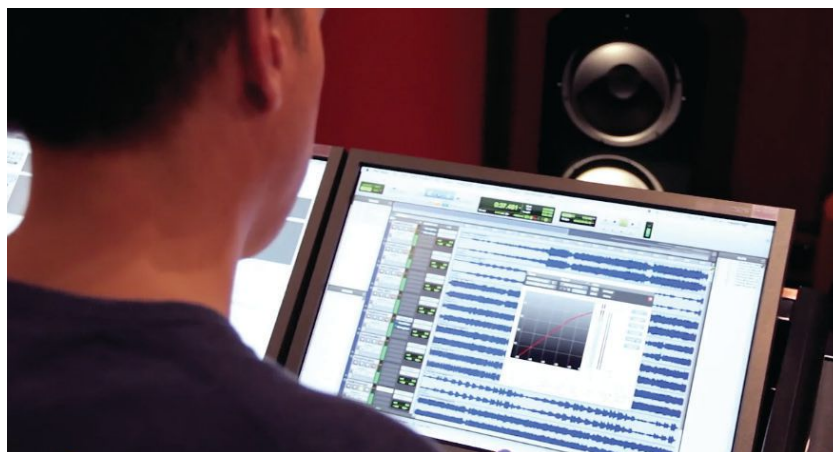
Pieter Stenekes founded Sonoris software, developers of high-end audio products for the recording and mastering professionals, in 2004. Prior to Sonoris, Pieter studied computer science, followed by a stint at a company that specialized in high-end audio. From there, he spent 13 years at Amsterdam-based electronics powerhouse Philips. He also developed audio software for himself, and began developing plug-ins for a DAW called SAWStudio. After getting good feedback from SAWStudio's loyal users, he began to get requests for more and more plug-ins. Eventually, he left Philips and founded Sonoris.

me directly to get one-on-one support—a benefit that is especially important for mastering engineers who work on tight deadlines.

Looking Toward the Future

I get plenty of requests for excellent new plugins or features, but I would rather develop a new product around a very good idea than to add it to an existing product when it doesn't make sense to do so. That's what is driving much of Sonoris' product development today. As the industry evolves, mastering engineers face new challenges all the time and will need new tools to help them overcome those challenges. Sonoris is anticipating and preparing for those challenges now.

With the rapid pace of change in the past decade alone, I can only imagine exactly how the industry will look in another 10 years, but one thing is for certain: Sonoris has proven the premise that audio software tools don't have to be elaborate, complicated, or expensive to be powerful and effective. We expect that premise to help us serve mastering studios of any size and budget well for years to come. Visit www.sonorissoftware.com for more information. 



For Disc Makers, Pieter Stenekes developed the Master Uploader App, which brings valuable time savings and efficiencies for clients submitting material. The app creates an electronic image file and establishes an FTP connection to Disc Makers' server, with built-in error checking to ensure that no errors are introduced during the transfer. Disc Makers is able to get to work on the content immediately with the assurance that it arrived in the exact form in which the client intended it.

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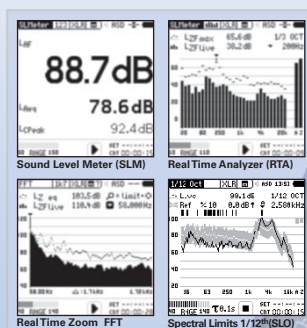
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PCB Piezotronics Phantom Power Microphones



Photo 1: The 426A14 microphone preamp and body is supplied with adapters to allow use of both 1/2" and 1/4" capsules. (Photo by Cynthia Wenslow)

Stuart Yaniger explores some of the phantom power microphones that PCB Piezotronics has developed for the loudspeaker and recording market.

By
Stuart Yaniger
(United States)

A couple years back, I had the pleasure of trying out some of the measurement microphones from PCB Piezotronics, located in Depew, NY. It was the first exposure I had to this company's products, which were focused on precision high-reliability industrial applications (e.g., environmental noise control, HVAC, appliance design, aerospace, and automotive). Common to this part of the market, the mics were self-polarized, operated from a constant current interface, and used coaxial cables for connection, much like other transducers in these applications—strain gauges, vibrometers, accelerometers, and torque sensors.

The basic capsule technologies for these industrial measurement condenser mics are the same as those used in audio measurement and recording. So, given that PCB Piezotronics' product manager Mark Valentino is an avid and accomplished musician, and that the market for instrumentation-grade measurement for loudspeakers and recording is growing, it is not surprising that PCB Piezotronics has decided to offer versions of its microphones suitable for those uses.

Yes, one can certainly use the constant-current self-polarized mics here, but the installed base of studio recording and loudspeaker measurement

equipment is designed for mics using phantom power (typically +48 V to +200 V) with balanced lines, so plug-and-play is not an option. Rather than expect studio and speaker measurement folks to change all of their equipment to accommodate the common interfacing for measurement mics, PCB Piezotronics introduced a new mic body/preamp, the 426A14, which runs off standard studio phantom power and has a standard balanced XLR connection.

The 426A14 is designed to work with existing PCB mic capsules (see **Photo 1**). These capsules are most self-polarized, that is, the backplate is coated with a fluoropolymer that is made into an electret by means of ion implantation and/or poling. The diaphragms are a stainless steel alloy, which tends to be more environmentally and thermally stable than the usual metallized polyester diaphragms typically found in measurement mics in this price range.

The Products

The new preamp is compatible with PCB Piezotronics' existing measurement mic capsules so that the advantages of the stability, calibration accuracy, and reliability coming with industrial measurement mics can be directly translated to audio-related acoustic measurement.

PCB Piezotronics sent me the new preamp bundled with three capsules:

- the 377C01—a 1/4" capsule, featuring extended frequency response and high SPL capability
- the 377B02—a 1/2" capsule, featuring a very low noise floor and high sensitivity
- the 377A06—a 1/2" capsule, featuring a higher SPL capability than the 377B02

The basic parameters of the capsules and the part numbers for complete mics (capsule plus preamp/body) are summarized in **Table 1**.

All of the PCB Piezotronics mic capsules are self-polarized (electret) with metal diaphragms (stainless steel alloy). Metal diaphragms offer the advantage of greater environmental stability and repeatability compared to plastic diaphragms and are the common choice for industrial measurement. By contrast, most recording mics use metallized plastic (usually polyesters such as Mylar with a vapor coating of aluminum), the properties of which change significantly with changes in temperature and humidity. This usually isn't a problem in studios and homes with climate control, but for field use, this can cause issues with frequency response and sensitivity. The PCB Piezotronics mic capsules are also all calibrated for free field (i.e., the polar patterns are basically omnidirectional).

This patent-pending adapter system is quick and convenient. In order to be able to use the same preamp with different capsule diameters, PCB Piezotronics has done some clever mechanical design work. The basic preamp comes with two different metal sleeves that screw onto the body. To use the 1/4" capsule, the cone-shaped adapter is screwed onto the preamp body, forming a streamlined and acoustically "clean" shape. To use the 1/2" capsules, a cylindrical adapter is screwed onto the preamp, forming a smooth cylinder. This adapter system is quick and convenient for capsule changes and is certainly a cost-saver for the user who needs the versatility of different capsule characteristics for different recording or measurement applications.

Measurements

My usual measurement setup comprises an Audio Precision APx-515 analyzer, an Audio Precision APx1701 transducer interface, a calibrated Brüel & Kjær Sound & Vibration (B&K) 4133 lab microphone (NIST-traceable), and a B&K 4231 SPL calibrator. Frequency response and polar measurements are taken using a quasi-anechoic method, a Farina chirp test signal that yields an impulse response, which can be gated to remove room reflections. This provides

Model	376A31	376A32	376A33
Capsule	377C01	377B02	377A06
Diameter	1/4"	1/2"	1/2"
Sensitivity (mV/Pa)	2	50	12.6
Noise floor (dB SPL A-weighted, ref. 20µPa)	40	15.5	22
Maximum SPL (dB SPL at <3% THD)	165	137	150
Note: All models use a 426A14 phantom power preamp/body			

Table 1: This is a summary of PCB Piezotronics' phantom power microphones.

valid data above 500 to 600 Hz or so, which is the region of interest (most precision mics are flat below that frequency). Frequency responses are normalized to the calibrated reference mic positioned in the same spot, which eliminates the frequency response variations of the acoustic source. The acoustic source was a Vanatoo Transparent Zero addressed via its Bluetooth input.

I first checked the sensitivity of the 1/2" mics using the 4231 SPL calibrator. The results are shown in **Table 2**. Because the SPL calibrator is a borrowed unit and I do not have a 1/4" adapter, the sensitivity of the 376A31 had to be estimated by substitution when measuring an acoustic source. The Calibration Sheet sensitivity from PCB Piezotronics was 2.18 mV/Pa, and my rough measurement confirmed this within the 1 dB or so error inherent in the substitution method.

I measure noise using the BLANK chamber, described in the PCB Piezotronics measurement mics review listed in Resources. Basically, the mic is wrapped in plastic and duct tape, then buried in a mixture of sand and kitty litter. This works remarkably well, pushing the noise floor below 4 to 5 dB SPL. My noise measurement numbers are over a 20 kHz bandwidth with A-weighting. It's important to note that when measuring random (e.g., thermal) noise, in order to get an accurate result, the Fast Fourier Transform (FFT) measurement should not be windowed. The noise measurements are shown in **Table 3**.

On-axis frequency responses from 1 to 20 kHz with each of the three capsule options are shown in **Figure 1**. The mics can be seen to be extremely flat

Model	376A32	376A33
Rated Sensitivity (mV/Pa)	50	12.56
Calibration Sheet Sensitivity (mV/Pa)	47.39	12.96
Measured Sensitivity (mV/Pa)	47.9	13.1
Note: All measurements taken with the 426A14 phantom power preamp/body		

Table 2: The sensitivity of PCB Piezotronics' phantom power microphones

Model	37A31	376A32	376A33
Rated Noise Floor (dB SPL A-weighted ref. 20 µPa)	40	15.5	22
Measured Noise (dB SPL A-weighted, ref. 20 µPa)	40	15	20.5
Note: All measurements are 20 kHz bandwidth, A-weighted			

Table 3: The noise measurements of PCB Piezotronics' phantom power microphones

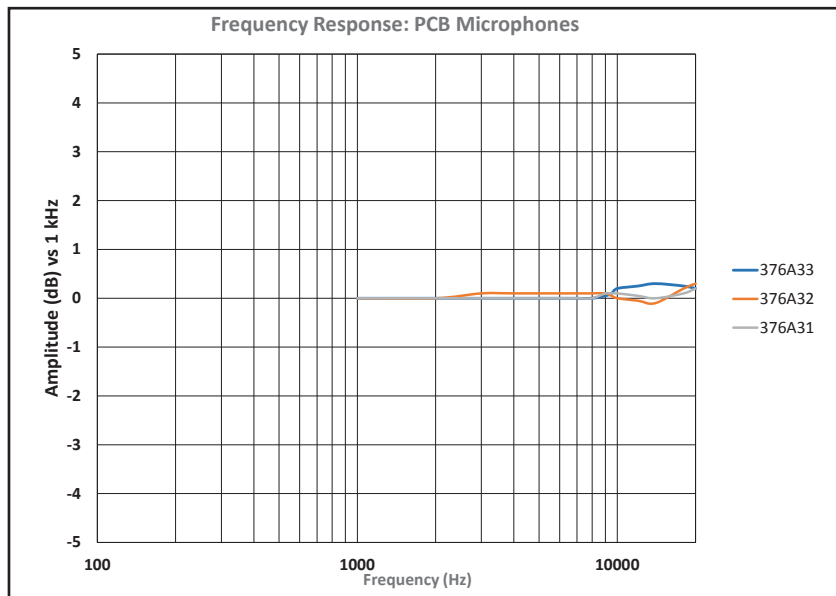


Figure 1: The on-axis frequency responses of the PCB microphones above 1 kHz are within 0.5 dB of flat, and correspond well to the supplied calibration curves.

up until the top octave, and even there, the deviations are low enough (less than 0.3 dB) to challenge my measurement accuracy. Calibration curves are included with each capsule and the supplied response curves agreed reasonably well with the measured data. The mics as supplied already have very flat frequency response, but inputting the calibration data into measurement software will assure that frequency response measurements are as accurate as possible. The nature of the mics' construction

(e.g., stainless steel diaphragms) will ensure that the response curves will be accurate across a variety of environmental (temperature and humidity) ranges.

Polar patterns were acquired by rotating the microphone body, with the diaphragm position with respect to the acoustic source kept at a fixed point. Responses at each frequency were normalized to the on-axis response, and each point is a power average of six measurements with the mic angle rotated slightly (less than 5°) between each measurement to average out high frequency diffraction effects.

Figure 2 shows the polar pattern for the 1/4" 376A31, which has an excellent omnidirectional characteristic up until the 10 kHz maximum frequency of my measurement. And even there, the polar pattern is reasonably smooth and does not have excessive roll-off. The polar pattern for the 1/2" 376A32 is shown in **Figure 3**. It can be seen to deviate from omnidirectional at a somewhat lower frequency than the 1/4" mic, but the response is still rather smooth and well-controlled. The 376A33's polar pattern matched that of the 376A32 within the limits of my measurement.

Distortion vs. SPL for the 376A33 is shown in **Figure 4** and compared to a similarly priced electret laboratory mic using a plastic diaphragm. These curves were obtained using a 6" loudspeaker driver measured nearfield (about 2 mm distance to the microphone from the cone), with an amplitude-swept sine wave. The measurement frequency was 1 kHz, though lowering or raising the frequency by a factor of two had no effect on measured distortion. My test limit was 130 dB SPL due to my desire for domestic harmony. Up to that point, the distortion curves for

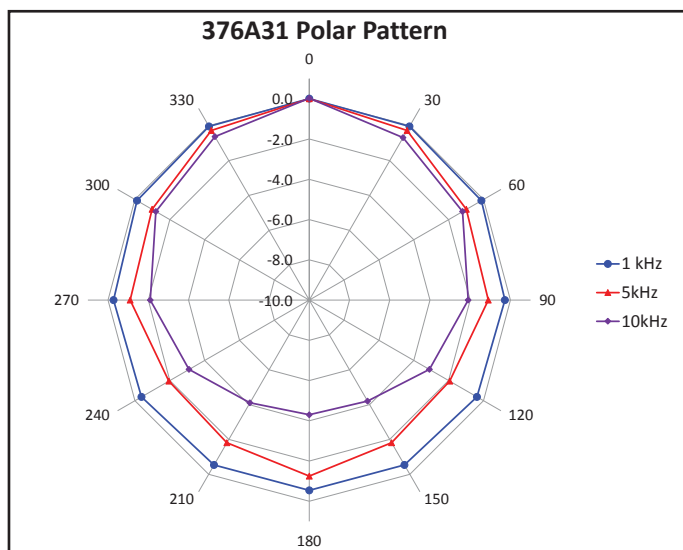


Figure 2: The polar pattern of the 376A31 mic (1/4" capsule) shows good omnidirectional behavior up to relatively high frequencies.

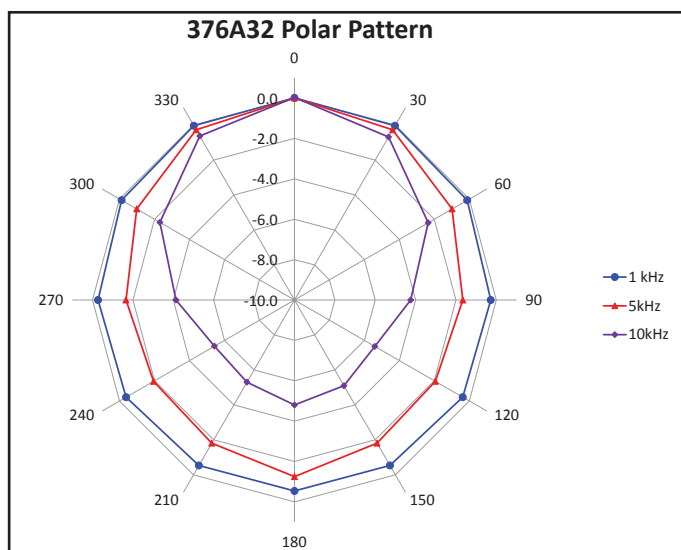


Figure 3: The polar pattern of the 376A32 mic (1/2" capsule) is smooth and well-controlled. The 1/2" capsule 376A33 has a polar pattern nearly identical to this.

the 376A31, the 376A32, and the 376A33 overlaid almost perfectly, so clearly the distortion limitation was the loudspeaker. Over the next few months, I will be experimenting with a few other ways to generate higher SPLs at lower distortion, but even extrapolating this curve, the 376A33 would be likely to comfortably exceed its specified 155 dB SPL limit for 3% distortion.

The PCB Mics in Action

I've been using these mics for loudspeaker measurement over the past year—some examples can be seen in the recent review of the Vanatoo Transparent Zero powered speakers (see Resources). The high SPL capability of the 376A31 1/4" mic makes it ideal for things like nearfield woofer measurement, and the flat frequency response, low noise, and low distortion of the 376A32 1/2" mic are great assets in measuring loudspeakers at 1 m distance. The excellent polar pattern comes in handy for in-room measurements, such as those seen in my recent review of the Sonarworks Reference 4 room correction system (see Resources). The high output (50 mV/Pa) makes mic preamp noise very non-critical, so paired with an inexpensive audio interface, this is a highly capable and reliable acoustic measurement system.

To get an idea of the PCB Piezotronics mic's performance in a recording setting, I recorded Michael Wenslow (of the WenslowShear Jazz Orchestra) warming up on his trumpet using the 376A33 mic. I chose this capsule/preamp combo, with a slightly higher noise floor but higher SPL capability because of the proximity to a rather loud brass instrument. The mic was placed about 1 m in front of the trumpet bell, and its output run through a Focusrite 2i2 interface at 24-bit resolution and a 96-kHz sample rate. The one-channel recording can be downloaded from my website (see Resources). The lack of compression and distortion as well as a relatively uncolored frequency response yield a very natural and dynamic sound. For quieter instruments or greater distances, the 378A32 would be a great choice for omnidirectional pickup. I should note that the cost of these mics is significantly lower than the closest alternatives.

I've also been doing some work recently on ultrasonic triangulation for distance and position measurement. The HC-SR04 module is very popular for this application, containing a transmitting and receiving unit on a compact circuit board. According to its datasheet, the HC-SR04 sends out a string of eight pulses at 40 kHz when the transmitter is triggered. The diagram shows the pulse train starting and stopping almost instantaneously. That struck me as a bit odd, since the transducers in ultrasonic

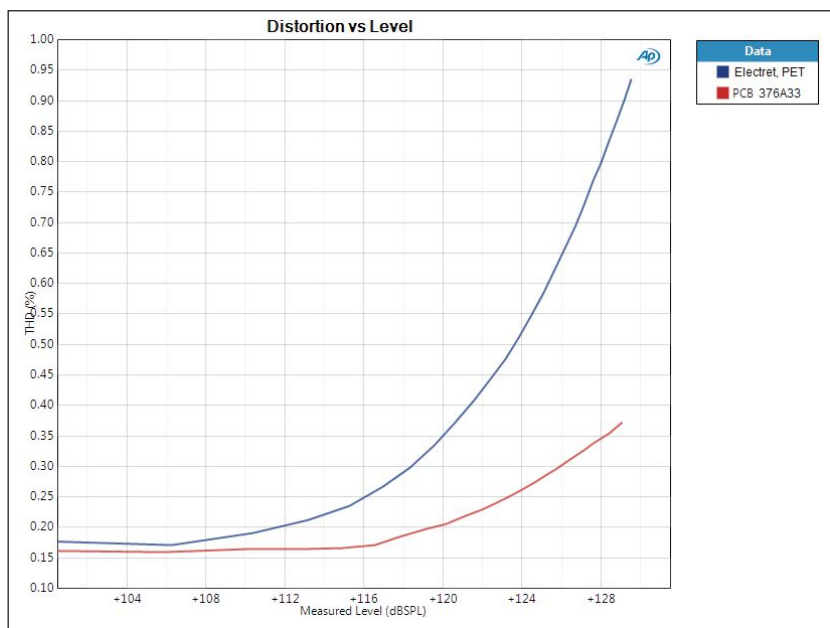


Figure 4: The distortion of the 376A33 vs. SPL is limited by the distortion of the signal source. Compared to a similarly priced phantom powered audio measurement mic, the distortion of the 376A33 is significantly lower.

transmitters are usually highly resonant at a single frequency, and thus they take some time for their output to build up in amplitude and then decay. Well, this calls for some measurement! The high frequency and high amplitudes suggested using the

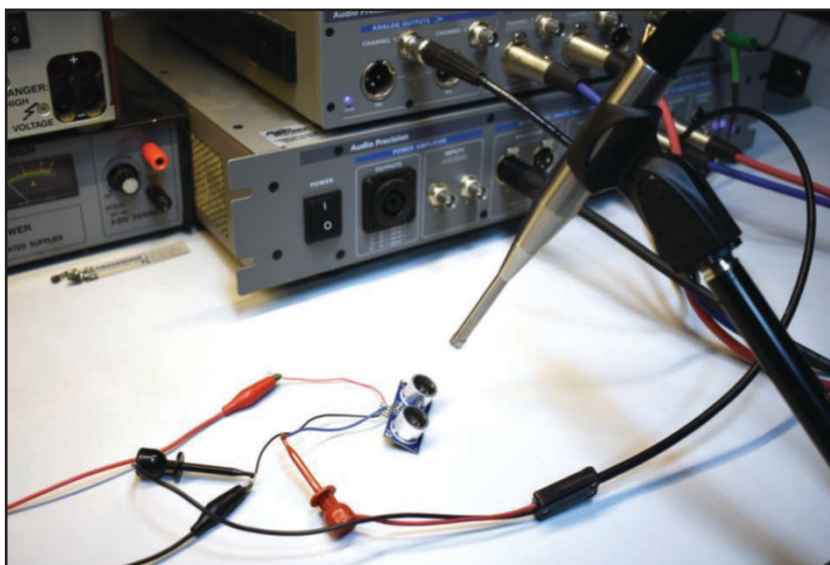


Photo 2: The 376A31 mic was used to measure the output of a popular ultrasonic transducer module. (Photo by Cynthia Wenslow)

About the Author

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

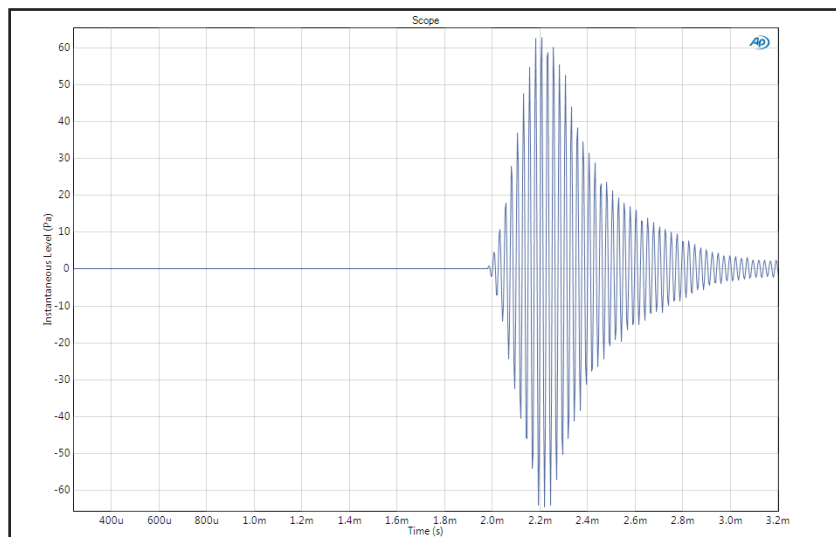


Figure 5: Measurement of the time domain acoustic signal generated by the ultrasonic module demonstrates the bandwidth and the high SPL capability of the 376A31 mic.

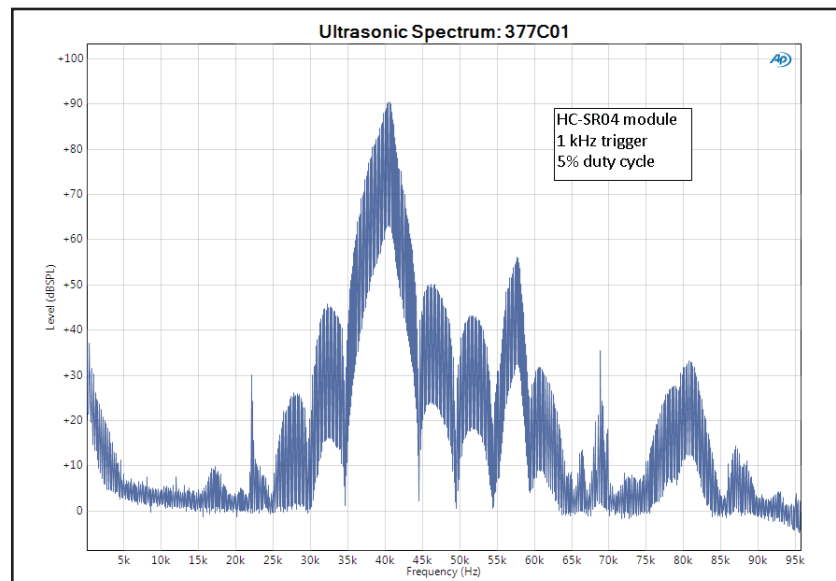


Figure 6: The frequency domain plot of the ultrasonic transducer output shows the expected nulls on either side of the fundamental from the repetition rate. Note the absence of mic distortion and the response to well above the second harmonic at 80 kHz.

Resources

"HC-SR04 Datasheet," www.mouser.com/ds/2/813/HCSR04-1022824.pdf.

S. Yaniger, "Sonarworks Reference 4 Studio Edition Loudspeaker and Headphone Correction System," *audioXpress*, July 2018.

—, "Vanatoo Transparent Zero Powered Speakers," *audioXpress*, January 2018.

—, "Innovative Measurement Microphones from PCB Piezotronics" *audioXpress*, December 2016.

Syclotron Audio, Sound clip of trumpet:
<https://syclotron.com/supplement-to-pcb-phantom-power-microphones-review-in-audioxpress>

1/4" 377C01 capsule because of its high-frequency response and ability to measure high SPL, which would be the case for a close-up measurement of the transmitter output.

The module was connected to a 5 V supply, and the trigger signal was provided by the Audio Precision analyzer. I positioned the mic about 3 cm away from the transmitter's ultrasonic transducer (see **Photo 2**). The time domain signal acquired is shown in **Figure 5**. This was closer to what I expected to see—the transducer takes all eight cycles to reach full amplitude, then rings for some time afterward. The ranging function then apparently looks at the echo from the initial wave train. Note that the SPL peaks at nearly 65 Pa, which corresponds to an SPL of about 130 dB. No wonder my dog went nuts when I did this experiment! In any case, also note that the peaks show no evidence of flat-topping or distortion, a tribute to the mic's excellent performance at high SPL and frequencies.

The frequency domain plot of this measurement is shown in **Figure 6**, with deep nulls at 5 kHz above and below the 40 kHz fundamental. This is expected because of the eight-cycle burst on the leading edge of the signal envelope. The mic adds no discernible hash or distortion and it can be seen that the response holds up well to beyond the 80 kHz second harmonic.

Wrap-Up

I had been very impressed with PCB Piezotronics' instrumentation mics, so it was not surprising that the new phantom power mics would be similarly impressive. For less than half the price of comparable instrumentation mics (e.g., the B&K that I have been using as a reference), the PCB Piezotronics mics allow laboratories and loudspeaker developers a reliable, repeatable, stable, and traceable way to gather acoustic measurement data. The materials and construction used are superior to nearly all other microphones in their price class, and the compatibility with the sort of interfaces used in studios means that recording engineers have a clean, low noise, and uncolored option available at a relative bargain price. The versatility of the preamp and adapters to accommodate different diameter capsules is a major cost and convenience advantage. I have no hesitation recommending these to studio engineers and for loudspeaker designers, and as long as I have these mics, they will become my reference tool for all other microphone evaluations.

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

A Visit to PCB Piezotronics

PCB Piezotronics, based in Depew, NY (a suburb of Buffalo), has been a leading player in the industrial measurement market for over half a century. Its specialties have been transducers for vibration, acceleration, torque, shock, and similar electromechanical measurements. Readers also might be familiar with the name Larson Davis, which in 1999 became a division of PCB Piezotronics, and has its sales and marketing operations based in Farmington Hills, MI, and Cincinnati, OH. As a manufacturer of acoustic test and vibration measurement instrumentation, Larson Davis is recognized for its sound level meters, sound sources, calibration systems and even headphone test fixtures, among many other instrumentation products, all manufactured at its Provo, UT, location.

PCB Piezotronics' industrial measurement microphones, using the ICP technology discussed in a previous *audioXpress* review, have become a standard for measurement of noise in the development and certification of machinery, appliances, autos and trucks, and aerospace products. Until recently, the company has been less well-known in the audio world, though that is certainly starting to change.

Part of becoming a significant player in those markets, as well as expanding its profile in its traditional markets, has been an ongoing commitment to customer training. Not only does this make for better-educated customers who will get more accurate and useful results from their measurements, it's also a chance to show off its factory. PCB Piezotronics' training seminars cover pressure, vibration, ballistics, and of course, acoustics. I attended an acoustics seminar, which was held near the Depew facility and was taught by Dr. Andrew Barnard, who is a professor of mechanical engineering at Michigan Technological University.

The seminar was a two-day event, with Day 1 being dedicated to the fundamentals of acoustics. If you've been through a vibrations and waves course, much of this will look familiar, but it's quite fun and educational to actually watch the measurement of polar patterns of dipolar sources as a function of wavelength. The mysterious decibel (dB) was explained, with an eye toward acoustic reference levels and what decibel of sound pressure level (dB SPL) really means. Of course, the use of decibels in electrical measurement was also presented, especially in the concept of understanding microphone signal to noise and the origins of different contributions to the mic noise floor. The basic concepts of acoustical metrology such as random vs. diffuse field measurement, were also covered in depth.

An interesting sidelight detailed microphone design and construction. This includes the concept of constant charge operation of condenser mics and the way constant-current interfacing can be used to connect the mic with appropriate electronics. And that brought a thought to my mind, which had been already bubbling away in the background—the audience for this seminar was primarily people doing different sorts of industrial noise measurement, be it for product design (especially appliance and automotive) or safety compliance (e.g., noise levels of the sort that interests OSHA). The industrial sector really does the same sort of thing that people do when developing loudspeakers, measuring performance and sound reproduction spaces, and recording musicians. But despite the two similar missions, the industrial people and audio people speak completely different languages. You'll rarely hear a recording engineer mention the term "diffuse field" and likewise, a noise abatement engineer for a truck company will not be familiar

with terms such as "Blumlein stereo bars." In fact, the reason for the reference to 0.775V in dBu measurements was unfamiliar to everyone in the class. (It's the voltage that causes 1 mW power dissipation in 600 Ω , long an impedance standard in sound studios.)

The metaphorical language barrier even extends to the electronics, with industrial measurement mics being almost entirely single ended constant current, and audio measurement and recording mics being balanced phantom power voltage sources. And never the twain shall meet! With my experience being almost entirely in loudspeaker measurement and music recording, this exposure to a different world and a different language was a particularly educational experience, raising my awareness of a lot of issues that we, on our side of the barrier, often ignore.

Day 2 started with a discussion on calibration and traceability. I should mention that PCB Piezotronics is now owned by MTS Systems Corp., a well-known and reputable provider of a variety of mechanical and thermal testing equipment. (Disclaimer: The company I work for in my day job has divisions that are competitors of MTS, though not in the product areas where I work.) The Modal Shop is another MTS company that specializes in calibration and measurement, and its representatives presented this part of the course. Besides the usual calibration and traceability information, they went into detail about uncertainty and error budgets, something too often ignored in reporting of measurements. Calibration methods discussed included shock tubes, a method not commonly seen in audio measurement but yields some very useful information on high-frequency response limits that's difficult to obtain using more common methods.

A tour of the factory and R&D areas was perhaps the most interesting part of the visit. PCB Piezotronics' manufacturing facility is certified under ISO9001 and AS9100, and the measurement and calibration is certified under ISO/IEC 17025. In my professional career, I have set up and run several manufacturing sites so I can say without hesitation that this is one of the best laid out and maintained facilities I've ever seen. **Photo 1** shows a bird's-eye view of the CNC machining area for producing the mechanical components of the mics, which is notably clean, well-lit, and efficient. **Photo 2** shows the laser marking



Photo 1: The CNC machining at PCB Piezotronics' Depew facility produces the mechanical components for the microphones. (Photo courtesy of PCB Piezotronics)

process for the mic bodies—Goldfinger would be envious! Final assembly is done in a carefully controlled clean room (see **Photo 3**), reminiscent of the processes used in semiconductor fabrications.

The calibration processes include electrostatic actuation, a method which eliminates the effect of room acoustics but can generally be done only with metal diaphragms (see **Photo 4**), and a state-of-the-art anechoic chamber (with heroic noise isolation measures) for true free-field measurement (see **Photo 5**). Our tour of PCB Piezotronics' anechoic chamber and acoustical calibration lab was led by Dr. Chad Walber, an R&D engineer at the Depew facility. If you've never been inside an anechoic chamber, the experience is somewhat unnerving, with an almost oppressive feeling from the total lack of reverberant sound. The noise level is low enough that the dominant sound you hear is the blood pumping through your ears.

The most fun part for me was the demonstration of high-level acoustic pulse measurement, which took place in a basement chamber. The source was an AR15 rifle firing a round into a water trough. Ear protection required! I kept the shell of the test round as a souvenir, and fortunately didn't run into any issues with TSA. As with my

measurements of a loud ultrasonic transducer, the mic used in this demonstration had a 1/4" capsule for optimum high SPL performance.

All in all, this was a very concentrated program, but the knowledge imparted and the discussions with so many experts made it more than worthwhile. I came away with a much deeper knowledge of microphone manufacturing and measurement, as well as an increased appreciation of the discipline of industrial acoustic measurement. If you have an interest in acoustic metrology or are involved in music or television sound production, this course will be extremely enlightening.

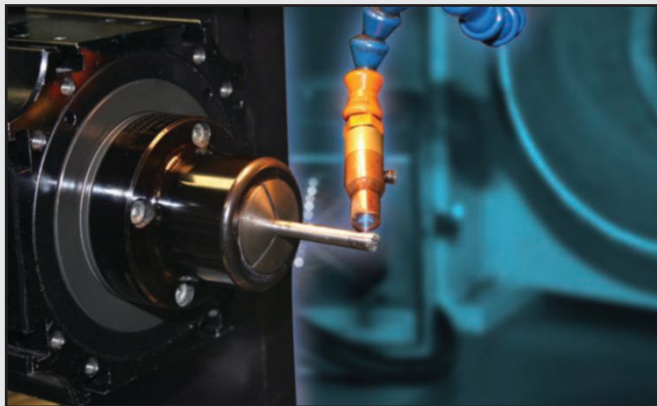


Photo 2: Microphone bodies are marked, welded, and etched using a precision laser process. This provides more robust marking than conventional printing. (Photo courtesy of PCB Piezotronics)



Photo 3: Final assembly of the mic capsules is done in a clean room environment. (Photo courtesy of PCB Piezotronics)

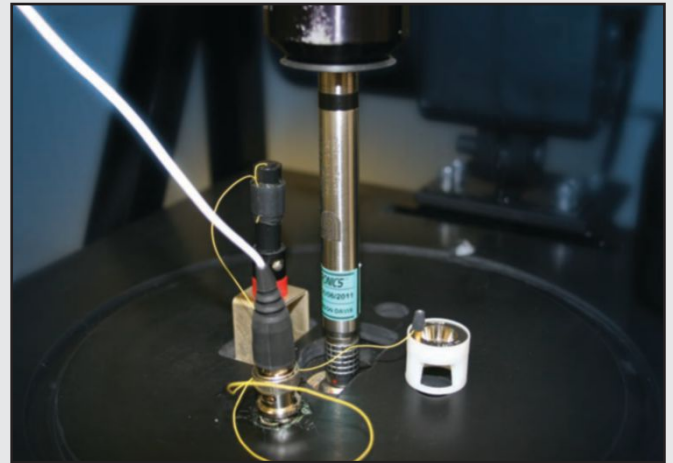


Photo 4: Electrostatic actuator calibration is performed on each mic capsule. (Photo courtesy of PCB Piezotronics)

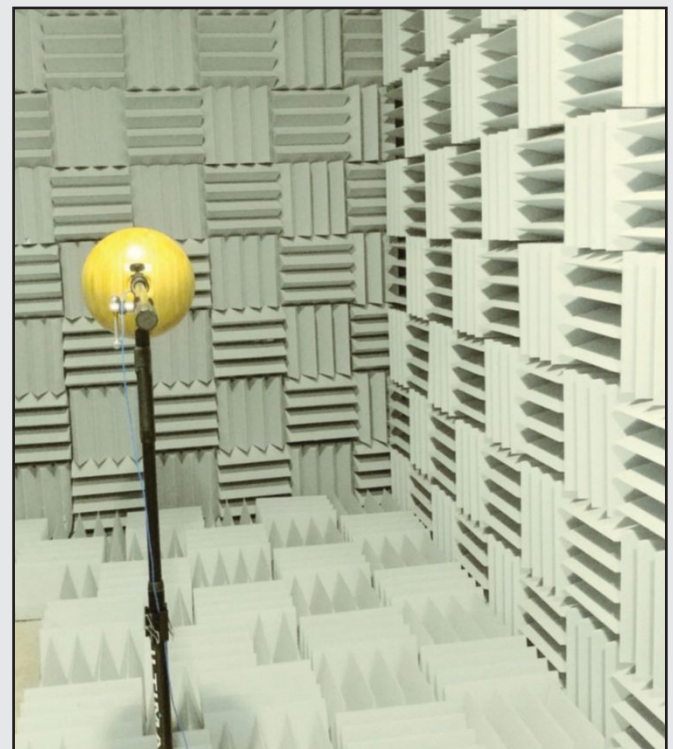
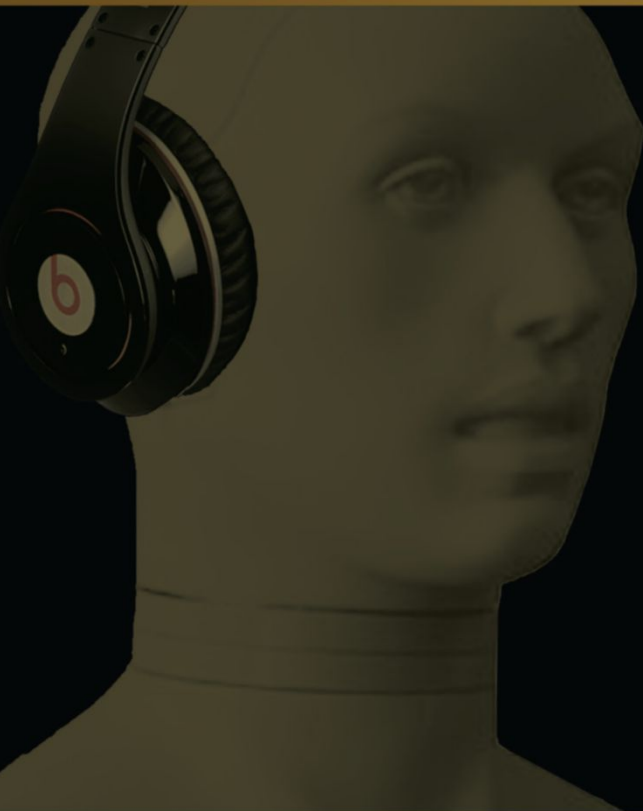


Photo 5: Calibration and development are validated in a custom-built anechoic chamber. (Photo courtesy of PCB Piezotronics)

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Advanced Test Methods for Improving ANC Headphone Performance

Active noise-cancelling headphones are designed to electronically remove the sound coming from your surroundings. They work by using internal microphones that listen to what's happening in the world around you, inverting the noise, and sending it into the loudspeaker. The idea is both the output and the input will cancel out, leaving you with near-silence—or to whatever you choose to listen. This article attempts to gain better data to draw stronger conclusions about the ANC performance.

By
Hans W. Gierlich
(HEAD acoustics)



Photo 1: Testing an ANC headphone with the artificial head measurement system HMS II.3 and labCORE—the modular multi-channel front end platform

Active noise cancellation (ANC) in headphones has been available for several years, and has seen significant improvements in the amount of peak attenuation achievable. Today, customers use ANC in many different scenarios:

- Users want to shut themselves off from their environment. They'd like to avoid unwanted background noises (e.g., when traveling by plane, bus, or sitting in a noisy room). That is why they turn the ANC to maximum. In this scenario, it is not possible to play back any audio.
- In the second scenario, users still desire maximum noise cancellation but want to play back audio files such as music, news, or podcasts at the same time—or simply have speech in a phone call.
- The third use case scenario is about hearables and the idea of acoustic augmented reality.

Users will want to hear the environment while they listen to audio or speech. The ANC is activated, but not at the maximum level. Thus, a pleasant and seamless interaction with the environment, and with other people around and on the other end of a phone call is still possible without having to increase the vocal effort and not use too much listening effort to understand the other person.

This article focuses mainly on the first and second scenario. The question we investigated is: How can we gain better data to draw stronger conclusions about the ANC performance? While testing ANC systems or headphones today users often have stationary noise playback and look at overall attenuation.

In some cases, ANC systems and headphones are also evaluated based on their ability in the bass (active), mid (cross-over) and treble (passive)

regions, as well as how much they “leak” sound (see **Photo 1**). Those fundamental tests are good, but we identified four avenues that would allow us to improve our data.

First, instead of playing white or pink noise out of one or two speakers in a simplified noise field, we suggested playing back realistic background noise sources. There are standards existing in the sector of telecommunication testing such as European Telecommunications Standards Institute (ETSI) TS 103 224 that address this. This standard uses an eight-microphone array and an eight-loudspeaker configuration for recording sounds and playing them back in a laboratory.

The microphones are used to capture the spatial characteristics of the original sound field, so, during the equalization process and the playback process users are able to accurately playback the sound in terms of magnitude vs. frequency and also in terms of phase vs. frequency. This configuration is perfectly suitable for stressing beamforming devices or multi-microphone position devices, as well as providing a good impression on how the product behaves in the real world.

Second, when looking at the residual noise left behind by an ANC system, we suggested applying hearing models to gain a better understanding of the noise impact. The human brain is not a Fast Fourier Transform (FFT) analyzer, so it makes sense to evaluate ANC performance from the perspective of a hearing model (e.g., Zwicker Loudness) that is meant to approximate human hearing.

Third, we proposed looking at the temporal aspects of the ANC performance for two main reasons—one, our brains more closely resemble a time waveform analyzer and time domain events can, therefore, impact our sense of hearing, and two, the ANC systems and headphones may respond in different ways when they are subjected to transients in the background noise environment found in the real world. A simple level vs. time analysis can reveal much in this area. In contrast to the frequently used pink noise—which is very predictive, deterministic, and free of transients—the noise scenarios found in ETSI TS 103 224 (e.g., pub noise) include lots of activity in the background and lots of transients.

Finally, when activating an ANC system, the last thing we wanted was to compromise the integrity of the reproduced audio signal. That is why we chose to use sophisticated hearing models (e.g., the Relative Approach) to evaluate what the audio quality would be like when the ANC was switched on and when the ANC was switched off under various background noise scenarios.



Photo 2: 3PASS setup in a laboratory

Test Setup with HEAD acoustics Equipment

When testing ANC headphones with HEAD acoustics equipment, the setup consists of an ACQUA system—the advanced communication quality analysis software that is connected to a labCORE front end. This modular multi-channel is equipped with coreBT—the Bluetooth module for labCORE. The front end is connected to a pair of Bluetooth headphones and sends the stimulating source signal from ACQUA to the headphones and into the ears. Then, the signal from the ears is sent back to the labCORE and further to the ACQUA analysis.

In the meantime, the eight loudspeakers are driven by the background noise simulation system 3PASS (see **Photo 2**). A predefined background

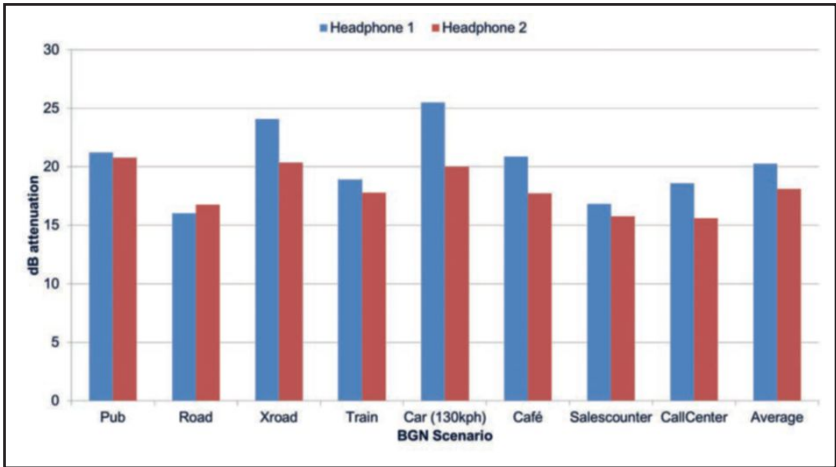


Figure 1: Overall active attenuation in decibels (dB)

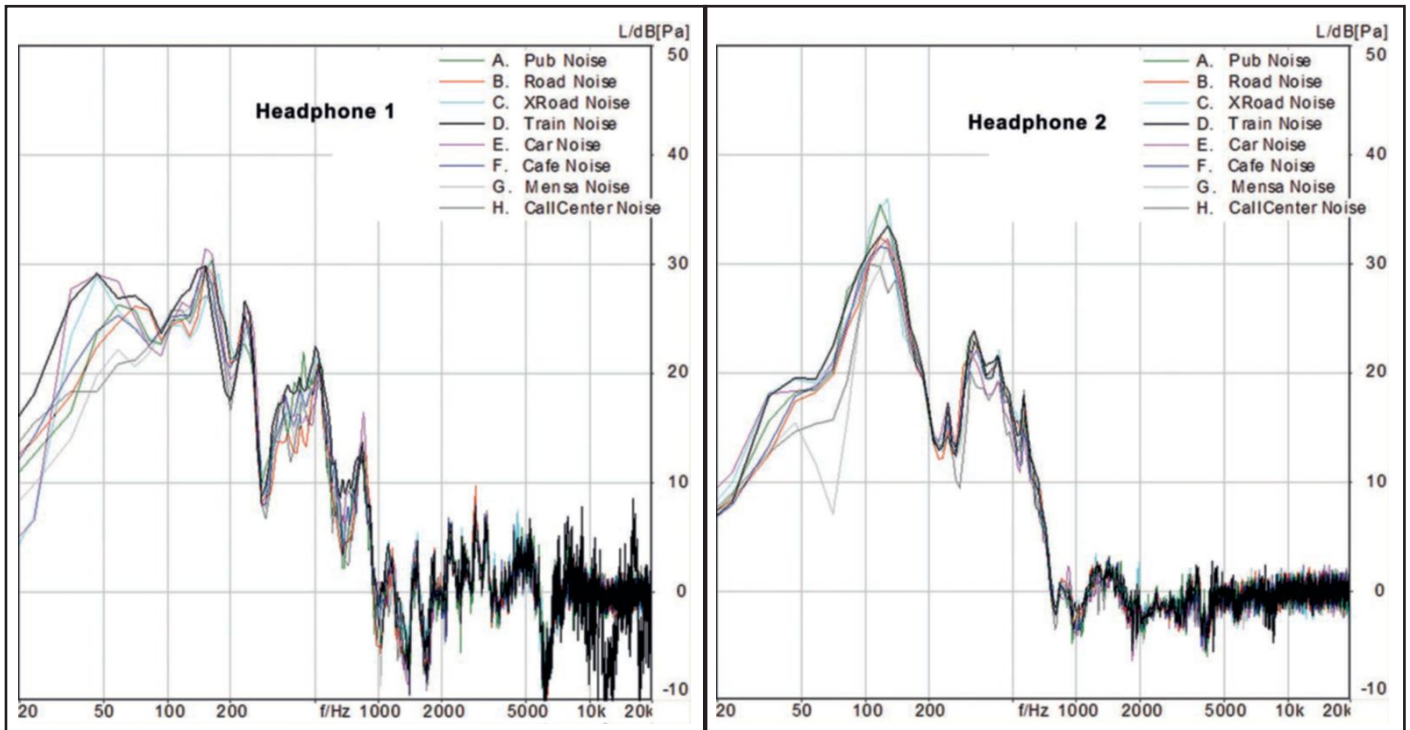


Figure 2: Overall active attenuation for Headphone 1 (a) and Headphone 2 (b)

noise was played back while measuring an ANC headphone. The ACQUA and the 3PASS computer are connected and synchronized such that any audio playback is going to occur simultaneously with the background noise playback. That made it possible

to look at every time slice across headphone tests and compare them individually.

Comparison of Two ANC Headphones

Based on these advanced test methods and with the above described setup, we compared two commercially available ANC capable headphones. First, we measured the active attenuation achieved by these two headphones across different background noise scenarios and evaluated the performance according to traditional FFT-subtraction methods. We exposed the headphones to different noise scenarios (e.g., pub noise, road noise, call center noise, and more). The tests revealed that Headphone 1 performs generally better throughout the different scenarios (especially driving noise, which happens to resemble pink noise) than Headphone 2. The average active attenuation of Headphone 1 is about 20.2 dB, whereas Headphone 2 achieves an average score of 18.1 dB (see **Figure 1**).

Looking at the data from a frequency-based perspective (20 Hz to 20 kHz; ANC on minus ANC off), it once again showed that Headphone 1 slightly outperforms Headphone 2 (see **Figure 2**). While the peak attenuation at 150 Hz isn't quite as high as Headphone 2, Headphone 1 is clearly doing a better job attenuating the low frequencies and as well as offering a much higher upper frequency (1 kHz vs. ~750 Hz) than Headphone 2.

About the Author

Hans W. Gierlich started his professional career in 1983 at the Institute for Communication Engineering at RWTH, Aachen. In February 1988, he received a Ph.D. in electrical engineering. In 1989, Hans joined HEAD acoustics GmbH in Aachen as vice president. Since 1999, he has been head of the HEAD acoustics Telecom Division and in 2014, he was appointed to the board of directors. Hans is mainly involved in acoustics, speech signal processing and its perceptual effects, QOS and QOE topics, measurement technology, and speech transmission quality. He is active in various standardization bodies such as ITU-T, 3GPP, GCF, IEEE, TIA, CTIA, DKE, and VDA and chairman of the ETSI Technical Committee for "Speech and Multimedia Transmission Quality."

About HEAD acoustics—Telecom Division

HEAD acoustics was founded in 1986 and has been involved in noise and vibration, electroacoustic and voice quality testing since its inception. HEAD acoustics is based in Herzogenrath, Germany, with affiliates in China, France, Great Britain, Japan, South Korea, and the US, as well as a worldwide network of representatives. The Telecom Division of HEAD acoustics manufactures telecom test equipment and provides consulting services in the field of speech and audio quality. Moreover, HEAD acoustics closely co-operates with DECT Forum, ETSI, ITU-T, 3GPP, TIA, CTIA, GSM, and other standardization bodies with regard to the development of quality standards for voice transmission and speech communication. In many partnership projects, HEAD acoustics has proven its competence and capabilities in conducting tests and optimizing communication products with respect to speech and audio quality under end-to-end as well as mouth-to-ear scenarios.

Testing Headphones?

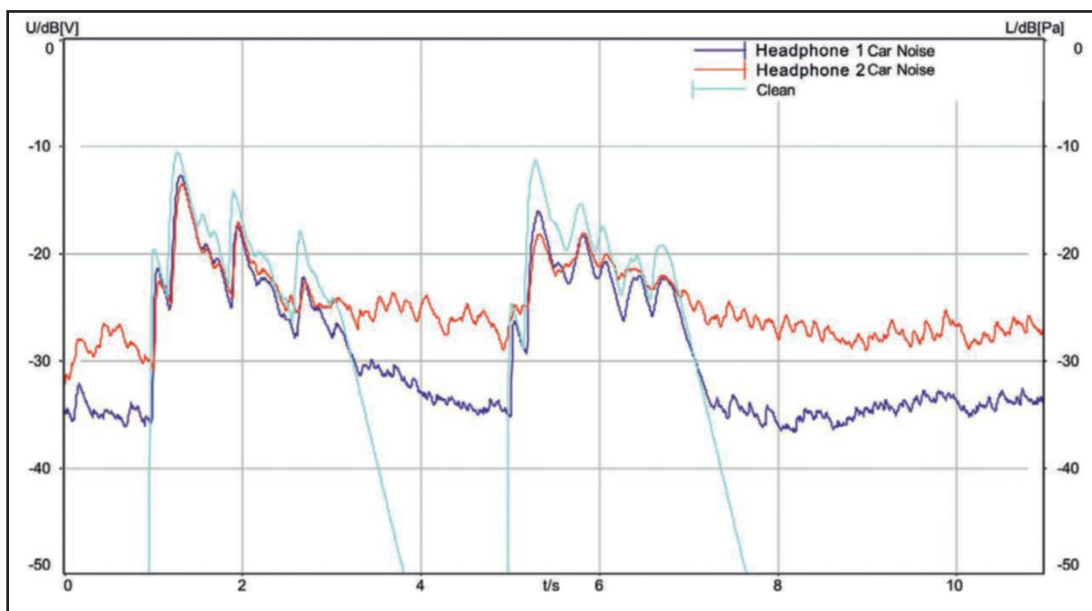


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Figure 3: Level vs. Time analysis



The conclusion we drew is that the traditional objective FFT-based analysis methods would have crowned Headphone 1 the unequivocal king of ANC performance.

When we examined the temporal aspects in

the level vs time analysis (see **Figure 3**), where we compared the headphones in noise to the level vs. time of the speech stimulus, it indicated that Headphone 1 provided better speech tracking. Furthermore, the noise level was generally lower and less prone to fluctuations.

However, when we looked at the data through the lenses of a frequency-based hearing model (Zwicker Loudness), which takes into account things such as frequency masking to get a better sense of how a human brain perceived a noise event, we got a vastly different picture. In ANC off mode (only passive attenuation), the overall noise through Headphone 1 is perceived as louder (see **Figure 4**). When switching ANC on, of course the loudness rating of both headphones 1 and 2 decreases, yet the gap between them widens. Headphone 2 reduced the overall noise perceived not significantly more compared to the noise perceived at Headphone 1. When we looked at Zwicker Loudness levels for the two headphones in individual noise scenarios, Headphone 2 outperformed Headphone 1, except in the driving noise scenario (most akin to pin noise).

Another advanced test method is the Speech Intelligibility Index (SII). Ironically, the SII metric does not measure speech, rather it measures a noise environment and then estimates how intrusive that noise environment is against idealized speech. It turned out again that across the different noise scenarios, Headphone 2 achieved better active and passive SII scores and that ANC improved SII more in Headphone 2 than in Headphone 1 (see **Figure 5**).

So while the traditional methods painted one picture, we were starting to see a different image of the two headphone ANC systems appear.

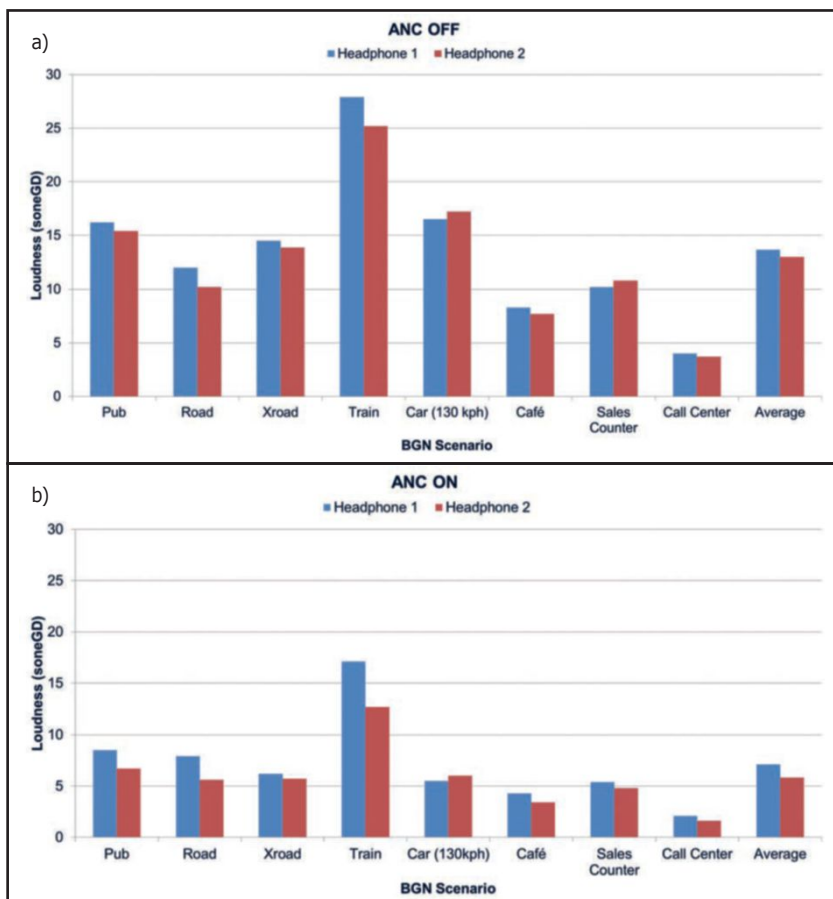


Figure 4: Zwicker Loudness in ANC Off (a) and ANC On mode (b)

As an additional way to evaluate the ANC performance, we decided to use a new Mean Opinion Scores (MOS) metric from ETSI TS 103 281. It uses the sophisticated Relative Approach hearing model, which is both temporally and spectrally sensitive to approximate human hearing. It was originally designed for evaluating speech quality in the presence of noise for a communication device that is transmitting sound, so the application of the metric in the receiving direction is a little bit experimental. However, the metric actually gave us three scores in one. We got an S-MOS score (indicator of speech quality), an N-MOS score (indicator of noise suppression quality), and a G-MOS score (global score). We used the N-MOS portion of ETSI TS 103 281 metric to evaluate the noise suppression quality of both headphones—which revealed that both perform well, however, Headphone 2 once again performed a little bit better. When applying ANC, both headphones improved equally well and the overall difference between them was minor (N-MOS Headphone 1 = 3.2; N-MOS Headphone 2 = 3.3)—but Headphone 2 performed technically better.

Finally, we wanted to evaluate the speech quality performance in the presence of background noise

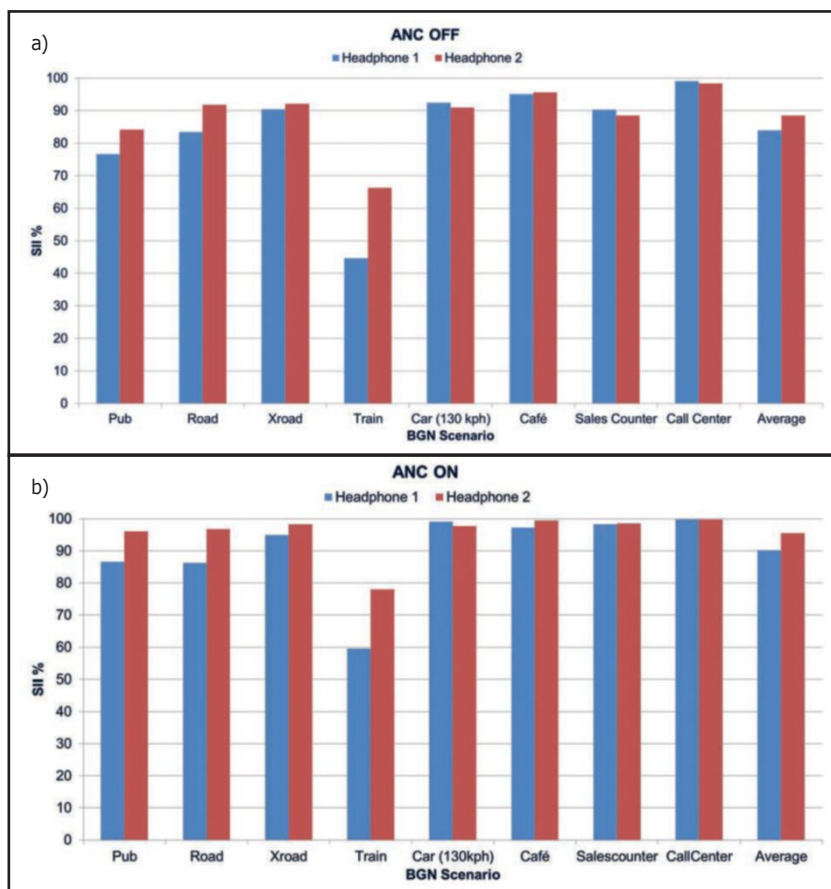


Figure 5: Speech Intelligibility Index (SII) with ANC Off (a) and ANC On (b)

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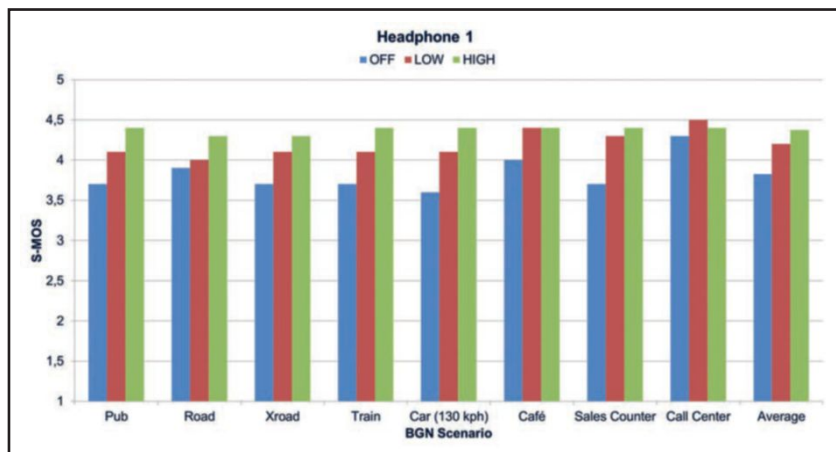


Figure 6: ANC effects on speech quality using Headphone 1

by using the S-MOS portion of ETSI TS 103 281. Headphone 1 can be used in three different ANC settings: ANC on, ANC low, and ANC high. In general, the measurements showed that we tend to get improved speech quality when increasing the level of active noise cancellation (see **Figure 6**).

Since the ETSI TS 103 281 metric uses speech files as its stimulus, we did have to make some assumptions about audio quality: If the speech quality is good, the audio quality is good, too. Looking at Headphone 1's performance across noise environments, this holds true except in the case of call center noise. This is relatively low-level noise (people talking quietly in the background, keyboard typing, etc.). When applying too much ANC, it lead to a slight drop in speech quality. Perhaps this was due to the fact that call center was the quietest predefined noise scenario and the ANC system

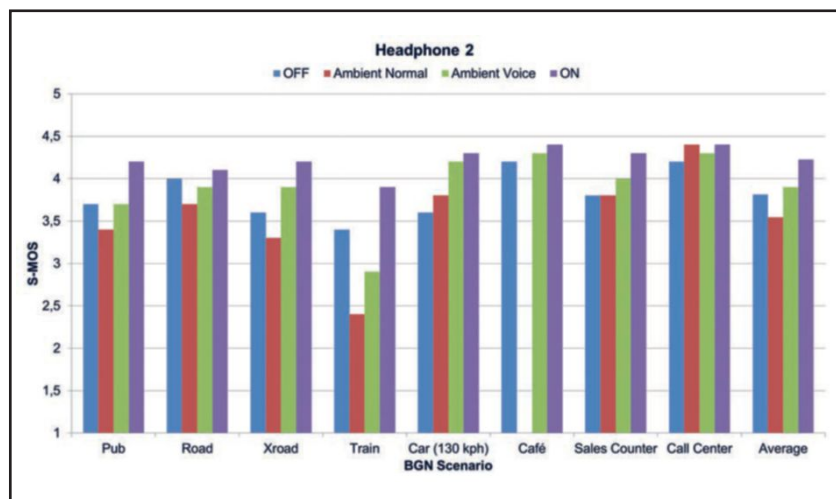


Figure 7: ANC effects on speech quality using Headphone 2

in Headphone 1 was so aggressive that it sought something to cancel—and, unfortunately, affected the speech quality—or that we are affected by the self-noise of the ANC system.

Headphone 2 can be used in four different ANC settings: ANC off, ambient normal, ambient voice, and ANC on (see **Figure 7**). Ambient normal and ambient voice settings are meant to apply gain at the higher frequencies in order to cancel out the passive attenuation and essentially give the impression that you aren't wearing any headphone. In addition, the ambient voice setting has some slight attenuation at the lower frequencies in an attempt to improve speech quality. In some sense, those two settings are an attempt at creating a "hearables" type product as we defined in scenario three. As with Headphone 1, increasing the amount of ANC in Headphone 2 led to the same positive trend of improving speech quality.

Comparing the S-MOS performance of Headphone 1 and Headphone 2 showed that Headphone 1 seemed to perform better both with ANC on and ANC off (ANC off/ Average S-MOS: Headphone 1 = 3.8; Headphone 2 = 3.8 // ANC on/Average S-MOS: Headphone 1 = 4.3; Headphone 2 = 4.2).

Conclusion

Comparing the performance of both headphones, using the different test methods across different background noise scenarios, revealed a more balanced view of the two.

- Headphone 1 scored well using traditional FFT-based analysis techniques, level vs. time and in speech quality reproduction with and without ANC.
- Headphone 2, however, tended to score better when using hearing models and psychoacoustic metrics for evaluating the residual noise.

The primary message is that judging the ANC performance solely on FFT-based analysis can be misleading.

For those who want to explore further, we have presented four areas where testing can be expanded in order to gain better insight into ANC headphone performance. First, more realistic noise source with spatial characteristics should be applied for better use case validation. Second, use loudness models and psychoacoustic analyses to judge impact of residual noise. Third, look at temporal aspects and behavior regarding performance of ANC systems and regarding the human brain's performance. Finally, assess the influence on audio quality in different ANC settings.

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You Can DIY!—Speaker Feature

The Incredible Egg (Part 2)

Assembling the Eggs

By
Thomas Perazella
(United States)

In the first installment of this article, Tom Perazella discussed the philosophy of the project and preparing the egg enclosures. In this article, we proceed with the assembly phase.

Tom Perazella's completed Egg Speakers are shown as they premiered at Capital Audio Fest 2017.
(Photo courtesy of Paul Elliot)

Normally assembling a wood enclosure is relatively straightforward, especially where biscuits are used for the alignment function. Not so with the eggs. There were several factors that had to be considered. The first was the order required to assemble the rings. Because when assembled, the rings would create an internal surface with many stepped edges that could affect the sound,

the edges had to be coarsely smoothed during the assembly process. In order to access those edges, I assembled the upper and lower halves separately. With each half, the two outermost pieces were first assembled and adjacent rings added one at a time until the halfway point of each section was reached.

This assembly was not a continuous process and the shape also created some clamping problems. Starting with one end, I applied glue to the mating surfaces and registration holes. Then, I added four register pins and clamped the rings together using bar clamps.

After the first few rings, it was not possible for the relatively short arms of the bar clamps to accommodate the egg's changing shape. In order to continue using the bar clamps, I made "X" shaped fixtures from two pieces of 2 × 4 lumber (see **Photo 1**). I designed the lengths of the arms of the fixtures to allow the clamps to reach the edge of the largest ring. The smallest section was rested on the center of the fixture and one end of the clamp fastened to the ring being added and the other end



Photo 1: The egg clamping fixture



Photo 2: The clamping of the rings

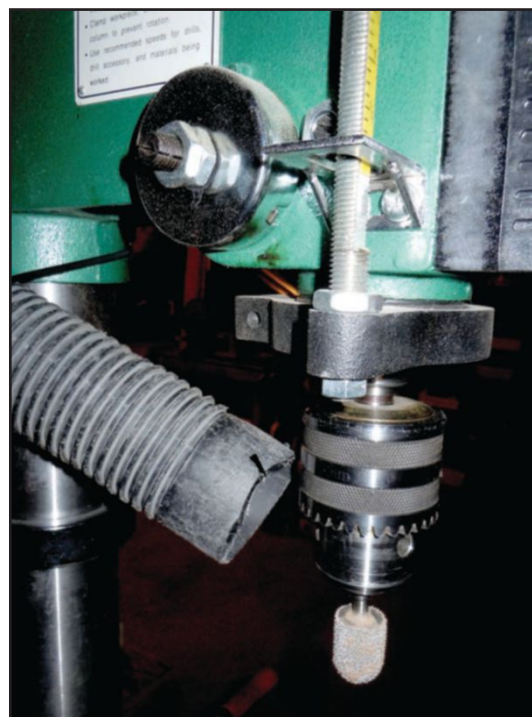


Photo 3: I used the burr in the drill press to remove the steps.

to a spot along the fixture that corresponded to the larger ring dimension. That way the clamp could be kept vertical with respect to the center line of the emerging egg (see **Photo 2**). The fixture coupled the clamping force to the smaller part of the egg.

After every few rings were added, I had to stop the gluing process so that the stepped edges could be removed. To accomplish that, I used a carbide round top burr. I placed the burr in the chuck of a drill press and positioned the hose from a shop vacuum adjacent to the burr to remove the dust created in the process (see **Photo 3**).

MDF is great for building speaker enclosures, but the dust is not good for your health and vacuum removal is always warranted when using that material (see **Photo 4**). The glued sections were handheld against the burr to remove the edges. It is only necessary to break the sharp edges and a perfectly symmetrical inside surface is not only unnecessary but undesirable. Random fluctuations in the inside surface will further reduce the tendency of standing waves to form (see **Photo 5**).

Once the rings for the top and bottom halves of the egg were glued, I glued the pieces of MDF



Photo 4: The inside of the egg before removing the steps



Photo 5: The inside of the egg after removing the steps



Photo 6: The diffuser in the top of the egg



Photo 7: The diffuser in the bottom of the egg

that were irregularly shaped into the top and bottom halves. They provided surfaces that would break up reflections and also provided an extra thickness for the bottom where a hole would later be drilled to hold a locating pin (see **Photo 6** and **Photo 7**). Finally, the two halves of each egg were aligned and glued together. The resulting raw eggs had a stepped outside surface. I got several comments that they looked like beehives (see **Photo 8**).

Finishing the Egg Woodworking

To finish the woodworking part of the egg enclosures, three steps remained. The outside stepped edges had to be smoothed, the holes for the drivers had to be drilled, and the outside had to be sanded before finishing. For the first step of smoothing the shape of the eggs, I made a shape template from the original drawings to check the dimensions of the outside of the eggs. To create the

template, I glued a copy of the original egg-shape drawing to a piece of Luan board and then cut it to shape with a jig saw. Then, I laid it against the drawing to confirm the shape (see **Photo 9**).

I used a standard wood lathe to remove the steps and produce the final exterior shape of the eggs. Before starting the shaping, I drew the mounting holes for the drivers on the mounting surface to make sure that any cuts did not go into the area where the drivers were mounted.

In comparison to conventional wood turning, two problems had to be solved. First, because of the flat front, the egg is out of balance when spun on the lathe. To correct for that imbalance, I mounted a steel plate of appropriate weight on the front, making sure that the holes used for the mounting screws did not interfere with the driver holes (see **Photo 10**). Second, since tools used with a conventional wood lathe simply lay on a tool rest and



Photo 8: The raw assembled egg



Photo 9: The template I used to check the egg's shape

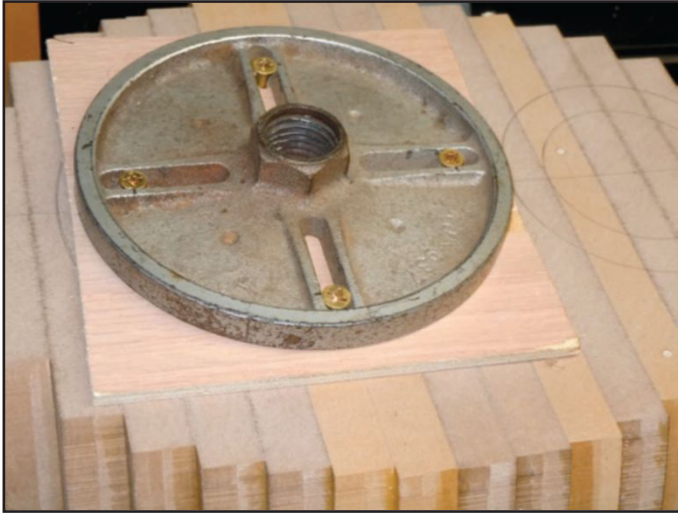


Photo 10: I placed a counter weight on egg.

are handheld, the continuous shape of the work allow consistent hand pressure to achieve the cutting. With the eggs, with every revolution when working on the areas where there is a flat, the pressure against the tool is released by the work when the flat area appears before the tool. This resulted in gouging as the round section reached the now forward positioned tool as the egg continued its rotation.

To mitigate this tool slippage, I did two things. I covered the tool rests with a rubber friction material to stabilize the tools. Second, I brought the tools into the work from the side instead of straight in. It still was very difficult as the friction surface did a good job of preventing the tool from slipping but also made it more difficult to smoothly advance the tool. Although not a physical problem, another problem was that I did not have a lathe and had not used one since high school shop class. Fortunately, I have a friend, Tom Meyer, who lives nearby and not only has a nice lathe



Photo 11: Tom Meyer at the lathe

but also a lot of experience using it (see **Photo 11**). He instructed me and started the initial work until I re-learned the techniques. Completing the lathe work on the two eggs took about 8 hours on two separate days.

When working with odd wood shapes and especially with the difficulty of working on the lathe with the flats, things do not always go as planned. There were places on the surface of the eggs where either the register pins showed through or until I got the technique right, gouges happened. I also still had marks from the lathe attachment points that needed to be fixed before finishing. To fix these problems, I discovered a new friend. 3M makes a Bondo two-part wood filler that is a lifesaver (see **Photo 12**). Once set up, the wood filler is very hard and sands extremely well. It is very fast acting, which is both good and bad. If you work in small areas and mix small amounts, there is little waste and you can get to the finishing steps quickly. If you mix too much at one

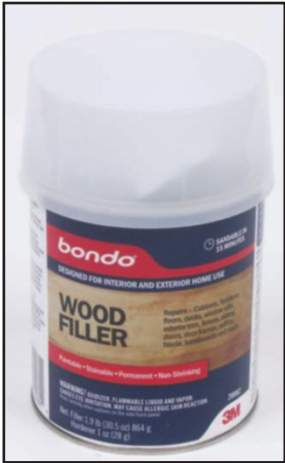


Photo 12: Bondo two-part wood filler



Photo 13: The unfinished shaped egg



Photo 14: Cradle built to hold egg



Photo 15: The egg with holes for the drivers



Photo 16: The unfinished egg with the drivers

time, it will set up before you get a chance to use it and you will waste a lot.

After the Bondo touch ups, I sanded the eggs, first using a half sheet sander and finally a palm random orbital sander with grits increasing to 800 for the final sanding. At the end, they looked more like eggs than beehives and were as smooth as glass (see **Photo 13**).

To cut the driver mounting holes and the hole for the Speakon sockets that were used to connect the eggs, I had to make another fixture. As you can imagine it would be quite difficult to work on the eggs with a router or drill press equipped with a Forstner bit because of the continuously changing shape.

To secure the eggs, I constructed a cradle from pieces of 2 × 4 lumber with surfaces that would mate with the eggs beveled. Some of the edges also had strips of rubber friction material fastened to increase slip resistance (see **Photo 14**). Once the egg was in the cradle, I used two clamps to apply pressure to the upright arms to hold it in place. In order to not damage the small top section of the egg from clamping pressure, I made a top plate from a piece of 3/4" MDF with a recess cut into it that fit the curve of the egg top. It was placed between the egg top and one upright arm also using an intervening piece of rubber.

Using a router with a sizing jig (e.g., the Jasper 200 Circle Jig) is usually a simple process. However, with the shape of the eggs, I had to use great care to assure that the router did not rock on the

surface of the egg. I made careful measurements of the drivers including outer diameter, depth of the basket mounting flange, and inside clearance hole. I set the circle jig to cut the diameter of the driver's outer diameter with a cut depth that would allow flush mounting of the driver. Then, I made the first cut. That cut had to be at least as wide as the mounting part of the basket, which in the case of the midrange required a second cut to get sufficient width with the bit size used.

Next, I reset the jig to make the cut for the driver's clearance hole. This part was again very tricky with the eggs as it was not possible to use a backup piece of material to stabilize the router when the cut went all the way through the material. If not careful, at that point, the router could move and cut into the mounting area of the egg.

Once I cut the circles, a few remaining actions were required. The terminals for the tweeters needed clearance slots cut to prevent interference with the mount. Then, pilot holes for the fastening screws had to be marked and drilled.

In the photo of the egg after holes were cut, the bottom diffuser pieces and connector mounting hole can be seen (see **Photo 15**). What is not visible is the recess cut into the rear of the egg at the connector mounting hole to allow for flush mounting of the connector. I made that recess with a Dremel tool and small router tip. I also made the pilot holes for the connector mounting screws at this time. After all the holes were finished, I mounted the drivers to confirm proper fit before



Photo 17: The Wixey digital protractor

the painting operations, which followed (see **Photo 16**).

The Great Pyramids of Egg

In keeping with the original design philosophy of the Mach 17 speakers, the eggs would rest on top of truncated pyramid bases that would hold the woofers. Unlike the Mach 17s, the pyramids would not be square at the base. To minimize the visual impact of the cabinet that had the required volume for the two 12" woofers, I decided to make the base rectangular with the small dimension of the rectangle

facing forward. I also wanted the top of the pyramid to blend smoothly with the bottom of the egg, which was round. The closest shape that would produce a rough match was a square that could then be rounded to the proper dimension. Little did I realize what that meant in terms of construction complexity.

The height of the pyramid was to be 40" which with the egg dimensions would put the centerline of the midrange and tweeter at ear height when seated. To decide the shapes of the 3/4" MDF panels that would be needed, I first did

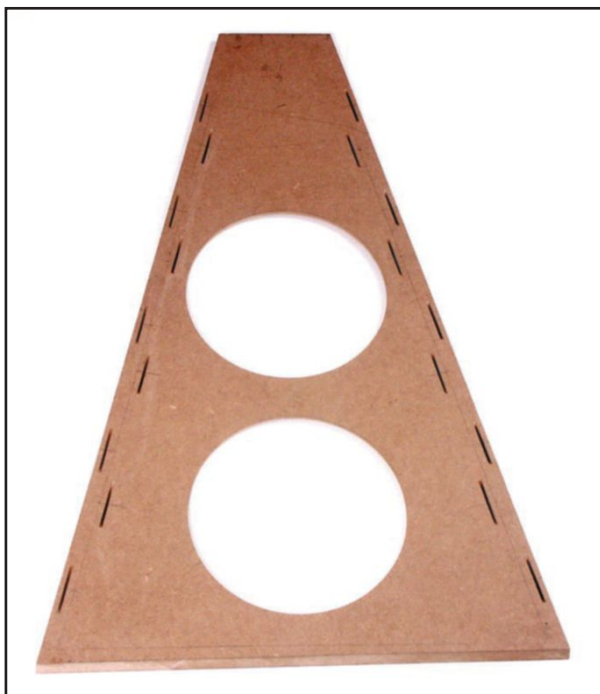


Photo 18: The inside of the front bass panel



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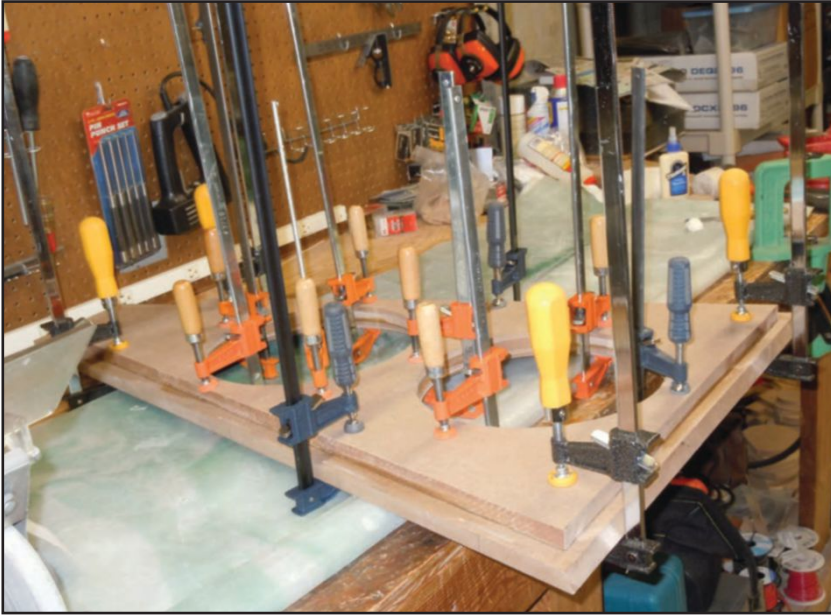


Photo 19: I clamped the front panel sections during gluing.



Photo 20: I clamped the fixtures to assemble the pyramid sides.

some rough drawings and then used an online trig calculator to determine the final dimensions. For the project, I ultimately needed four 4 × 8 sheets of MDF. I did the first rough cuts on my front lawn with a circular, leaving extra material around the edges so I could more easily handle them while getting them into my basement work shop. I made the final cuts on my table saw where I could control not only the shapes of the pieces but also the saw blade angles.

Why do I mention the blade and cut angles? As it turns out, when going from a rectangular base to a square top you wind up with some very strange angles. For example, on the front and rear panels the base to panel is an angle of 75.9°. The cut angle on the top is the reciprocal or 14.9°. For the side panels, the base to side is 69.2° and the cut angle on the top is 20.8°. On the sides, the angle where the sides and front and rear meet is not 90° but rather 85°.

Trying to set and maintain those angles on a circular saw while doing the initial cuts would have been effectively impossible. Even with the table saw it was a stretch. To set those angles, I relied on a Wixey digital protractor, which can set blade angles to 0.1°. It also has magnetic fastening capabilities to firmly couple to the saw table top and blade (see **Photo 17**). Did I mention this was not a first time speaker builder's project?

To allow recessing of the woofers so they were flush with the front surface, I had to use a double thickness of MDF. That also provided additional rigidity to handle the forces created by the high moving mass of the drivers. I made all the holes and the recesses in the front and rear panels before assembly. The construction method I used was with biscuits and glue. Biscuits usually allow precise positioning of the panels during the gluing operation and provide additional joint strength.

In the case of the pyramids, the severe angles from bottom to top resulted in additional challenges. For the front reinforcements, secondary panels were cut to size and biscuit slots cut on the mating faces (see **Photo 18**). If you look at the left and right sides of the inside of the front panel, you will notice two sets of biscuit slot rows. The inside rows are for the reinforcing panel and the outside for the mating sides. When fastening large panels like these together you need lots of clamps. The old woodworker saying is that clamps are like dollars, you can never have too many (see **Photo 19**).

Stopping the Resonance of Side and Rear Panels

Having built many other speaker projects using materials such as wood and steel, I learned that the best way to deal with cabinet resonances is

to prevent them from starting in the first place. Although various damping materials can help, they generally only lower the Q of the resonance without preventing the resonance in the first place. The best solution is to effectively break up the panels or supports into sizes that will not resonate at the frequencies being reproduced in the enclosure.

In this project, I essentially broke up the panels into many smaller surfaces by using internal ribs. For the rear and the side panels, before assembly I used the plate joiner to cut rows of slots for biscuits that were off center of the panels. Those slots would be used to fasten ribs to the panels. From those ribs, additional cross ribs would be clued in place further breaking up the panels into small shapes whose resonances were well above the frequencies being reproduced.

Assembling the Sides

Unlike square or rectangular boxes, the shapes and angles of the enclosures created significant problems during assembly. During the gluing and the clamping operation, the panels tended to slide out of position. In addition, the sloping surfaces caused the clamps to slip as they were tightened.

To solve those problems, I created three fixtures. Two were used to provide locating grooves for the clamps so they would not slide down the panel surfaces. I fabricated them from pieces of 2 x 4 studs with cross slots cut at positions along the length. I glued rubber friction material to the back side to help keep them from slipping (see **Photo 20**). I used the third fixture to keep the front panel from sliding down the sides. It had an adjustable base and a long arm with a right angle end stop to keep the rear panel in position during



Photo 21: The anti-slide fixture

the process (see **Photo 21**).

As soon as I started to assemble the sides, it became clear that because of the strange angles I could not glue each piece in sequence using the commercial right angle clamping fixtures I had. Each time I glued a panel, I had to dry assemble all the sides except for the one surface I was gluing. That way I could insure the alignment of the sides. Because of the time involved in the complete assembly process, I used Titebond III glue because it gives you double the work time before setup occurs. I also clamped the enclosure to one of my work benches to make sure the sides stayed in alignment. When each joint dried, I disassembled the unglued sections, glued another joint, and reassembled the complete housing. This

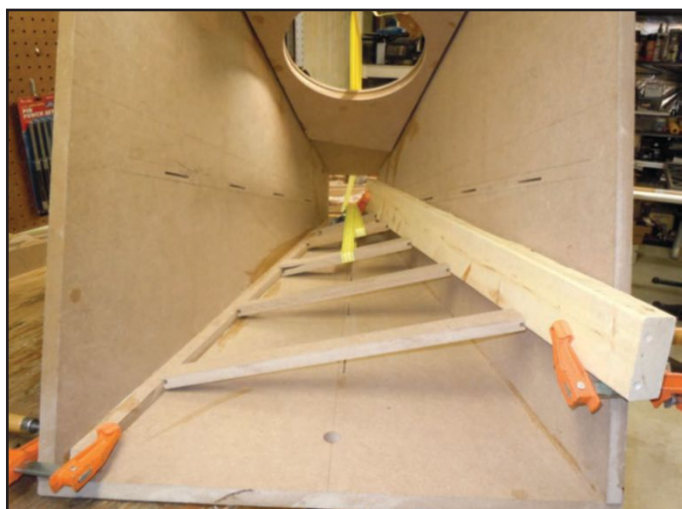


Photo 22: The fixture to hold the inside corner reinforcements

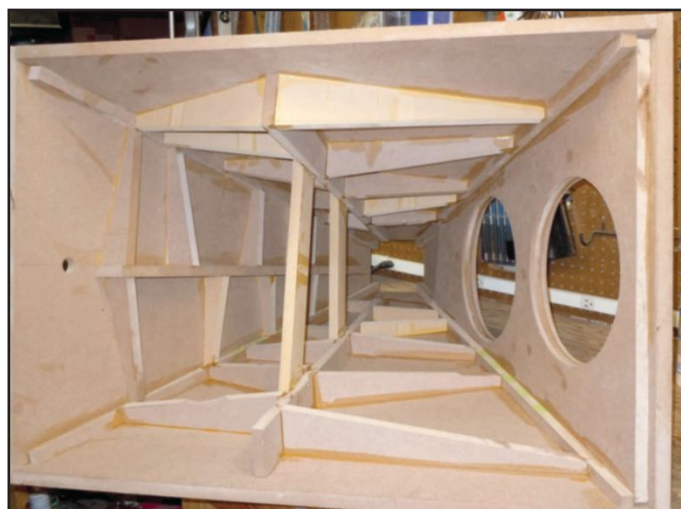


Photo 23: The inside panel braces



Photo 24: I used tie down straps to fasten the pyramid bottom.

continued until all sides were glued.

Since the top section of the bases were to be later rounded to mate with the bottom of the eggs, I needed to introduce additional material into the inside corners of the pyramids. That material would both provide additional stability and prevent cutting through to the inside of the housings when the rounding was done. I glued 3/4" strips of MDF with

the appropriate 85° angled edge into the inside corners. To hold them in place, I cut slots into the end of scrap strips of MDF and then wedged them between the strips and a piece of 2 x 4 clamped to the side of the housing (see **Photo 22**).

Adding the Internal Braces

Once I finished gluing the sides and the corner braces, I added the internal panel braces. The first step was to glue the main ribs using biscuits. When they were dry, I added the supplemental ribs, gluing them to the main rib and the sides at unequal distances and angles to produce asymmetrical panel shapes. Note that I fabricated these ribs from scrap pieces of MDF as the exact shape was unimportant. Since the two side panels were the widest and subjected to the largest bending moment, I added two cross braces between the main ribs on those sides. That way any bending forces were turned into tension and compression forces in these cross braces. MDF and other woods are vastly stronger in tension and compression than under bending forces (see **Photo 23**).

Closing the Pyramids

The only remaining parts necessary to close the pyramids were the top and the bottom pieces. Because of their size, the bottoms were the most difficult with which to work. Not only would the clamps I had not span the distance from the top to the bottom, the angles of the pyramids made clamping a poor choice. The fastening method for the bottoms was the familiar biscuit and glue process. The question was how to apply the compression necessary to insure proper glue joints. The answer turned out to be the fabrication of another fixture and the use of ratcheting tie downs normally used to secure cargo on trucks.

The fixture that I placed on top of the pyramid was made from pieces of 2 x 4 lumber designed to fit on top of the pyramid. The grid used several sizes of wood pieces to provide maximum resistance against bending as the tie down straps were tightened. I screwed the various pieces together using #8 x 3" drywall screws, starting from the inside section and moving toward the outside so that access to the screws could be maintained.

To fasten the bottom, I laid a sheet of plastic on my basement floor to prevent any glue drips from reaching the floor. I temporarily placed the pyramid on its top and filled the biscuit slots with glue and put biscuits in the slots. I added a layer of glue to the bottom faces of the sides alongside the biscuits. Next, I applied glue to the slots in the base and placed the base on the sides of the pyramid. I

About the Author

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

placed the tie down straps on the plastic sheet on the floor, inverted the pyramid, and placed it on the straps. I used pieces of waxed paper on top of the straps to prevent glue drips from attaching to them.

Finally, I centered the grid fixture on the top of the pyramid. I passed the straps across the grid in both directions and tightened them to provide the necessary tension to assure a proper glue joint with the base. The straps had to be repeatedly tightened in small increments around the fixture to keep it aligned (see **Photo 24**). I constructed the second pyramid in the same fashion.

Attaching the top sections were much easier because of their small size. The grid fixture was not needed as the tops were simply glued and placed in position with weights added on top. I used a combination of lead weights from my diving weight belt and cans of sand.

Shaping the Pyramids

One of my standard procedures when working with MDF is to round the edges to minimize the possibility of nicks. This was tricky with the pyramids as the edges were not at 90° angles. My rounding bit on the router was set at a shallower than normal

depth and after the routing was done I used a half-sheet sander to achieve the desired profile.

To insure a smooth transition from the square section at the top of the pyramids to the round bottom of the eggs, the tops of the pyramids had to be rounded. First, I drew a circle on the tops using a compass the same diameter as the base of the eggs. Then, I measured the same distance down on each edge of the pyramid from the top and placed a pencil mark.

The rounding was done from that top circle to each of the edge marks in an entirely freeform fashion using an electric planer set to a very shallow cut. I started at the top and worked down toward the bottom mark along the edge. Once I removed the majority of the material with the planer, I did the final shaping with a half-sheet sander. This was a very time consuming process and I had to exercise a great deal of care to assure that the right amount of material was removed at each point to achieve the correct shape. A mistake in this process could prove to be a disaster as the pyramids were already assembled.

In the next issue, we will discuss the passive crossovers and finish the project. 



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Designing and Testing the Omni Pipe

An Omni-Directional Stereo Speaker Pair



By
Ron Tipton
(United States)

Photo 1: This is a view of the rear wall of my listening room. The new omni-pipes are shown with my three-way columns and the feedback DVC subwoofer in front of the right side omni-pipe.

This article was inspired by Ken Bird's write up of his Omni Hex speaker pair in the December 2017 issue of *audioXpress*. Ron Tipton took Bird's idea and made a few tweaks of his own. He also included the performance data to see how well his Omni Pipes stacked up.

This project was inspired by Ken Bird's Omni Hex speaker pair. He didn't include any performance data, but perhaps he didn't have the test equipment. He also used a vented hexagonal wooden tube which has three pairs of parallel walls. So he had to add damping material to minimize the internal reflections. I wanted to use a design that improved these limitations and also to include performance data. I call my design the Omni Pipe (see **Photo 1**). The reason will soon become apparent.

The New Design

Instead of fabricating a tube, I used a pair of 40" length, 8" diameter schedule 40 PVC plastic pipe. I contacted a couple of plastic pipe vendors with

my only requirement—the cut ends must be square to the pipe's length to 1/16". FlexPVC in Nevada assured me they would meet this requirement and their location gave me a reasonable shipping cost. I ordered a pair of pipes for a total price of about \$200.

The ends were properly cut but the outside surfaces were, well... not very clean. I put the pipes, one at a time, in my bathtub and scoured them with a powdered cleanser (e.g., Ajax or Comet) and a stainless steel pad. When dry, I went over the outside surfaces with a small file to remove or improve physical blemishes (there were not many of these). I gave the outside surfaces a coat of Kilz Adhesion, which is recommended if you intend to

paint them, which I did.

Next, I had to decide how to make the tops and the bottoms. The top is also the speaker mounting platform and the bottom should be fairly heavy for good mechanical stability. I've used red oak in the past and I like working with it, so this seemed a good choice. I ordered a 4' length of 2 × 12" (Actually, 1-1/2" × 11-1/2") from Baird Brothers. This would be enough for both tops and bottoms, if I were careful enough. Unfortunately, I wasn't. I had to order another 2'.

Two Tops

The Top "top" plate (see **Photo 2**), which holds the reflector hemisphere and 1/2" × 6" dowel spacers, I made from an 11-1/2" diameter disk of 3/4", 13-ply Baltic birch plywood. I beveled the top edge with a table router. The two "tops" were anchored to each other with a stainless steel rod through their 1/8" diameter center holes. I drilled through the top and 3/4" into the oak top with a Forstner 1/2" diameter bit using the depth gauge on my drill press. I enlarged these holes in the oak top with a 17/32" bit and the drill press depth gauge. This Top also has the concentric PVC pipe mounting groove on its reverse side and it must be cut before removing the wood for the speaker, which also removes the 1/8" center pilot hole (see **Photo 3**).

Next, I drew a light pencil line perpendicular to an edge and through the center of the 1/8" hole. I measured along this line from the center of the 1/8" hole and put a pencil mark at 4" and at 4-1/2". With a 1/2" bit in my plunge router, I adjusted



Photo 2: The top, top plate is an 11-1/2" disc of 13-ply Baltic Birch plywood stained golden oak. The black hemispheres are Styrofoam sound reflectors. The wooden dowel spacers are glued to the plywood disc and just set in the slightly oversize holes in the oak top.

the router's position so the bit was between my two pencil marks. I made a light cut and checked the fit with one end of a pipe. (This is another application of measure twice but cut once.) When the fit was okay, I continued cutting until the groove was 1/2" deep.

For variety, I decided to make the oak top an octagon rather than leaving it square. This was



Photo 3: The top side of the red oak top shows the speaker cutout. The circular groove for gluing to the PVC pipe is on the other side and it must be cut before the speaker cutout is completed.



Photo 4: I built this jig from two C-clamps and a scrap of wood to cut off the corners off the two oak tops. I think the octagon tops are more attractive than their original square. However, I left the bottom oak pieces square for better stability.



Photo 5: I used a hot knife cutting tool to cut the Styrofoam reflectors down to the proper size.



Photo 6: A view of one of the oak bottoms shows the circular glue ring, a half inch wide by a half inch deep. The top corners have been rounded on the table router.



Photo 7: This is the corner spike and its holder before being pressed into the oak bottom. The top spacer covers the tip of the spike.

simply done using the setup shown in **Photo 4**. The table saw's protractor sets the 45° cutoff angle. I loosened the C-clamp holding the oak top, moved the top to the next corner, tightened the clamp, and cut. It took about a minute to make all four cuts.

I purchased two 8" diameter Styrofoam half-spheres to use as reflectors. I cut a 1-1/2" from each one using my hot-wire knife as shown in **Photo 5**. Using a "dab" of white construction caulk in several spots on the cut edge, I carefully placed the half-sphere in the center of the reverse side of the plywood disc. When dry, I filled the space between the sphere and the plywood with caulk, making a smooth transition between them. When the caulk had set, I painted the Styrofoam wire several coats of black enamel. It was water-based paint, but I still tested it on a scrap of Styrofoam.

I glued the 1/2" diameter dowel spacers into the reverse (bottom) side of the plywood top and used the oak top as a jig to keep the dowels properly spaced while the white glue dried. Then I gave the disc and dowels a coat of "Golden Oak" stain and allowed it to dry without wiping to preserve the deepest color. When this had dried overnight, both tops got three coats of clear, satin polyurethane varnish with a light sanding (220 grit) between coats.

I looked up how to bond PVC pipe to wood and an adhesive was recommend. I used Loctite PL Pro Premium because it's fairly soft at room temperature and can be applied with a standard caulk gun. Also, it does not set up immediately so I had some adjustment time.

I placed the oak top upside down on a piece of carpet. Then, I filled the tube mounting groove with the Loctite adhesive. Next, I picked up a tube and gently lowered it into the adhesive. I rotated the tube slightly to properly seat it. Then, I found something else to do for two days while the adhesive cure. If all the saw cuts and holes have been properly made, the plywood top will be 4-1/2" above the oak top.

Bottom

For the bottom, I drew a light pencil line perpendicular to an edge and through the center of the 1/8" hole. I measured along this line from the center of the 1/8" hole and put a pencil mark at 4" and 4-1/2". With a 1/2" bit in my plunge router, I adjusted the router's position so the bit was between my two pencil marks. I made a light cut and checked my work. Then, I continued cutting until the groove was 1/2" deep.

I drilled a 1/2" hole through the bottom for the speaker wire. You can be arbitrary about the location

because there is no crossover and no front and back sides in an Omni speaker. I added a banana binding post connector, a Dayton Audio 260302 (see **Photo 6**).

I had already purchased two sets of spike feet, where the spikes screw into metal inserts in the bottom. The instructions say start with a 1/2" diameter hole for the inserts but I knew red oak to be rather tough so I used a 0.5975" drill. I followed the online installations but it didn't work out well. It appears red oak is tougher than I thought—so I pressed them into place with my hydraulic shop press (see **Photo 7**).

I placed the oak bottom upside down on a piece of carpet and filled the pipe mounting groove 1/4" deep with the Loctite adhesive. I picked up a tube and gently lowered it into the adhesive. Next, I rotated the tube slightly to properly seat it. Then, I found something else to do for two days while the adhesive cured. I tried to keep the top side vertically aligned with the bottom side, but if you try it, have a 4' length of 1 x 4 at hand to check it. Again, wait two days while curing takes place.

With the two nearly completed pipes lying on the workbench, I cut two 6' pieces of two-



Photo 8: The twin alligator socket connects the omni-pipe to the power amplifier. The two conductor wire routes over the edge and through the 1/2" hole and through the fiber-fill up to the speaker. As a final operation, I caulked the 1/2" hole.

conductor speaker wire (AWG 18), which I fed through the bottom 1/2" holes leaving about 1' for connecting to the double banana connector shown in **Photo 8**. I fished the in-pipe ends up to the top of the pipe and used some masking tape to hold them in place.



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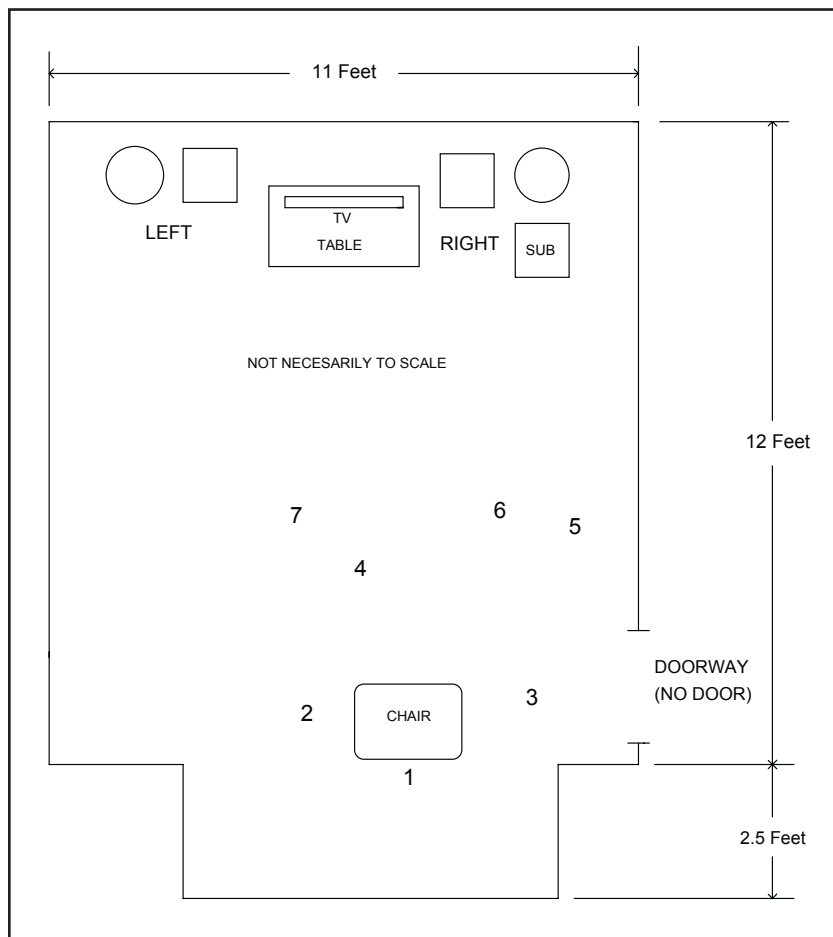


Figure 1: This is the floor plan of my listening room. The dimension numbers are correct but the drawing is not necessarily to scale. The numbered locations indicate the Room EQ Wizard microphone locations.

Speakers

I also spent some time looking at 8" coaxial speakers, 8" drivers, and separate tweeters that could be paired to make a coaxial. In the end, I came to the same conclusion as Ken Bird, the Goldwood 8004 was the best buy, so I ordered two from Parts Express. This model does have a downside—the Thiele-Small (T-S) coefficients are published for just

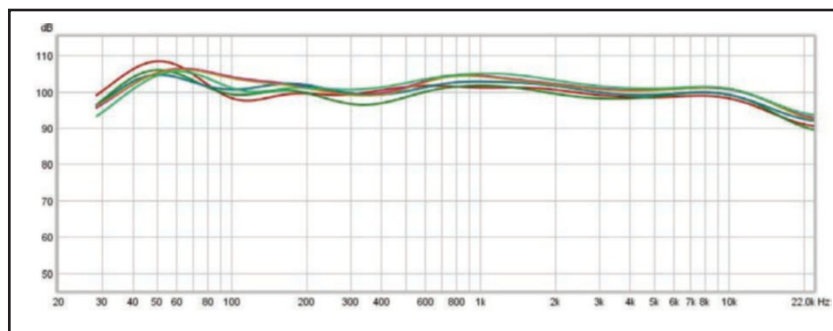


Figure 2: The frequency response curves for the seven microphone locations shown in Figure 1. Although not flat, the similarity of the curves shape is a good indication of an omni-directional sound field.

the 8" driver with no information on the tweeter. A Bass Box Pro simulation, which ignores the tweeter, shows that a sealed box for this size PVC pipe is too small. But I compensated for this the best I could by filling each pipe with about 6 oz. of polyester fiber fill.

I put the speakers in place in their oak tops and marked the screw locations with a pencil. With the speakers removed, I drilled a 1/8" hole at each mounting position. I applied stick-on sealing tape to the underside of both metal flanges and used 1/4" fast-ons to connect the wire—observe polarity! The 1/8" holes are the right size for a #6 3/4" sheet metal screw. I gave each pipe a coat of "Sweet Treats" Valspar semi-gloss, which goes on smoothly over the Kilz Adhesion primer.

Testing

I used the free Room EQ Wizard (REW) for frequency response measurement. Testing an omni-directional speaker system is somewhat different from a directional system because the sound is supposedly coming at you from all directions. I used the seven microphone positions in **Figure 1** with the microphone kept pointing to the center of the sound field: the TV set center. The microphone was a Dayton Audio EMM-6 electret, which needs a +48 VDC polarizing voltage so I used a TDL Model 411 preamp.

Although the curves are not flat, they all have the same shape—a pretty good indication of an omni sound (see **Figure 2**). The differences in amplitude are likely due to the difference in speaker to microphone distance.

RT-60

RT-60 is the length of time it takes a signal to decay to 60 dB below its original level. REW calculates this based on the slope of the Schroeder curve. **Figure 3** is the RT-60 of the averaged frequency response in REW and close to the optimum of 200 mS to 300 mS for a small room.

Project Files

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

Sources

Baird Brothers Sawmill | www.bairdbrothers.com

Bass Box Pro | www.ht-audio.com

Room EQ Wizard | www.roomeqwizard.com

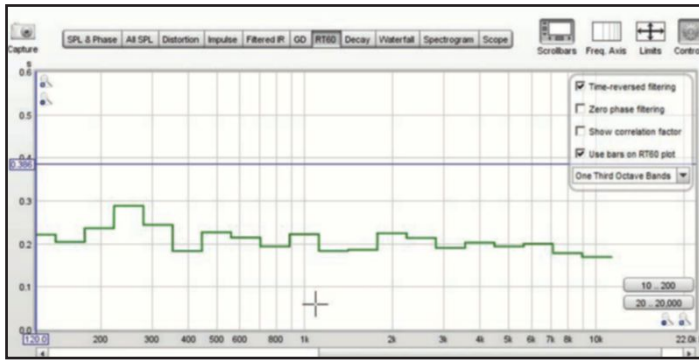


Figure 3: The RT-60 curve from the averaged frequency responses shown in Figure 1 mostly lay between 200 and 300 mS. This is considered a good range for a small room.

Final Remarks

Particularly for orchestral and pipe organ music, an omni-speaker system has an excellent sound stage. I was somewhat concerned about the spacing between the pipes, the three-way columns (6" apart) and the walls (13" apart) as shown in **Photo 9**. And Bird recommends about 24". However, the music sounds light and uncramped with my measurements confirming the arrangement. Yes, it was a bit of work to build them, but I'm very pleased with their performance. 



Photo 9: I was somewhat concerned about the spacing between the pipes, the three-way columns (6" apart) and the walls (13" apart). However, the music sounds light and uncramped.

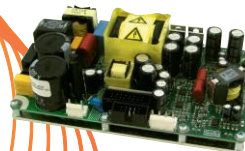
hypex NCxxxMP series

overview

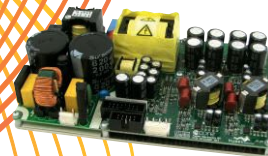
The NCxxxMP amplifier module incorporates a low power standby power supply (meets 2013 ERP Lot 6 0.5W requirements), a highly efficient switch mode power supply and a high-performance Class D amplifier in one compact and easily applicable power brick.



NC122MP



NC250MP



NC252MP



NC500MP



NC502MP

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



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Using Technology to Encourage Self-Expression

An Interview with Michael Goodman, Founder of CEntrance

Founded in 2000, CEntrance develops, licenses, and distributes cutting-edge audio products worldwide. The Chicago company CEntrance is known for its co-development work with audio industry leaders and its innovative designs, inspired by its founder and admittedly powered by...coffee and tacos.

By
Shannon Becker
(United States)



Michael Goodman, founder and CEO of CEntrance, leads a dedicated team of engineering, manufacturing, and marketing professionals in bringing quality products to market in record time both as an outsourced technology provider and product manufacturer.

SHANNON BECKER: Tell us a little about your background.

MICHAEL GOODMAN: In Moscow, where I grew up, the winters were long and cold, and I walked to school in the dark. When I was 12, I built a transistor radio from a parts kit and it was mesmerizing. This tiny device transformed my morning walks. A confident voice from a warm, far-away studio suddenly made me feel less alone.

Audio gradually became a passion and I chose to study Analog Electronics in college. When I came to Chicago, I enrolled in Columbia College to study Audio Engineering. At night, I worked in studios around town, recording anything I could get my hands on, and learning the craft from the local pros. My first “real” job was at Shure, where I started as a design engineer and worked my way up to managing analog and digital R&D groups. At night, I studied Computer Science, and then got an MBA. That finally gave me the confidence to start an electronics company.

SHANNON: You have developed many high-end portable DACs and interfaces. How did you get involved in this field? What were some of the biggest challenges when you first started out?

MICHAEL: CEntrance started out as a design firm with a mission “To Work for every Audio Company in the World.” We assembled a group of talented engineers and created a portfolio of reference designs in FireWire, USB, and PCI audio interfaces. Then we shopped these designs around, by exhibiting at the Audio Engineering Society (AES) conventions and other shows. After about seven years, we realized that we had achieved our mission. By then, our client list read like the A-to-Z of the Audio Industry—we have successfully worked with dozens of companies, literally from Alesis to Zoom.

At this point, we decided to start our own product brand and released the MicPort Pro, a breakthrough product concept that ended up being a long running hit with Broadcasters and Podcasters around the world. MicPort Pro is a cigar-shaped

interface that plugged into any professional mic and instantly transformed it into a high-quality USB mic. We started a new product category and hit on a trend of portable recording and MicPort Pro just celebrated its 10th anniversary..

DACs came a couple of years later. I met up with Jude Mansilla of head-fi.org and we discussed the state of portable audio. At that time, in 2009, there were very few options for portable Hi-Fi listening and we decided to enter this category with the DACport, a small cigar-shaped audiophile DAC. (No, I don't smoke cigars; it just seemed like a good form factor at the time). DACport proved quite popular and ended up on the cover of *Stereophile* magazine. We followed that up with DACmini, HiFi-M8, and several other portable DAC/Amps. We realized that we excel in personal audio and our motto became "Making Audio Personal."

Having worked for years in both Pro Audio and Consumer Audio we learned about the different preferences of customers in these two contrasting fields. Sometimes the preferences are downright contradictory, as some musicians seem to pay more attention to producing good sound and ease of work flow rather than pristine hi-fi listening playback. On the other hand, audiophiles can easily sacrifice the weight and usability in search of perfect sound quality.

SHANNON: Tell us more about CEntrance (www.centrance.com).

MICHAEL: CEntrance continues to be distributed and nimble. We still do some consulting work, and therefore, our staffing needs change often. While our core staff is small, we rely on a large network of folks we know and work with on project basis. We are a very international company, with resources located across the world. We just moved our headquarters close to the O'Hare Airport—the airline industry is part of our support network. The other pillars of our support are coffee and tacos.

SHANNON: In the 18 years CEntrance has been in business, how has it changed?

MICHAEL: We started out as a service company, a design firm. That taught us to be humble and understand the value of customer service. Now that we are a product manufacturer we try to approach all of our customers like clients – we are grateful that they choose our products. We work closely with them to understand their world.

SHANNON: CEntrance now supplies USB technology

to many established audio brands and is a partner with other semiconductor companies, supporting audio development. How and why did you move in that direction?

MICHAEL: After Shure, I got a taste of the excitement in the VC-backed startup world. For a while I was VP of Product Development at Music Now, a music streaming startup, which was later sold to AOL. The ups and downs of the fundraising game were a thrill, but also a distraction, as I was trying to focus on product engineering. CEntrance started as a design firm as a way to seek stability. We began working with a number of client companies to ensure a steady stream of challenging technical assignments, and lessen the focus on market fluctuations.

SHANNON: What are some of the challenges that you faced in this highly competitive market?



MicPortPro, CEntrance's first product, is a USB recording interface for professional microphones (a) and is the only portable mic preamp with a 24-bit/96 kHz A/D converter, switchable 48 V phantom power, loud headphone amp, and studio-grade control knobs



MixerFace R4 is a portable mixer and mobile recording device that can transform a smartphone into a full-featured portable recording studio and it is about the size of a US passport.

MICHAEL: We have survived a number of ups and downs and there was no shortage of challenging times. From simple things like component shortages, when you cannot buy parts to make product, to more complex things such as the changing landscape of retail and the move to the online economy, there are always enough things to worry about. Fortunately, we have a strong, dedicated team, a reliable supplier network, and our wonderful customers, who continue to inspire us, as we work hard to inspire them.

SHANNON: Tell us what makes CEntrance different.

MICHAEL: We try not to make “me too” products. We like to surprise our customers with the unexpected, enrich our products with new technologies, and deliver more performance than anticipated.



MixerFace R4 is a mobile recording interface allowing to capture sound anywhere using a smartphone—iPhone or Android phone—tablets and laptop computers.

pated. That’s a great source of the “wow” factor and hopefully would encourage people to stick with the CEntrance brand.

SHANNON: What is your role within the company?

MICHAEL: I work with our sales and marketing staff, work trade shows and presentations, but also enjoy my time in the lab, inventing new product concepts and working with our brilliant engineers and technicians. Deep down inside, I’m a product designer, a musician, and a recording engineer. All of those skills culminated in creating the MixerFace R4.

SHANNON: Can you tell us about the MixerFace R4 mobile recording interface?

MICHAEL: MixerFace R4 is a Mobile Recording Interface. It’s a new type of device, because it’s really two products in one—a Mixer and an Interface. The addition of the internal rechargeable battery means you no longer have to waste money on buying AA batteries as you take this product with you to record anywhere. You can record right into your phone or stream a concert or a live event—the phone gives you that instant path online and MixerFace provides the broadcast-quality audio connections.

MixerFace R4 has two inputs that can take mics as well as electric guitars, basses, etc. It’s great for songwriters and gigging musicians who find themselves on the road a lot—on the tour bus, in a hotel room, even on an airplane. And it’s equally convenient for Broadcasters and Podcasters who record episodes or interviews on the go. We feel that the product will also appeal to videographers, as it improves DSLR audio quality for location shoots.

Recording, post-production and uploading can be done from the field, with professional quality. That’s what MixerFace R4 is for. You are no longer tied to the studio, and even your laptop is the thing of the past. Although MixerFace R4 works perfectly with laptops and desktop computers too.

SHANNON: What do you feel is CEntrance’s most successful product to date and why do you think it is successful?

MICHAEL: We feel that MixerFace R4 is our best product yet. It is based on all of the research we have done to date—it incorporates what we know about modern battery technology, portable recording needs, miniaturization and manufacturing techniques, and our experience building tough products that survive the rigors of daily use. We really think it will be a fun addition to every musician and

podcaster's toolkit. And since we use a lot of our audiophile circuitry inside MixerFace R4, it will not disappoint in the sound quality department, either!

SHANNON: What's next for CEntrance?

MICHAEL: Our recent consulting focus has been on applications of IoT in media. We have developed personal assistants that feed all kinds of data to an AI engine for several clients. There is a lot of new interesting technology there that is making its way into traditional audio products. Our new mobile audio platform transforms news reporting and music broadcast by making live, high-definition streaming affordable. We will continue to use our technology portfolio and customer focus to push the envelope.

Innovation is what we do best. We like developing products that give our customers the joy of self-expression through audio. You can master audio on many levels. As a listener, you can train your ear to perceive minute differences in equipment. As a performer, you can create moods that improve people's lives. That's ultimately what we want to support with our work. 



Michael Goodman uses the MixerFace R4 to podcast.

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- ✓ Quality Control script processor
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CLIO 12



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Industry Calendar

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

October 5-7, 2018

Rocky Mountain Audio Fest (RMAF)

Marriott Denver Tech Center, Denver, CO

www.audiofest.net

CanJam RMAF

www.canjamglobal.com/rmaf2018



The 2017 edition of the Rocky Mountain International Audio Fest (RMAF) was great. The 2018 and 15th edition of this annual gathering—the largest consumer audio and home entertainment show in the US—should be even better. Denver's own audio wonderland now features more than 160 exhibit spaces representing more than 400 companies from North and South America, Europe, Australia, and Asia. Attendees are encouraged to go from room to room, interact with salespeople and engineers, have a seat, and casually listen. CanJam @ RMAF is a great show-within-a-show, organized by Head-Fi.org, and it features the latest in headphone and personal audio technology.

October 16-18, 2018

The Loudspeaker Sourcing Show 2018

Nansha Grand Hotel in Guangzhou, China

www.loudspeakersourcingshow.com



The Loudspeaker Sourcing Show was created to connect suppliers directly with the companies that need their services. The exhibitors at this show provide OEM/ODM services in the manufacturing of loudspeakers and related products and components. This year, the show follows Prolight+Sound Shanghai (October 10-13) and starts immediately after the international Hong Kong Electronics Fair (Autumn Edition, October 13-16) trade show.

October 17-20, 2018

AES New York 2018

145th International Convention

Program: October 17-20 | Exhibition:

October 17-19, 2018

Javits Center, New York, NY

www.aes.org/events/145

NAB Show New York

October 17-18, 2018

www.nabshowny.com



The 145th Audio Engineering Society New York 2018 Convention will offer four days of Technical Program presentations, workshops, discussion panels, and more (October 17-20), as well as the year's largest dedicated professional audio showcase and exhibition, taking place the first three days of the show (October 17-19). The convention will be once again concurrent with the independent NAB Show New York 2018 event (October 17-18) in an adjacent exhibition hall. On the exhibition floor and throughout the convention hall, there will be showcases and demos from leading names in professional audio manufacturing and services, along with the popular Project Studio Expo, Live Sound Expo, and other specialized events, all free-to-attend. Additionally, AES Student and Career activities will include recording and design competitions, a new MATLAB plug-in design competition, the annual education and career fair and other ways to network with students and professionals from around the world. A new series of DIY events will offer insight and instruction on planning, design and construction of audio hardware and DSP. Hands-on demonstrations will offer attendees the chance to learn more about DIY audio device design and construction, following the tradition of do-it-yourself innovation and promoting the understanding of audio technology.

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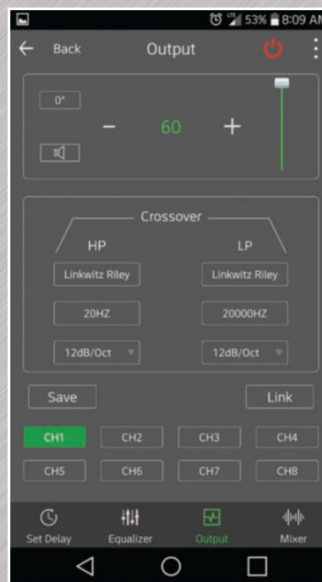
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