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Editorial Inquiries:
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audioXpress, Editorial Department
P.O. Box 417
Chase City, VA 23924, US
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New Audio Sensors, Waiting to be Measured

Test and Measurement is a fascinating topic, given that requirements never cease to expand. While test and measurement companies respond to audio research demands, that same research generates higher demands in creating reliable, repeatable, and quick processes for production and quality control tests.

As I wrote previously, there's nothing more challenging. The people who daily provide the research on which the audio industry evolves, depend on test and measurement tools to generate the data to measure and demonstrate their areas of work. Meanwhile, the test and measurement companies must follow those cutting-edge research challenges very closely to develop and provide the tools upon which that research depends. The diversity of the field work needed to understand continuously improving areas of research and convert that knowledge into practical solutions that can verify and certify all the generated data, is precisely what makes test and measurement instrumentation so interesting.

And considering what has happened in the audio industry, just in the past two years, it becomes clear how challenging this is. Just read some of the topics that are covered in this edition of *audioXpress*.

Daniel Knighten (Listen, Inc.) and Glenn Hess (Indy Acoustic Research), write about some early challenges of measuring smart speakers—one of the product categories with the fastest adoption cycle in the history of consumer electronics. The authors know this represents one of the most formidable challenges for the audio industry, with the increased use of more sophisticated arrays of sensors and DSP functionalities being implemented in completely new designs, which sometimes don't even offer any other type of input other than the audio signal streamed from a cloud service. As more manufacturers dig deep into the research for implementing differentiated products in the "smart speaker" category—in itself, constantly evolving—test and measurement companies are trying to respond with the adequate solutions to support those efforts, while at the same time, factories are struggling to find the most effective ways to evaluate and control these products' production quality, with all its numerous features and components. And bear in mind, that smart speakers are currently being assembled in quantities unheard of for any other speaker category.

More importantly, as Knighten and Hess write, "Audio quality is a strong decision-making factor for customers and a key selling point for manufacturers, with each manufacturer incorporating its own unique speaker design and positioning for optimum sound quality."

And remember headphones? Not long ago the fastest growing product category in the consumer audio industry (now superseded by smart speakers,) headphone research and development continue to face its own set of challenges, with the need to measure an extended range of frequencies and respond to increasingly demanding ANC applications.

Recently, in response to those challenges, G.R.A.S. Sound & Vibration released its new RA040X ear simulator series, able to measure headphones reproducing content up to 20 kHz or even to 40 kHz bandwidth, when working with high-resolution audio formats. And that's why we included a timely review of these brand-new devices, written by Brent Butterworth, someone who understands the pressing need to perform those measurements in its daily review activity.

From another similarly cutting-edge perspective, we look at what can also be considered an historic and significant milestone for the audio industry—with profound consequences that are easy to predict in the near future. In addition to the new IK Multimedia ARC 2.5 Advanced Room Correction System review, written by Ron Tipton, we decided to look deeper at the included measurement microphone and asked Stuart Yaniger to perform measurements and a comparative evaluation. This, because IK Multimedia was the first company (to our best knowledge) to actually design and ship a MEMS-based measurement microphone, replacing the electret-based model used in earlier versions of the system. Yaniger measured the new MEMS microphone, compared it with the previous ECM model and with a calibrated PCB Piezotronics pre-polarized condenser, and discovered that, even with significant limitations, the MEMS unit already provides a significant upgrade, with noticeable virtues. Like many other fast-rising categories in the electronics industry, MEMS microphones (and speakers) are now getting ready to "crash the party."

It is clear that there is no shortage of tasks to stimulate our devotion for audio measurements.

J. Martins
Editor-in-Chief

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COLUMNISTS

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

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

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

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

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

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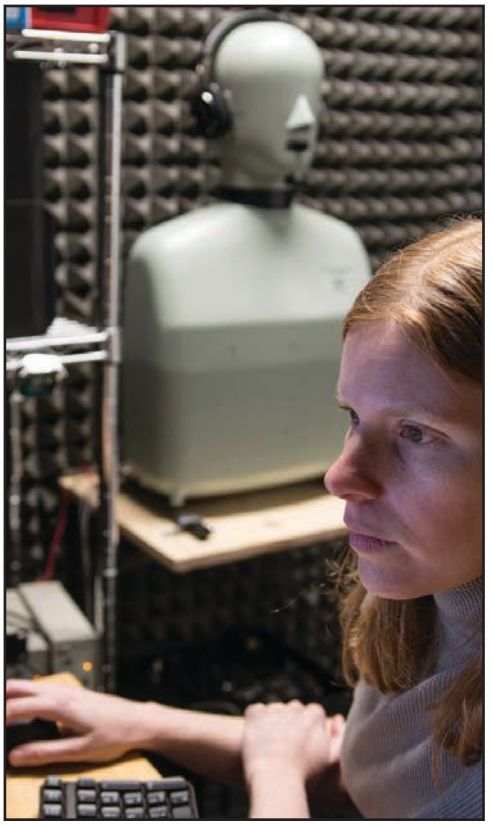

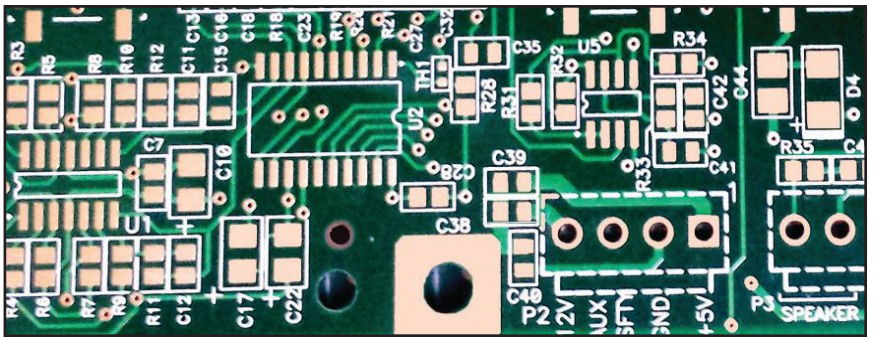

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An Audio Legend Remembered

C. Victor Campos 1933-2017

Photo 1: This ad for the KLH Model Thirty-Three loudspeaker appeared in the September 1970 issue of *High Fidelity* magazine. It beautifully sums up Victor's many talents and colorful personality. (Image courtesy of the American Radio History Archive, www.americanradiohistory.com)

Like so many people in the audio community, I was saddened to learn of the passing of my friend Victor Campos on December 12, 2017, at the age of 84. For more than four decades Victor was an important figure in the audio industry, beginning at Acoustic Research (AR) in 1960, KLH in 1963, and returning to AR in 1974. During this period he held positions in both customer service and engineering. Victor was a unique individual, and the 1970 ad for the KLH Model 33 loudspeaker (see **Photo 1**) pretty well sums up his many talents and colorful personality. One of his most ambitious projects was AR's second series of "Live vs. Recorded" demonstrations, during his second appointment with that company, to illustrate the accuracy of the AR 10π loudspeakers. Victor had worked with Edgar Villchur on the first "Live vs. Recorded" demonstrations in the early 1960s, some involving the Fine Arts Quartet. For the new demonstrations, he produced a series of very demanding drum set tapes, all superbly engineered by John Newton.

I first met Victor in 1989 when he was Product Development Director at Adcom. Although we both spent our lives working in the audio field, it was our mutual passion for classical music that sparked the beginning of our 28-year friendship. On more than one occasion, Victor mentioned that his life-long interest in classical music began when he first saw Walt Disney's *Fantasia*, which featured the Philadelphia Orchestra conducted by Leopold Stokowski. Indeed, Victor's love of music was always the driving force behind his many contributions to the audio industry, throughout his long career. In the hundreds of phone conversations I had with Victor during the nearly three decades that I knew him, we spent far more time talking about music than we did about audio.

I was fortunate to have had the opportunity to review three products that, in retrospect, are probably his crowning achievements at Adcom—the GTP-400 tuner preamp, the GFP-565 preamplifier, and the ACE-515 AC Enhancer. For the development of the GTP-400 and the GFP-565, Victor engaged the op-amp design expertise of Walt Jung. The

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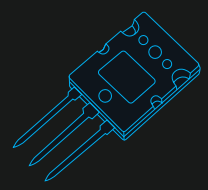

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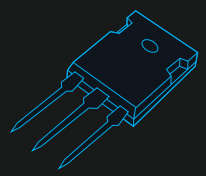

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results were two products that offered remarkable performance in their price categories. (They originally sold for $395 and $795 US, respectively.) When the ACE-515 was introduced, few in the audio industry were paying attention to the importance of clean AC power. The ACE-515, priced at a very affordable $199 US, convinced me and many others that good audio equipment could never perform to its potential when fed by a noisy power line. Victor had the common-mode choke wound with 14-AWG wire, and the general consensus among audio reviewers was that it was one of the few power line filters that didn't current-limit power amplifiers.

Victor was never interested in producing *ne plus ultra* products that only a handful of audio enthusiasts could afford. His goal was always to manufacture products that offered real value to music lovers. The GFP-565 may have been the best full-featured preamp for less than $1,000 at the time it was introduced, and the revamped Adcom power amplifier line produced during his tenure were also favorably received in the audio press. But, Victor later told me that the product he was most proud of was the GTP-400, since he felt that if offered the greatest value of anything he produced for Adcom.

After leaving Adcom, Victor became Product Development Director for NAD, later returning to Adcom, and finally going back to NAD before retiring in the early 2000s. One of the most interesting products made during his tenure at NAD was the model 118 digital preamplifier. The 118 had four S/PDIF digital inputs and a pair of digital outputs, with advanced digital signal processing for tone controls, infrasonic filtering, stereo width and spread, FM hiss reduction, and a compressor/expander—all switchable. The 118's main digital output could be connected to any powered loudspeaker or power amp with a built-in DAC, for an all-digital signal path up to the point of analog amplification. A full array of analog inputs and outputs were also included, and the DSP-processed signals could also be routed to the tape outputs for recording.

In the 1970s Victor hosted a radio program on Boston's WGBH-FM called "Adventures in Sound." His goal was to show just how good FM stereo could sound if properly executed. By that time, Victor had made many contacts in the recording industry, and had acquired several thousand 15-IPS, Dolby-A reel-to-reel copies of master tapes from several major record companies, including RCA Victor, Columbia, Deutsche Grammophon, and Vox/Turnabout. Victor revamped WGBH's audio chain and, rather than decoding the Dolby-A tapes in the studio, he located the Dolby-A decoders at the transmitter site. This ensured that any noise picked up between the studio and the transmitter would also be attenuated by the Dolby-A decoders. When Victor did his program, all compressors and limiters were bypassed, allowing listeners to experience the full dynamic range of the tapes. I never heard Victor's radio program, but eight years ago Victor sent me several of his favorite tapes for digital transfer, for his personal use. Although I was familiar with many of these recordings from commercial LP and CD releases in my own collection, I was amazed at the sound on these carefully prepared tape copies. They were far superior to any of the commercial releases, and showed just how much is often lost in the translation. I'm sure that Victor's listeners in the Boston area were equally impressed.

One of the most prized recordings in Victor's tape collection was of Bach's *St. Matthew Passion*, an uncut performance done in 1969 at the Marlboro College Concert Hall, featuring the Brattleboro Community Chorus, Marlboro College Chorus, and the Brattleboro Music Center Festival Orchestra, conducted by Blanche Honegger-Moyse. Victor attended that concert and said this performance had convinced him of the greatness of that work. It is certainly one of the finest performances of the *St. Matthew* that I have ever heard, and transferring these tapes for Victor was a great pleasure. The recording was superbly engineered by Marc Aubort, and was undoubtedly a treat for WGBH-FM listeners when Victor broadcast it on his program. Victor also loaned me the "Live vs. Recorded" tapes he produced, so I could hear first-hand the remarkable realism and dynamics of those recordings. They must have given the AR 10π loudspeakers a workout!

In 1989, I was asked to write an entry on the history of the loudspeaker for the *Encyclopedia of Recorded Sound* in the United States. During his tenure at AR, Victor worked with company President and co-founder Edgar Villchur, inventor of the acoustic suspension loudspeaker, and with Roy Allison, AR's Chief Engineer. He also worked closely with Henry Kloss during his tenure at KLH. As a colleague of these "primary sources," Victor amassed a considerable amount of technical and historical knowledge related to the early history of the acoustic suspension loudspeaker, information he freely shared with me during my preparation of that article. Thanks to Victor, I was able to include information not found in any other sources. As Chuck Hansen notes in his piece that follows, Victor wanted to start a new company. But, multiple health issues prevented him from getting this new venture off the ground. I contributed a phono preamp design for the ADD-345 stereo preamplifier that Chuck mentions.

Those who worked with Victor could find him demanding and sometimes uncompromising. But, he drove himself at least as hard as those around him. And, he was quick to express his gratitude and pleasure for work well done. I feel privileged to have known Victor and will long remember our many stimulating conversations. Our thoughts and prayers go out to Lillian Janik and Victor's family

—Gary Galo

*****

Back in 2008, Victor Campos asked me to design a headphone amplifier he wanted to market under a new name. I used OPA604 op-amps and BUF634 output buffers, with mid-frequency cross-talk feedback to prevent "inside the head" imaging. I built it in an aluminum Bud Minibox clam-shell enclosure. He told me it was the best headphone amp he ever heard, and he paid me for my time and cost of parts.

He was going to have a company he worked with in Taiwan lay out the board and chassis and build it for him under a new company name, ADD (Analog-Digital Design; he tried but could not buy the Adcom name rights). He never mentioned who was going to design the "digital" part of the new audio equipment line he was planning.

If the headphone amp was successful he wanted me to help him design a preamp called ADD-345, which would also incorporate the headphone amp. I still have the Adcom GFP-345-series chassis that he wanted to use to package the prototype ADD-345 (the GFP-345 was a preamplifier that Adcom marketed outside the US). He sent me two blank GFP-345 preamp circuit boards on which to build the prototype. The phenolic "yellow board" was an earlier layout, and the epoxy "blue board" was the last GFP-345 layout. It would take some cuts and jumps to put the circuit on the existing boards, but it was do-able. Unfortunately, the cost to tool and build either unit exceeded what his Taiwan designer set as an upper limit for profitability.

Victor and Lillian were great people, and he always wanted to talk about his dog Quigley (named after their mailman) and our dog Koli. Ed Dell said he would have me write an article about the new equipment for *audioXpress*, once the headphone amp was finished and offered to the public. Sadly, multiple health issues prevented Victor's Analog-Digital Design project from materializing.

—Chuck Hansen

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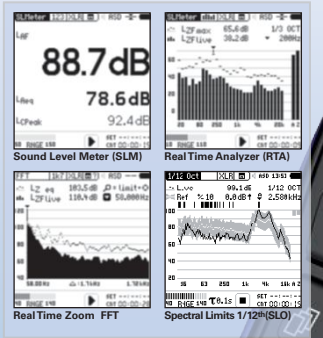

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Audio Editing Software Roundup (Part 4)

Audio Tools But Not DAWs

By
Fernando Rodrigues
(Portugal)

In this article, Fernando Rodrigues concludes a roundup of software audio editors currently available in the market. Learn what differentiates audio editors from the so-called Digital Audio Workstations (DAWs), how those tools have historically evolved, and what features to notice. To conclude the roundup, we look at Ocenaudio, which is donationware and Internet Co. Sound It! 8 Pro.

Ocenaudio

Ocenaudio is donationware. This means it is free, but the author(s) appeal to our good will for a donation. It's one of the two in this roundup that is "truly" cross-Platform (meaning that it runs in all three major operating systems—Windows, Mac OS X, and Linux). It is also a 64-bit application. All versions of Ocenaudio have a uniform set of features and the same graphical interface, which means that, once the application is launched, the user will feel at home no matter which OS it's in.

It supports up to eight channels of audio, which also means audio formats up to 7.1 surround. It works with sample rates up to 192 kHz and resolutions up to 32-bit linear. In terms of supported and recognized file formats, the list is comprehensive: AAC, APE, ASIG, AC3, AIFC, AIFF, CAF, NSP, DTS, IMA, FLAC, MP2, MP3, M4A (AAC and Lossless), ADPCM, AU, OGG, PCM, PCM a-law, PCM μ-law, Wsig, W64, TTA, Text Decimal (DEC), Text Hexadecimal (HEX), Text IEEE Float (FLT), Text Octal (OCT), WAV (several kinds, like a-law, mu-law, AAPCM, etc.), and WavPack (WV)

Support of plug-ins is a breeze. It performs the usual scan the first time we open it (using the system VstPlugIns directory and two other directories). We can define other directories and the application will scan those new directories in the next launch. The scanning process is long and tedious but having the possibility to define which directories should be scanned is good if, for example, we want Ocenaudio to not scan a plug-ins folder that has lots of virtual instruments (like the one we have) and use another one dedicated to audio editing. Anyway, this is the correct way to support and work with VstPlugIns, and way ahead from the convoluted format we experienced with other programs in this roundup. The program supports VST2 in Windows, VST2 and AU in OS X and presumably any other standard on Linux (we didn't test Linux).

So, we let it scan its own VstPlugIns directory where we copied only a small amount of plug-ins we wanted to use (FabFilter, IK Multimedia, iZotope, Nomad Factory, FabFilter, Melda, etc.). It also can work with 32-bit plug-ins, despite being a 64-bit application, although we did not use that option. (It would only be necessary if there was really a 32-bit only plug-in we couldn't live without.)

The GUI is very modern and elegant, and has some interesting features. The analysis tools are good. We can have both spectrum and waveform displays at once, and we have a dedicated Fast Fourier Transform (FFT) analysis. There is also a statistics window, and an extensive metadata editor, which can even include the artwork for the file (something we rarely see in an audio editor).

The included DSP processing is not bad at all, considering it is a free application. We have two graphic EQs, compressor, expander, limiter and noise gate, delay, flanger, chorus, reverb, vibrato, several filters, and two noise reduction algorithms. The only con is the fact that all of them use a generic GUI with just a bunch of sliders (common to included processors in other audio editors).

There are no chains, no mixer, and no automation. This is

a plain and simple single file audio editor. Therefore, no batch processing either. But we can import audio from video.

This is a simple and direct audio editor which, thanks to its good support of plug-ins and extensive metadata editing, can do a great job at managing and preparing a music library out of recordings of LPs or ripped CDs. It even can perform some noise reduction.

Ocenaudio: www.ocenaudio.com
Available for Windows (Windows 7 x64 and later); Mac OS X (10.10 and later); GNU/Linux (several builds available)
There are versions available for older OS versions, including 32-bit.
Free

Internet Co. Sound It! 8 Pro

This one is a newcomer to the market that we saw advertised when we were already writing the roundup. After getting a license, we went through the process of installing and reviewing the software. It was (and still is) another Work-In-Progress, which in the meantime has received a Mac OS version and several correction updates. We started with betas for Mac OS and went through the release version. The current version already seems stable. This audio editor places itself apart because it is the only one that directly supports DSD audio. We were frankly surprised, because none of the other editors focused on this particular aspect, maybe because it implies dealing with strange formats and very high-resolution audio data, and DSD may be considered a failure nowadays.

Currently, the software supports Windows from 7 to 10, with both 32-bit and 64-bit versions. Mac OS X is supported from Mavericks (10.9) onward up to High Sierra. Regarding supported formats, audio resolutions, and bit-rates, that's where this program stands apart from anything else. It supports a really extensive list of formats, including the usual linear Mac and Windows (WAV and AIFF, with a recent update extending the support to Wave 64), as well as the less popular NeXT/Sun AU, the most popular compressed formats (MP3, AAC, WMA), the most popular lossless audio formats too (Apple Lossless M4A, FLAC, Ogg Vorbis) PCM RAW data, audio from video (WMA, WMV, MOV, MP4, M4V) and even MIDI files (files are converted to audio playing internally with the Roland GS virtual sound modules, in a way similar to Audacity).

Besides these, the program also supports the more esoteric DXD formats and high-resolution PCM audio, with a maximum of 768 kHz at 32-bit. It is also 100% compatible with DSD Native Recording and Playback, without the need to convert to PCM. Supported DSD formats are DSD64 (2.82 MHz/3.07 MHz), DSD128 (5.64 MHz/6.14 MHz), and DSD256 (11.28 MHz/12.28 MHz).

But the program capabilities are not just DSD audio. It is a very competent and well-equipped application for such tasks as audio mastering, editing, and restoration. For a start, it supports all the major audio plug-in formats (VST2 and VST3 in Windows, and these two plus Audio Units in Mac OS X). However, that's not what makes it especially tailored for the tasks mentioned above. The fact is that the program not only supports these plug-in formats as it also carries a good pack of plug-ins, not just their own (which are very good) but also the renowned (although a little old already) Sonnox plug-ins.

For Restoration, we have the complete Sonnox Restoration Suite, which includes De-Noiser, De-Clicker, and De-Buzzer, and for audio editing and mastering, we have also the Sonnox Reverb, EQ, and Limiter. We also have the ability to manually correct small artifacts by zooming at max and using a pencil tool. On top of this, Internet Co. included a collection of its own (excellent) plug-ins (more than 40) among them things like Linear EQ and Linear Multiband Compressor, IR (Impulse Response) Reverb, Loudness Meter, Advance Pitch Shifter (which includes formant correction), and a lot more, among them the usual EQs, compressors, modulators, distorters, and so forth. Since we have more than 40, you can imagine how broad the choice is. However, with so many processors, it's a pity that a program like this doesn't have a suite of analysis tools. Besides the mentioned Loudness Meter, and the obligatory Level Meters, of which we have two (one in the transport panel and another in the mixer channel strip), we only have a Spectrum Analyzer. Frankly, we were expecting more in a program of this quality. Things such as a correlation meter, goniometer, spectrogram, etc., could be part of the pack. We have what we consider the bare minimum, but this program deserves more.

Being focused in high-resolution audio, as well as mastering and restoration, this audio editor doesn't have any support for loops either, or any kind of loop tool. It can play an audio in loop, or a region in loop, if there is a region selected, but it doesn't even read the loop information embedded in the files, nor does it save any kind of loop information. However, it can save an audio file in ACID loop format.

Sound It! 8 Pro has some other special features (besides DSD support). The display clearly shows the bit-rate and audio resolution of the file (we don't know any other audio editor that shows that in the control panel). Processing may be done in real time, using the master channel strip, in the way of Wavelab, or destructively, as is usual in audio editors. The most recent update even enables users to save a complete mixer configuration (plug-in stack) for later use. We also have a scrub playback function, a Performance Monitor (not that we were expecting to exhaust the capacity of our computer with a stereo file being processed, but with high-resolution DSD audio, this may be handy), and a Freeze option, in case we need to save processing power.

Recording is very straightforward, and we can record dry or through effects. The program also has a playlist window, with which we can assemble a complete CD out of separated tracks, export the result audio (complete CD), create and play a DDP file, or burn a CD directly (another example of how straightforward this program is). It also supports CD-Text editing. If the individual files have been tagged (we can do that inside Sound It!), the meaningful information, like title track and author is also displayed in the playlist.

Besides playlists, the program also supports markers extensively. Sound It! PRO can even create markers automatically based on intervals of silence, which may be good for automatically separate tracks in a vinyl recording, for example. Marker window also allows the creation of Enhanced Podcast Data, a popular format for podcasts, which includes titles, pictures, and web links in AAC format.

DSP processing is very limited. We have Gain, Fade In and Fade Out, Reverse and Insert Silence, Remove DC Offset, Reverse Phase, Exchange Left and Right, and Change Audio Format (which can go as low as 8-bit and 4 kHz, or as high as 32-bit and 768 kHz). Since the program comes with so many plug-ins, the list of built-in DSP capabilities is trimmed to the minimum for the focused tasks. We also have a very streamlined Batch Processor.

Finally, the program has a good support for volume and pan automation, almost like what we can find in a DAW. Internet Co. also launched right before we went to print a lighter version of Sound It! 8, called Sound it! 8 Basic, and priced at $69.00

Internet Co. Sound It! 8 Pro: www.ssw.co.jp/en/products/sit
Windows 7 and later / Mac OS X 10.0 and later - 32-bit and 64-bit
Cost: $199.00

About the Author

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

Why Audio Editors Are Not DAWs

DAW is an acronym for Digital Audio Workstation. Personally, we never liked it. As a matter of fact, the so-called DAWs are more "Total Music" Workstations, as they comprise the composition (creation) part, the recording part, the mixing part, and the production, processing and post-production stages. Actually, they may even be used to create the final master. They include, at least, a full featured mixer and a track editor, complemented, most of the time, with a score editor, several MIDI and "audio editors", MIDI processors, audio processors and virtual instruments. Calling such all-round and varied applications as these a "digital audio" workstation is reductionist, to say the least. But the term is now established, so...

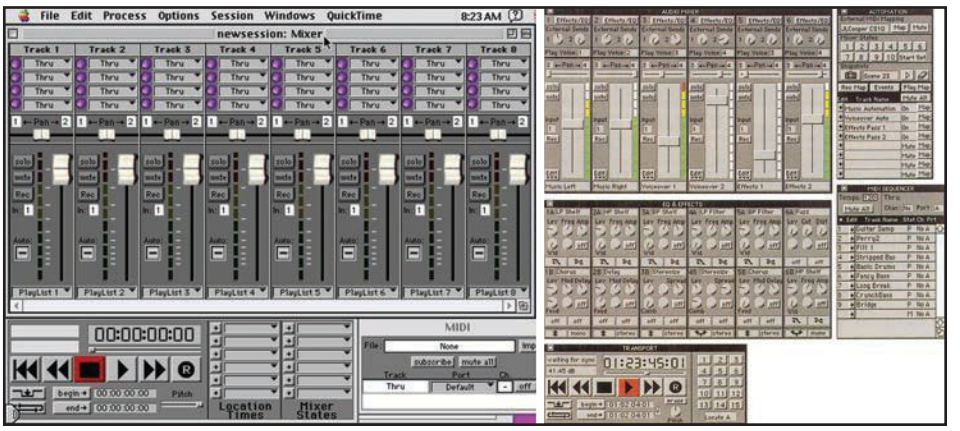
On the other hand, audio editors are specialized applications focused only on one aspect of the many quoted above—digital audio processing and mangling. These are the truly "digital audio workstations", but since the acronym is applied to the likes of Cubase Pro, Logic Pro X, SONAR, Digital Performer, and others, we must separate the subjects.

DAW History

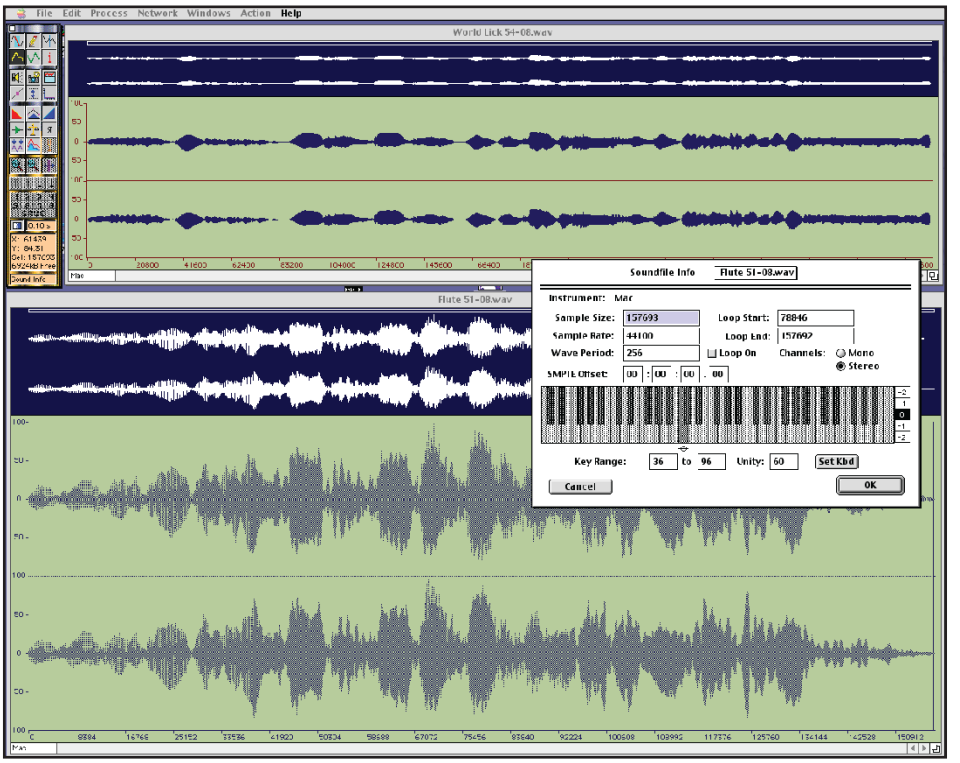
Let's revisit history to get some context. In the beginning of the 1980s, with the appearance of MIDI, an entirely new world opened to musicians. Just a couple of years after MIDI launching, the market was already filled with several applications that used computers to store MIDI data and play it back sequentially, re-creating performances and allowing an alternative method of recording music, no longer tied to magnetic tape. These applications were called "sequencers," maybe because MIDI messages were stored sequentially, according to a time map, and played back in the exact same sequence.

Soon enough this MIDI data was also translated into score data, and the two realities coexisted in some applications. Still we had no audio involved, and the audio was produced outside of the computer, by connected MIDI devices.

Almost at the same time, a company started to work with real audio, in this case for E-Mu devices. The company was founded by Peter Gotcher and Evan Brooks, and one of the first products (after some EPROMs for a drum machine) was a software called Sound Designer and an audio editor initially for the Emulator. Long story short, Sound Designer grew up, included support for other samplers, and later a new system called Sound Tools was launched, integrating



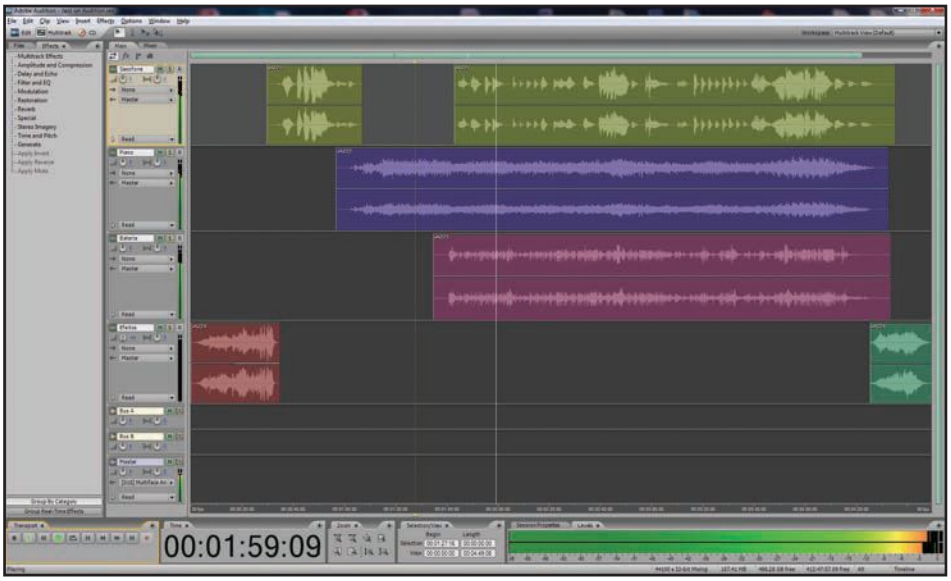
The first version of Pro Tools was a mixed bag, composed of two programs that run coupled—Pro Deck and Pro Edit. Pro Deck was based on OSC's Deck which had been launched earlier. In this picture, we can see the first Pro Tools (in this case the Pro Edit part) side by side with a later version of Deck. Even if the Deck version pictured is a later one, the resemblance is quite clear. Later versions of Pro Tools would "break" this connection. The "divorce", by the way, was anything but peaceful.



Created by a team that included Mats Myrberg, Josh Rosen, and John Dalton, who later would join to create Deck, Alchemy was the bigger competitor of Digidesign's Sound Designer. It was so good that it basically established a model that would be followed by many of the most successful audio editors that came after it (e.g., BIAS Peak and Sonic Foundry Sound Forge). It ruled as perhaps the best audio editor available for several years, but would not resist the fall of its latest owners—Passport Designs.

hardware and software, based in Sound Designer II.

Yet, and despite the hardware integration, which supplied better converters (Mac computers back then only supported 8-bit audio) and allowed for more sophisticated DSP functions, Sound Designer II was still a pure audio editor. This meant that its main goal was the editing, processing, and mangling of audio files. The biggest difference was that, by working directly on the hard drive, instead of the previous method of loading the file entirely into RAM, this system allowed for processing much bigger audio files.



Adobe Audition (formerly Cool Edit Pro) is a digital audio workstation (DAW) software from Adobe Systems featuring both a multitrack, non-destructive mix/edit environment, and a destructive-approach waveform editing view.

Harrison Mixbus took the mixer part of the DAW paradigm to the next level. It even has a version that, according to them, emulates the celebrated Harrison 32C console. Unfortunately, the MIDI part of the program is not up to the same standards, and even the audio editing part suffers some problems. But mixing with the Mixbus is perhaps the closest experience we may have to a real mixer in the virtual world.

Taking advantage of this feature (being able to record and play directly to and from the disk), a company called OSC ("(Mats Myrberg, Josh Rosen, and John Dalton, which were themselves previously involved in the creation of Alchemy, the biggest Sound Designer competitor) created a four-track recording software inspired by the four-track cassette tape-based devices, and called it Deck. The main interface was a kind of mixer, reminiscent of those devices.

Digidesign saw the potential of this approach, and partnered with OSC to launch a new product, called Pro Tools. Pro Tools was no longer oriented to digital audio manipulation, but rather recording and mixing. Based on new hardware, with more and more powerful DSP chips, "the software was a kind of Frankenstein, composed of two "collaborating" programs—Pro Deck and Pro Edit. Pro Deck was an expanded version of OSC's own Deck, while Pro Edit was basically a four-track version of Sound Designer II. The foundation for the future DAWs was launched. But it wasn't until Pro Tools III, which extended the track number to 16, and unified the software under a unique program, that Pro Tools was really established as a viable and convincing platform. The program evolved continuously, and the system became an industry *de facto* standard, dominating the professional recording studio scene up to today.

Still, there was work to be done. Computer hardware at the time was limited, and a professional musician would not be satisfied with only four tracks. Besides, MIDI was cool, MIDI devices were all over, and Pro Tools had no MIDI support at the time.

That's where a new company entered. Opcode approached Digidesign to reach an agreement to use Pro Tools hardware to create a product capable of mixing MIDI and digital audio. Approximately one year later that product appeared, and with it a true fusion between the two worlds became real. The product was Studio Vision, and besides all the MIDI capabilities that placed Vision as one of the main contenders in MIDI sequencing, it included all the audio recording and mixing capabilities allowed by Digidesign hardware, everything tied and unified under a single GUI. It was

the first time MIDI and audio were running side by side in a single application.

Similar products followed, by the main competitors—Cubase Audio, and Digital Performer. They were called, by then, digital audio sequencers (MIDI sequencing plus digital audio), but the name Digital Audio Workstation became the name used for all of these. Sure, back then not all DAWs had MIDI.

The Key Differences

So, what is the difference? As we said, these applications are much more than an audio editor. And the paradigm they follow is the recording studio. That's why we have, in the main GUI, the most important components of the studio: the mixer and the recording device. The mixer is, with more or less graphic realism, an emulation of the mixing consoles we have in studios, with sends, inserts, busses, several outs, studio monitoring, etc. The most complex routings imaginable are usually possible in these virtual mixers.

The track editor is the recording device, and in the computer we are able to "see" the tracks recorded, and edit them in detail. Some DAWs even follow an object approach, where each audio or MIDI region is an object, and can be edited individually, besides the entire track being edited globally and processed by the mixer.

Besides these, DAWs also have editors for the track content, which consist, usually, in a basic editor for the audio, and several MIDI editors (piano roll being the main one, but also a list or event editor, a score editor and some others are usually available).

The mixer also allows to insert audio processors (or "plug-ins") in the tracks or busses, and these also have their own editors.

Audio editors can be similar to the audio editors used in the DAWs to edit the audio content in each track or region. In fact, some DAWs have full featured audio editors, which rival, in terms of features and functionalities, the stand-alone ones. Anyway, as a rule of thumb, usually the editing possibilities of the editors inside the DAWs are more limited, since most editing operations are done in a track-based approach (track editor), or in a mixing approach (mixer editor).

Audio editors, on the other hand, are mainly used to process and mangle single audio files (of course, nowadays many can also play and edit multitrack audio files). The projects may be about mastering final mixes (the resulting stereo or surround mix of a complete project done in a DAW), restoring audio, forensic audio analysis, or simply mangling and editing samples for creating data to be used later in samplers like Kontakt, HALion, or Falcon.

Still, there is a broad grey area, which grows constantly (audio editors sometimes have mixers, and can playback multitrack audio files, and we know some professionals who perform mastering inside REAPER, for example—and REAPER is clearly a DAW according to our concept).

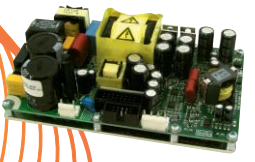
Some audio editors even have features that place them in the DAW area (we mentioned that about Soundop, for example, and DSP Quattro may fall under that too). Even Adobe Audition, which is considered a DAW, is used by many people as an audio editor, and as a DAW has some limitations. Anyway, we had to define some criteria, and, for the best and for the worst, what we described in this round-up was our criteria.

hypex NCxxxMP series

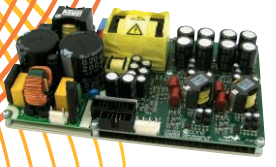
overview

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

Highlights

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

Features

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

Applications

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

NC502MP

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

ncore audio power conversion

hypex electronics www.hypex.nl

Practical Test & Measurement

Smarter Measurements for Smart Speakers

In this article, we describe techniques to characterize the frequency response, output level, and distortion of the device under test to enable direct comparisons between Internet of Things (IoT) smart speakers and conventional speakers.

By
Daniel Knighten and Glenn Hess
(Listen, Inc.) and (Indy Acoustic Research)

Since the launch of the Amazon Echo in 2015, the adoption of smart speakers has been perhaps the fastest in the history of consumer electronics, with shipments predicted to exceed 100 million units by 2024. While some manufacturers, such as Amazon and Apple, produce devices to work exclusively with their infrastructure, others such as Sonos are building smart speakers that will work with any voice control system. Although smart speakers have only been around for a relatively short period of time, players such as Apple, Sonos, and Harman are now active in this marketplace. Audio quality is a strong decision-making factor for customers and a key selling point for manufacturers, with each manufacturer incorporating its own unique speaker design and positioning for optimum sound quality. Design and improvement of far-field microphone response, which affects the speaker's ability to understand in a noisy environment, is another area where manufacturers are seeking to differentiate themselves with audio design advances.

With both sound and listening quality at the forefront of designers' minds, robust audio testing of both the speaker and the microphone array is essential. This presents new challenges in performing objective measurements as there is typically no direct path to inject or extract response signals from the embedded speakers and microphones. Rather than playing back and recording signals to local storage media, they store and playback signals from Internet-based cloud services. Furthermore, the use of voice recognition for control makes activating these devices challenging. These characteristics necessitate various additional steps to be taken in the testing process to ensure accurate and meaningful measurements.

Here, we discuss how to provide an objective evaluation of a smart speaker's audio performance by describing techniques to characterize the frequency response, output level, and distortion of the device under test (DUT) to enable direct comparison between Internet of Things (IoT) smart speakers and conventional speakers. We also discuss the selection of test signals, and the challenges around injecting and extracting signals from the device. As an example, we specifically demonstrate how to measure basic speaker and microphone audio parameters of an Amazon Echo Dot. The techniques described are applicable to any smart speaker or IoT device with an audio component.

Challenges

When measuring conventional speakers and microphones, the DUT is usually directly stimulated with a sinusoidal test signal swept across a frequency band. Measurements, such as frequency

response and distortion, are then derived from the response.

The fundamental challenges in testing the speakers and microphones in a smart speaker are:

- There is no direct output path for the microphone(s) or input path for the speaker. The devices record and playback from the "cloud," an Internet-based service of some kind (e.g., Amazon Music).
- Smart speakers are typically voice activated and have minimal or no manual controls to facilitate testing.
- Smart speakers are non-synchronous audio devices, therefore subject to sampling rate error or mismatch between the device and the measurement system.

Retrieving the Response Signal

In the case of Amazon Alexa-based devices, while we are unable to directly extract the microphone signal, a recording of everything captured by the device is available from a website. Trial and error was used to discover that Alexa devices record for about 8 seconds after the activation word is detected. A test signal was constructed that swept over the frequency range of interest within this window. For other brands of speakers, a method of accessing the recorded signal must be discovered, and the time for which it records ascertained to ensure that the test signal is fully included.

During the test, a composite signal consisting of a voice recording of the activation word followed by a stepped sine sweep is used. After capture of the recording by the device, the audio recording is downloaded from the associated website. This captured recording can then be analyzed to determine the microphone's characteristics.

Storing and Playing Back the Stimulus

To test the speaker function, a test signal can be uploaded to the account with the associated Internet music service as if it was a musical recording. The DUT can then be commanded to playback the test signal. From there, the playback of the stored test signal is captured by the measurement system. Specifically in the case of Amazon Music, the stored audio signal had to be encoded in the MP3 format. We verified that the MP3 encoding itself was not degrading the test signal by encoding and decoding the test signal, and using the transfer function analysis between the original signal and the encoded and then the decoded version.

Activating Smart Speakers

In general, smart speakers are always listening with their internal microphone(s) active. They respond to an activation word or wake-up phrase followed by a command string. To test the audio performance of such a speaker, the device must first be activated, and then a suitable test signal applied to keep the smart speaker active and operating in a normal state (see **Figure 1**).

Each smart speaker platform has its own activation phrase, and trial and error is needed to establish how long the device records, and then to get it to playback a specific signal. Once these parameters are established, the device can be made to either play back the desired stimulus for speaker testing or record a signal for microphone testing. When the response is analyzed, it is parsed so that the activation phrase is ignored.

Overcoming Sampling Rate Error

Traditional measurement systems rely on the

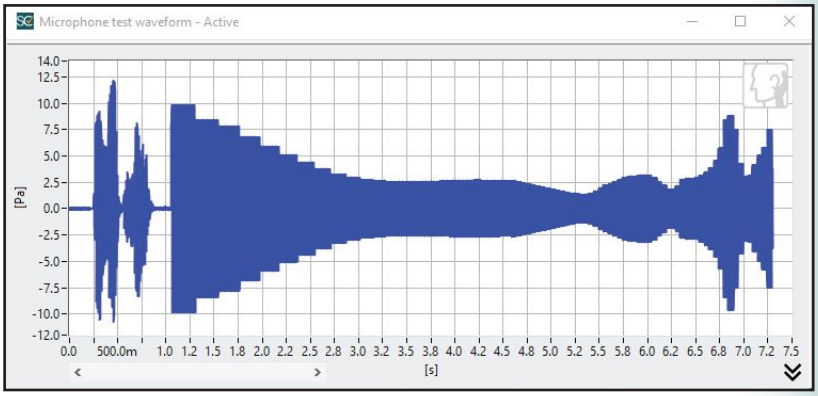


Figure 1: Activation word and test signal waveform for microphone measurement

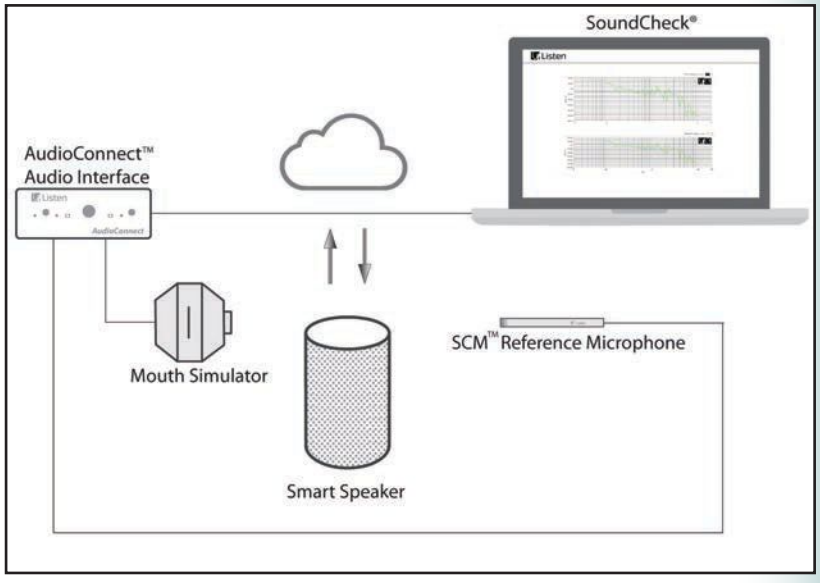


Figure 2: Measurement setup for testing the loudspeaker

DUT having a synchronous input and output. As a category, smart speakers are intrinsically open loop, which means that there is no synchronous input and output. Instead as described above, the device can record a signal or playback a signal but it does not have a synchronous signal path.

This introduces the possibility for sampling rate error. That is, the device may record a signal to a file with a sample rate of 44.1 kHz, but it may have been recorded at 44.09 kHz or another similar rate due to skew in the actual rate of the clock crystal used to drive the sampler. A similar error can occur when playing back test signals. That is, the test stimulus may be sampled at 44.1 kHz, but due to error in the playback sample rate the file is actually played back at a slightly faster or slower rate.

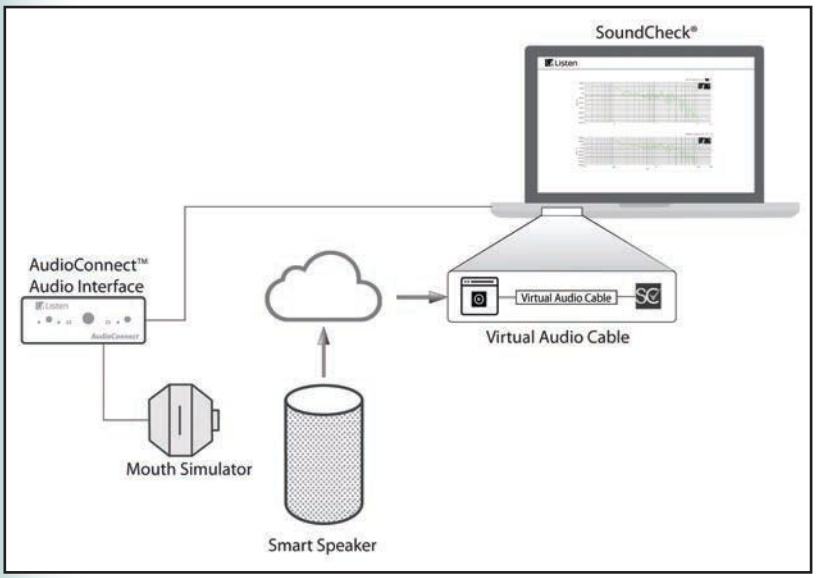


Figure 3: Measurement setup for testing the microphone array

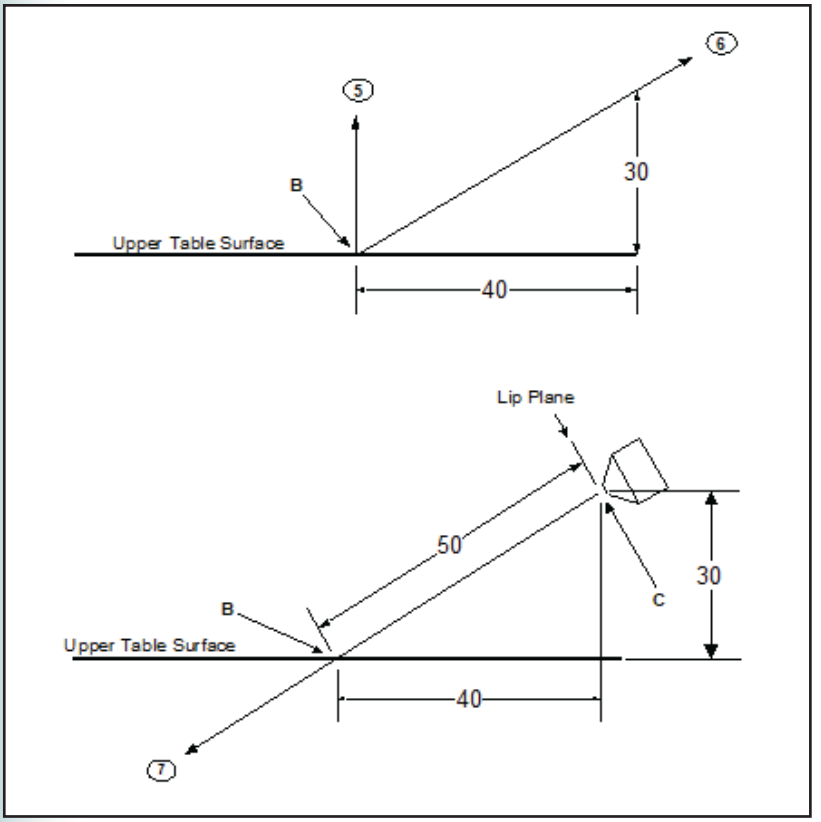


Figure 4: The IEEE 1329-2010 test setup with table (dimensions shown in cm)

This sampling rate error will result in the component tones of the test stimulus being shifted to either a higher or lower frequency. This shift can then lead to measurement errors due to loss of coherence between the stimulus and the response signals.

To overcome this sampling rate error, an algorithm is applied, which searches the beginning of the response waveform for a steady-state sinusoid at a preset frequency. The signal is then shifted to DC using a heterodyne filter and all other frequencies are filtered out. The output of the heterodyne filter includes the phase information, which is ultimately used to estimate actual playback or recording sample rate of the response signal. With this information, the entire response waveform is resampled to the correct stimulus sample rate prior to analysis. This frequency shift step corrects for sampling rate error in the device and makes testing them straightforward.

In our example test, the actual signal used to measure the frequency response and the distortion

Resources

K. Brandenburg, "MP3 and AAC Explained," presented at Audio Engineering Society (AES) 17th International Conference on High-Quality Audio Coding, August 1999.

G. Hess and D. Knighten, "The Challenges of IoT Smart Speaker Testing," presented at the Audio Engineering Society (AES) 143rd Convention, October 2017.

IEC 61094-4, "Measurement Microphones Part 4: Specifications for Working Standard Microphones," November 1995.

IEEE 1329-2010, "Standard Method for Measuring Transmission Performance of Speakerphones," October 2010.

ITU-T Recommendation P.51, "Artificial Mouth," August 1996.

S. Temme, et al., "The Challenges of MP3 Player Testing," presented at the Audio Engineering Society (AES) 122nd Convention, May 2007.

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of the device was a stepped sine sweep. For the microphone measurements, the amplitude at each frequency step of the sweep was equalized to produce an equal pressure response of 89 dB SPL (-5 dBPa) at the mouth reference point of a mouth simulator, per IEEE standard 1329-2010. The sweep provided 1/12 octave resolution from 100 Hz to 10 kHz. For the speaker test, the stimulus signal had an amplitude of –13 dBFS. A 1/3 octave resolution sweep was used from 20 to 100 Hz and 1/12 octave resolution from 100 Hz to 20 kHz.

Analysis was performed using a heterodyne filter, which extracted the fundamental and harmonic distortion products from the response signal.

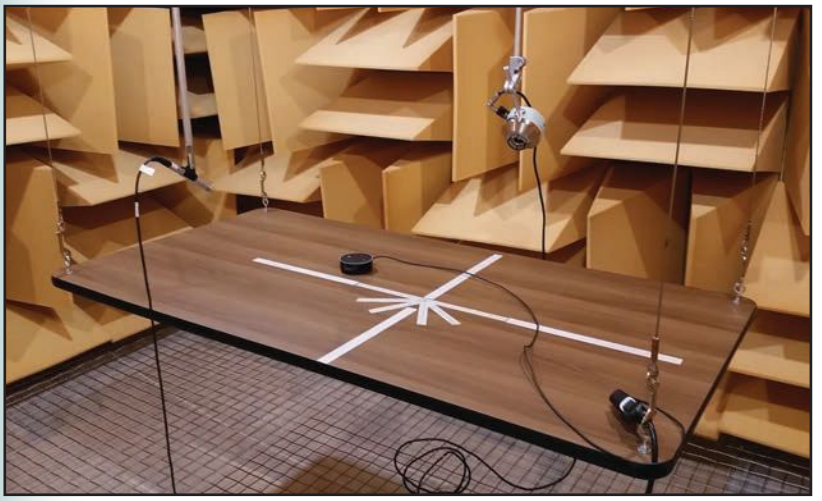

Photo 1: We followed the IEEE 1329-2010 test setup and conducted the test in a full anechoic chamber).

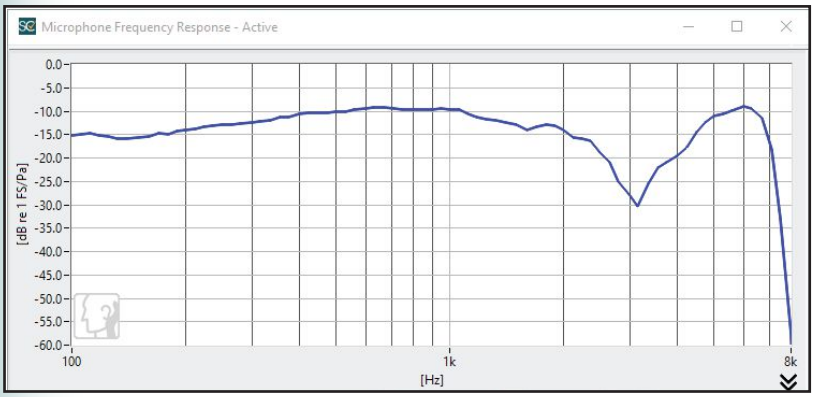

Figure 5: Microphone frequency response

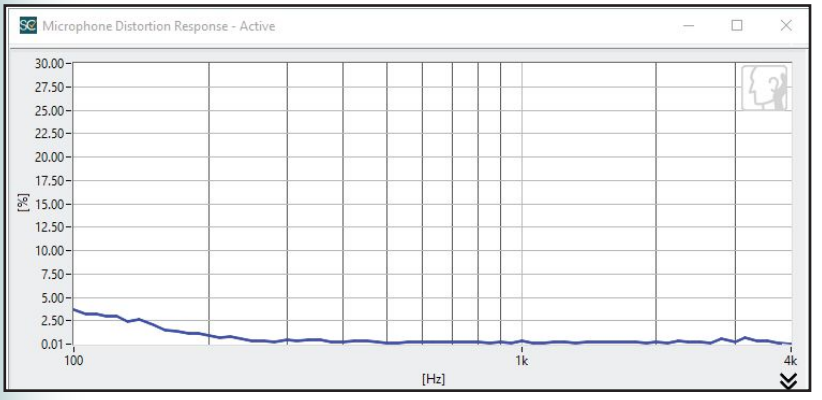

Figure 6: The microphone distortion response

Measurement Equipment

The measurement setup is similar to a regular loudspeaker measurement. At the heart of the setup is Listen, Inc.'s SoundCheck test system. This generates and plays all test signals, as well as acquiring and analyzing the response. An AudioConnect audio interface plays the signal from the computer to the mouth simulator, and similarly from the microphone to the computer (see **Figure 2**). The artificial mouth is required to generate the activation command and to play the test signal for a microphone measurement (see **Figure 3**).

Physical Setup

At this point there are no test standards specifically written for smart speakers. However, their practical use case closely resembles speakerphones, so for these measurements we chose to test using the physical geometry recommended by IEEE 1329-2010 (see **Figure 4**). All our tests were conducted inside an anechoic chamber with the DUT placed on a table (see **Photo 1**).

Device Microphone Measurements

Microphone system performance includes measurements of frequency response, sensitivity, and distortion. The microphone measurements were made by:

- Applying the test signal using a calibrated mouth simulator
- Retrieving of the voice interaction recording from the associated website and transferring it to the software system
- Analyzing the recording with the SoundCheck software

The microphone frequency response is relatively flat except for the dip at 3 kHz (**see Figure 5**). This is due to the device's microphone proximity to the table surface causing a "table bounce" effect. The sensitivity is measured in decibels F_S/Pa, where F_S is relative to digital full scale in the waveform recorded by the device,

and relative to the sound pressure in Pascals at the mouth reference point.

The total harmonic distortion is relatively low above 200 Hz, but increases at lower frequencies due to the residual distortion of the mouth simulator (see **Figure 6**).

Device Speaker Measurements

Speaker system performance includes measurements of frequency response, sensitivity, and distortion. The speaker measurements were made by:

- Uploading test signal to the associated online music library in MP3 format
- Dictating the activation phrase to play test signal (test operator or artificial mouth)
- Capturing the response signal with a measurement microphone
- Analyzing the captured signal with the SoundCheck software

The frequency response is relatively flat in the passband from 400 Hz to 6 kHz (see **Figure 7**). The 400 Hz roll-off is due to physical limitations of the device's loudspeaker. The sensitivity is measured in decibels re 20 µPa/FS, where output is sound pressure level (SPL) at the reference mic and input is relative to digital full scale in the signal played back by the device (see **Figure 8**).

Total harmonic distortion (THD) is relatively low above 500 Hz, but increases at lower frequencies due to limitations of device's loudspeaker.

Conclusions

Smart speakers present three main testing challenges. The practical challenge of playing back test signals and recording response signals, controlling the DUT, and sampling rate mismatch between the DUT and the test system. However, as we have demonstrated, all these challenges can be overcome and conventional results can be extracted.

A continuing challenge is that every smart speaker is different. Unlike conventional loudspeakers where an identical test can be used on a selection of speakers, a test will likely need to be tailored for each speaker due to differences in activation words, recording time, method of extracting the recording, and so forth.

There is also much additional work to be done exploring tests involving non-stationary signals, including noise, modulated noise, and actual speech and music. In addition, it would be of value to devise a test to explore how well the devices can detect their activation phrases while playing back music, in the presence of noise, and with multiple interfering talkers. Finally, a test that could independently evaluate the actual accuracy of the speech recognition algorithms would be valuable.

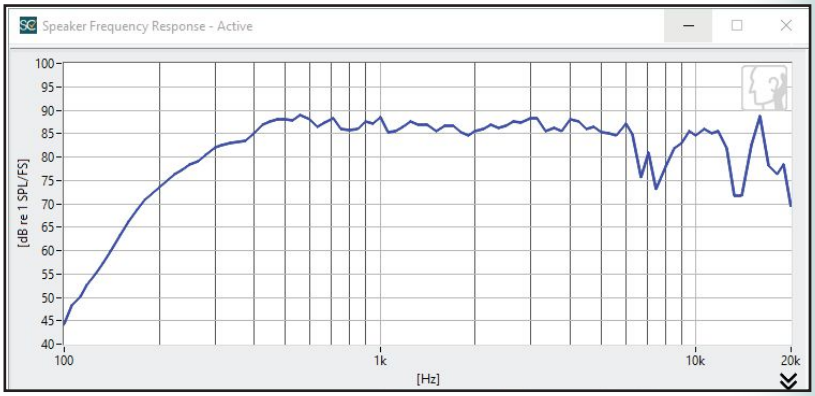

Figure 7: The speaker frequency response

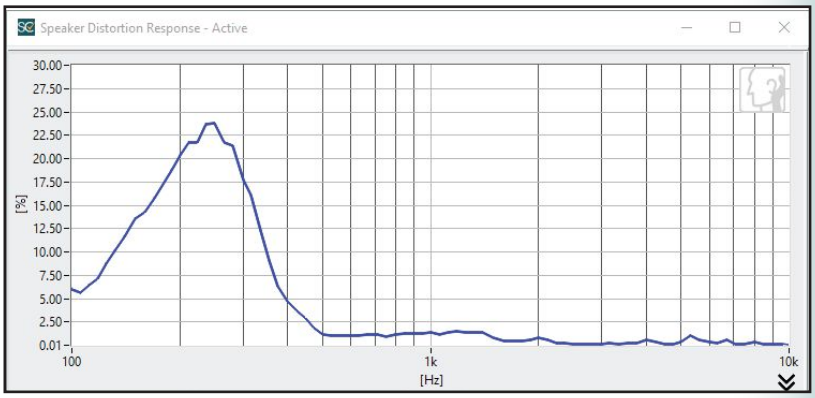

Figure 8: The speaker distortion response

About the Authors

Daniel Knighten has a wealth of experience testing all types of audio and audio electronic devices, and offers applications support to Listen Inc.'s global customer base. Prior to Listen, he held various sales, engineering and support positions at Audio Precision for more than 10 years, where he acquired a wealth of practical experience testing a wide range of electroacoustic and electronic devices. In addition to his role at Listen, he is also director and founder of Portland Tool & Die, a company that designs and manufactures audio test interfaces, which are exclusively distributed by Listen.

Glenn Hess has been involved in electro-acoustic testing of communication systems and more recently consumer audio devices for more than 30 years. His career has included engineering positions at Bell Laboratories, MWM Acoustics, and Harman International. He has served as technical editor and contributor on various Institute of Electrical and Electronics Engineers (IEEE) and Telecommunications Industry Association (TIA) industry standards for more than 25 years. And, he currently serves as vice chairman of the IEEE Working Group for Communication Electroacoustics. Glenn is a co-founder and general manager of Indy Acoustic Research (IAR) formed in April 2015. IAR provides acoustic engineering consulting for communication, consumer electronics, and specialty products.

Fresh From the Bench

G.R.A.S. Sound & Vibration RA040X Ear Simulator

Measuring headphone response at frequencies above-10 kHz has been an imprecise, non-standardized process. With the RA040X high-resolution ear simulator, G.R.A.S. Sound & Vibration attempts to make these measurements more meaningful. In this review, we determine how well it works.

By
Brent Butterworth
(United States)

Since the 2007 debut of the iPhone shifted much of the audio industry's attention to headphones, we've seen the emergence of countless headphones and earphones claiming outstanding sound quality. Yet, our ability to judge these claims has been impaired by outdated tools. The design of many devices used to measure headphone performance dates back to an era when "portable audio" meant a boombox, and consumers used headphones mainly so they could listen without bothering others.

The new RA040X ear simulator represents part of an effort by G.R.A.S. Sound & Vibration to update headphone measurement equipment to suit the needs of today's headphone industry.

Most current headphone measurement equipment relies on an ear simulator designed in accordance with the International Electrotechnical Commission (IEC) 60318-4 standard, updated in 2010 but based on the IEC 60711 standard established in 1981. In 1981, high-fidelity headphones barely existed. The focus of IEC 60711 was hearing aids, in which speech intelligibility is paramount and other concerns are much less important.

The 711 Ear Simulator

A so-called "711" ear simulator has resonant chambers designed to mimic the acoustical characteristics of the human ear, so the acoustical effects of the space between the headphone or earphone under test and the microphone diaphragm inside the coupler are comparable with those between the headphone and the human eardrum.

However, these ear simulators have a high-Q nuisance resonance centered at roughly 13.5 kHz (see **Figure 1**), which corresponds with the length of the main cavity inside the ear simulator (and which does not occur in the human ear canal due to its irregularities in shape).

With hearing aids, speech intelligibility is the main concern and an artifact at 13.5 kHz is of little importance as the human voice has insignificant harmonic content in this range. Thus, the performance of 711 ear simulators is specified only up to 10 kHz. According to the abstract for IEC 60381-4: "Due to resonances in the acoustic transfer impedance of the occluded-ear simulator above 10 kHz, high measurement uncertainties, e.g., in the order of 10 dB, can occur in earphone responses."

Because the resonance of the ear simulator chamber is determined in part by the distance from the microphone diaphragm to the headphone under test, the frequency of the resonance drops when the headphone under test is further away from the mic diaphragm. It can also change if the ear simulator is mounted in a test fixture that has its own artificial canal. These changes make high-frequency measurements even more problematic because it's difficult to know at which frequency the resonance will occur, and thus, difficult to compensate for it.

Yet, the proliferation of headphones and earphones claiming high levels of audio fidelity has created an increasing need to accurately measure the response of headphones and earphones above 10 kHz. When we listen to music through headphones, we experience significant content up to 20 kHz—and perhaps even above 20 kHz, if we are listening to high-resolution audio formats with bandwidth that stretches beyond 40 kHz.

Fixing the High-Frequency Problem

Some engineers have responded to this need with home-grown solutions, such as mounting a standard measurement microphone in a hole drilled in a flat plate and pressing the earpiece of the headphone under test against the plate. I've tried these solutions myself, but I've encountered two problems that limited their utility. First, because these measurements are not standardized, they can be used only for relative comparisons with other headphones measured using the same jury-rigged piece of equipment. Second, they produce measurements that do not correspond with, and can't be accurately spliced with, measurements produced using a 711 coupler.

Like many of the engineers, technicians, and reviewers using 711 ear simulators, I've learned to largely ignore the results above 10 kHz. When readers occasionally contact me to express concern about, say, a 17 kHz response peak in a headphone frequency response plot I've published on the SoundStage website or the other publications for which I've reviewed headphones, I can do little but suggest they disregard the top octave of my frequency response plots—which of course raises the question of why I publish such data.

G.R.A.S. Sound & Vibration designed the RA040X ear simulator to produce more meaningful and useful measurements above 10 kHz (see **Photo 1**). Unlike the G.R.A.S. RA0045 ear simulator that many reviewers, engineers, and scientists use, the RA040X (available as the externally polarized RA0401 or pre-polarized RA0402) incorporates acoustic damping intended to reduce high-frequency resonance. G.R.A.S. claims

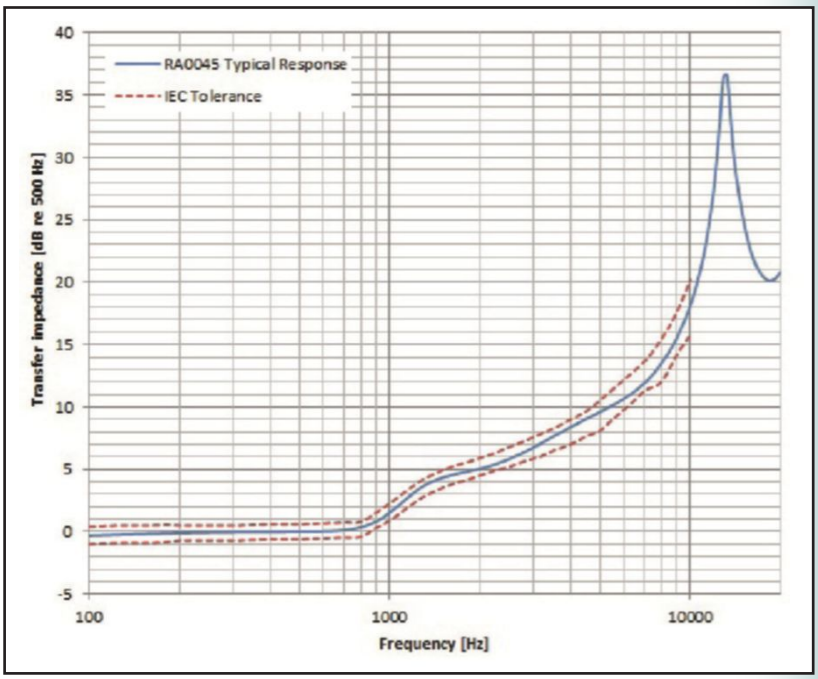

Figure 1: This graph shows the typical response of a G.R.A.S. RA0045 ear simulator (Image courtesy of G.R.A.S. Sound & Vibration)

the 13.5 kHz resonance is reduced by 14 dB without significantly affecting response below 10 kHz. The RA040X is a "drop-in" replacement for the RA0045, compatible with the 43AG ear/cheek simulator I use as well as with the 45BB KEMAR manikin and other G.R.A.S. test fixtures.

Eager to find a reliable method of measuring the high-frequency performance of headphones, I asked G.R.A.S. for an RA040X sample I could compare with my RA0045. My priority in these tests was frequency response measurements. I wanted to see if the RA040X could produce consistent, meaningful results from which I might eventually be able to draw conclusions about a headphone's performance above 10 kHz—conclusions I could justify to readers

Photo 1: G.R.A.S. Sound & Vibration designed the RA040X ear simulator to produce more meaningful and useful measurements above 10 kHz.

Photo 2: For my first tests, I connected the RA0045 and then the RA040X ear simulators directly to my Audiomatica CLIO 10 FW analyzer and measured the frequency response of an in-ear monitor.

and defend against the criticism that often appears in the comments sections of websites and social media pages.

After reading the G.R.A.S. white paper on the RA040X, I was also curious to see what effect it might have on my harmonic distortion measurements. The false 13.5 kHz response peak that shows up in measurements with the RA0045 might be interpreted by my audio analyzer as a false second harmonic of 6.75 kHz, a third harmonic of 4.5 kHz, a fourth harmonic of 3.375 kHz, etc., which would produce an inaccurate distortion measurement. Frequency shifts in the 13.5 kHz resonance would further complicate matters.

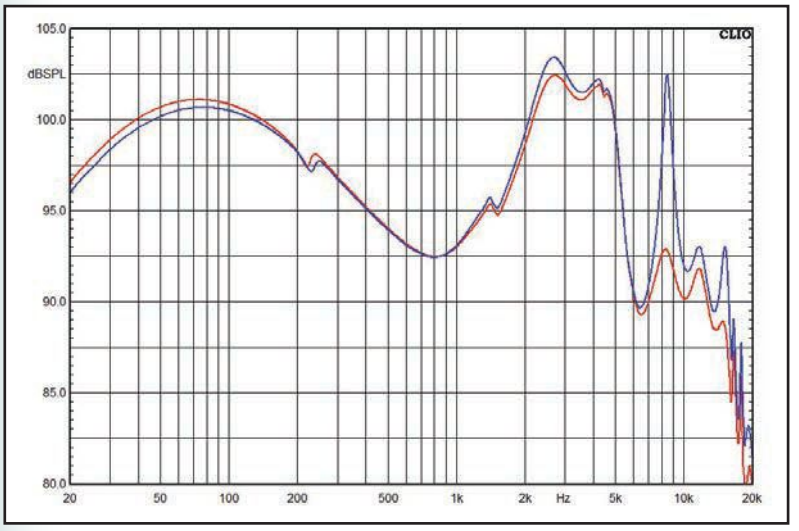

Figure 2: I measured the frequency response of the 1More Triple-Driver IEM with the G.R.A.S. RA0045 (blue trace) and the G.R.A.S. RA040X (red trace).

I considered seeing what effect the RA040X might have on my measurements of the active noise cancellation (ANC) capabilities of the headphones, but the measurement method I use employs ambient pink noise played at 85 dB SPL and starts with a reference response curve taken from the RA0045/43AG with no headphone attached, so it incorporates (and thus ignores) the RA0045's 13.5 kHz resonance. Thus, the RA040X wouldn't improve these measurements.

The Tests

Eager to see the differences between the RA0045 and the RA040X as quickly as possible, I grabbed a 1More Triple-Driver in-ear monitor (IEM) that happened to be sitting on my test bench, and measured its frequency response first with the RA0045 and then again with the RA040X, using their steel coupler tubes rather than the 43AG ear/cheek simulator. (I chose to measure an earphone first because it's easier to get consistent positioning of the earphone in the coupler than it is to get consistent positioning of a headphone on the 43AG.) I connected both ear simulators directly to my Audiomatica CLIO 10 FW analyzer (see **Photo 2**). A Musical Fidelity V-Can headphone amplifier was used to drive the headphones, and the 43AG sat on a platform isolated on an elastomeric suspension to minimize the effect of ground-borne vibration on the measurement. Except as otherwise noted, I used the CLIO's logarithmic chirp stimulus for the measurements.

The difference in response between the RA0045 and the new RA040X was easy to see (see **Figure 2**). With the 1More Triple-Driver in-ear monitor, the main high-frequency resonance of the RA0045 was apparently shifted down to 8.5 kHz. Substituting the RA040X reduced this resonance by -9.6 dB. Using the RA040X also damped weaker high-frequency resonances centered at 11.7 kHz and 15.1 kHz by -1.2 dB and -4.1 dB, respectively, and reduced overall response between 10 and 20 kHz by -1.8 dB on average.

G.R.A.S. says the response of the RA040X at midrange and low frequencies matches that of the RA0045. I noticed that the primary resonance of the units, centered at 2668 Hz, was -1.0 dB lower with the RA040X. Whether this is due to normal unit-to-unit variation or to innate response differences between the two units, I can't say without testing more samples. There was also a slight difference in the measured bass response, which is likely the result of minor positioning differences of the 1More Triple-

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Driver IEM in the ear simulator's steel couplers.

I then tried measuring a circumaural (over-ear) model, the Sony MDR-7506 professional headphone (see **Figure 3**). This measurement proved less illustrative, as the variability of placement of an over-ear headphone on an ear/cheek simulator and simulated pinna presents enough acoustical variables that precise repeatability of measurements becomes difficult. The result shown here represents the best match I was able to achieve in the midrange, from about 300 Hz to 2 kHz.

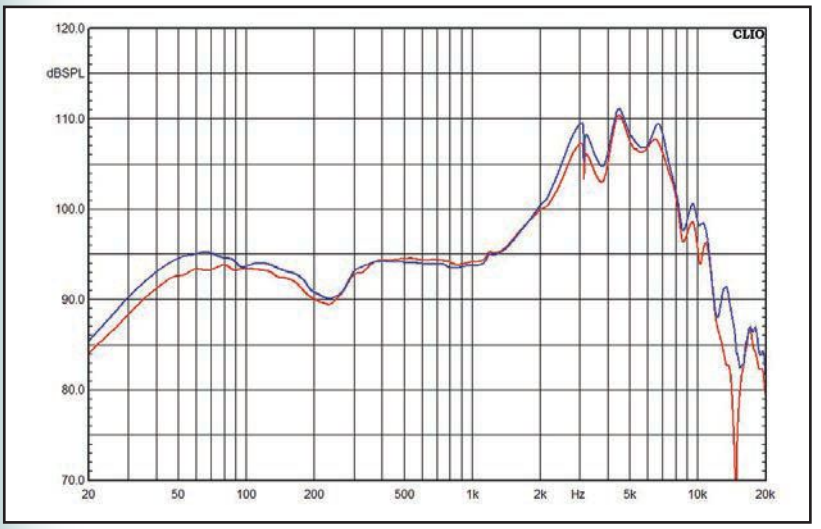

Figure 3: The frequency response of the Sony MDR-7506 was measured with the G.R.A.S. RA0045 (blue trace) and the G.R.A.S. RA040X (red trace).

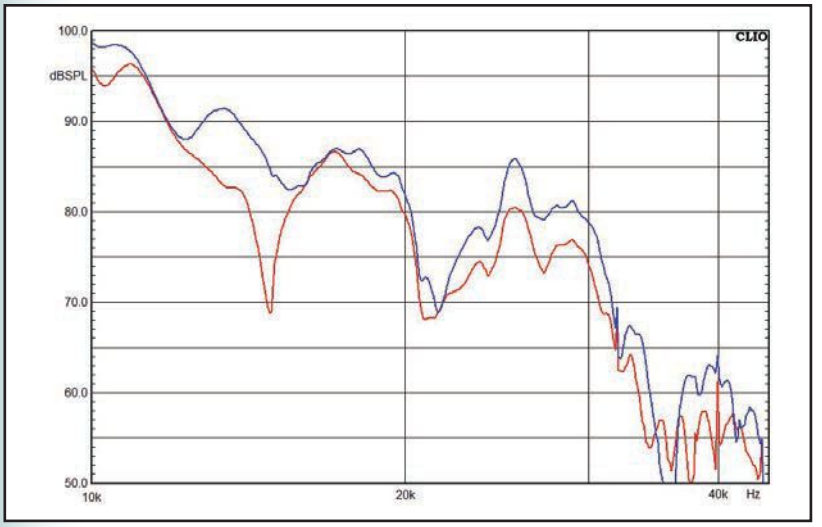

Figure 4: The frequency response of the Sony MDR-7506 from 10 kHz to 44 kHz is measured with the G.R.A.S. RA0045 (blue trace) and the G.R.A.S. RA040X (red trace).

About the Author

Brent Butterworth has been a professional audio journalist since 1989, and has measured and evaluated thousands of audio products. His headphone measurements and reviews currently appear on soundstage.com and wirecutter.com. He has served as editor-in-chief of *Home Theater* and *Home Entertainment* magazines, contributing technical editor for *Sound & Vision* magazine, senior editor of *Video* magazine, and reviews editor of *Windows Sources* magazine. He has also worked as marketing director for Dolby Laboratories, and as a consultant on the design and tuning of speakers, subwoofers, and headphones.

In this case, the characteristic 711 ear simulator resonance is evident, centered at 13.3 kHz, and damped by -7.9 dB when the RA040X is substituted for the RA0045. (The high-Q dip centered at 14.7 kHz is almost certainly an artifact created by a minor positioning variation of the headphone on the 43AG.)

However, with the circumaural headphone, the difference between the RA0045 and RA040X seems less pronounced than with the in-ear model. I assume this is because the much larger and more irregular acoustical space between the headphone's baffle and the microphone diaphragm (a space which also incorporates the KB5000 simulated pinna) reduces the effect of the ear simulator's resonance. The overall effect is a response reduction averaging about -2 dB above 2.7 kHz.

Normally I set my CLIO 10 FW for a sampling rate of 48 kHz, which with the CLIO allows a maximum measurement bandwidth of about 22 kHz. Considering the increased interest in measurement of above -20 kHz frequencies spurred by the audio industry's recent promotion of high-resolution audio downloads, I decided to switch the sampling rate to 96 kHz to see what effect the RA040X would have on such measurements.

Figure 4 shows the effect of the RA040X's damping (besides the fairly strong damping of the approximately 13.5 kHz resonance) can perhaps best be characterized as a general reduction in response by an average of -3.5 dB between 10 and 44 kHz.

Many, perhaps most, of the headphones I measure show no significant harmonic distortion even at extremely loud levels. However, the 1More Triple-Driver IEM does distort measurably, so I used it to see what effect using the RA040X would have on my distortion measurements. I decided to again use the CLIO's 96 kHz sampling rate to extend the distortion measurement bandwidth up to 20 kHz.

The level of the sine stimulus signal was set by playing pink noise from the CLIO and adjusting the level to reach 100 dBA, which corresponds to an extremely loud listening level. I use this level because most headphones can reproduce it without significant distortion, but perhaps one out of five models cannot.

Here, I expected the 8.5 kHz resonant peak from the RA0045 that showed up the 1More's original frequency response measurement to reappear as a false harmonic and produce elevated distortion measurements at 4.3 kHz, 2.8 kHz, and 2.1 kHz.

However, the expected artifacts did not appear (see **Figure 5**). And while some differences did appear in the harmonic distortion sweeps when the RA040X replaced the RA0045, they seem spurious and could easily be attributed to minor positioning differences of the headphone under test in the ear simulators' steel couplers.

Conclusions

Measurements of headphones and earphones at frequencies above 10 kHz is in its infancy, but it appears to me that the G.R.A.S. RA040X will help improve this situation. The internal damping G.R.A.S. has added clearly helps reduce the magnitude of the main chamber resonance of the ear simulator, which should produce above-10 kHz measurements that, while not perfect, are certainly more trustworthy.

I hope the emergence of better equipment for high-frequency measurement of headphones spurs new research into how variations in above-10 kHz response affect the subjective perception of headphone response. I would also like to see how well above-10 kHz measurements with ear simulators correspond to the response of human ear canals, which varies considerably at frequencies above 10 kHz due to variation in the shapes of the canals. At this point, the RA040X seems a little like the first transistor: Something likely to inspire important work in the future, but that will take some time to fully appreciate.

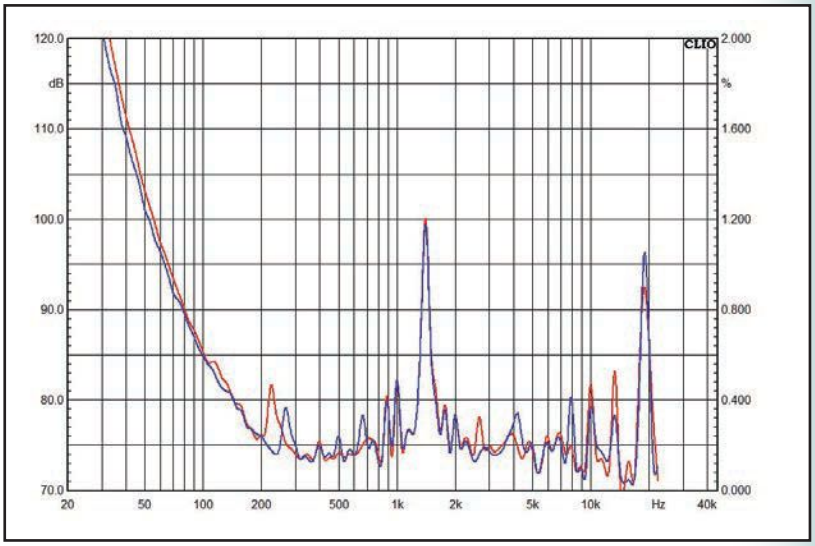

Figure 5: Total harmonic distortion (THD) vs. frequency of the 1More Triple-Driver IEM is measured with the G.R.A.S. RA0045 (blue trace) and the G.R.A.S. RA040X (red trace).

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Photo 1: The ARC 2.5 System comes with a new MEMS measurement microphone. Its response is specified as omni-directional, which makes it ideal for this application.

By
Ron Tipton
(United States)

Available since 2007, IK Multimedia's Advanced Room Correction (ARC) System is the most affordable, easy to use, and effective acoustic room correction system. Announced in June 2017, and released August 2017, IK Multimedia's ARC System 2.5 now offers a new MEMS measurement microphone (see the separate review by Stuart Yaniger in this edition). With the MEMS microphone, IK Multimedia says the ARC 2.5 improves the audio monitoring accuracy of speakers in any studio or listening room, taking it a significant step further.

ARC 2.5 is an updated version 2.0 with several new features such as 64-bit OS support and up to 192 kHz sample rate (although the User Manual still says 48 kHz). But the biggest improvement is the inclusion of a MEMS microphone (see **Photo 1**) with a flatter frequency response as compared to the 2.0 mic.

Both versions use the patented Audyssey MultiEQ XT32 room correction technology, which corrects both the frequency and time responses. Windows and Mac OSX are supported but the ARC system is 64-bit so the OS and DAW must be 64-bit. There are two plug-ins: the measurement module and the correction module. Both are standard VSTs for Windows. Of course, this means that all your music listening must be done through your computer, but the acoustic improvement is worth the effort.

Measurement Module

The measurement module's first screen is simply introductory, click NEXT. You are then asked to select a microphone, click the MEMS mic and click NEXT. The Step 2 screen (see **Photo 2**) is Audio setup. Select your audio hardware (sound card) and click NEXT. The Step 3 screen (see **Photo 3**) sets the measurement sound level. Click the TEST bar and adjust the speaker's volume control, mic preamp gain or the on-screen slider until the bar indicator is in the OK range, again click NEXT to load the Measurement screen (see **Photo 4**). A minimum of seven measurements are required around your listening position. The User Manual, included in the Supplementary Material, has some suggestions for various chairs and sofas. I used the positions shown in **Figure 1**.

Move your microphone to start position, number 1, near the center of the listening area. The vertical arrow will be under the on-screen "1" box. Click the TAKE MEASUREMENT bar. The arrow will move to box "2" so move your mic and take that measurement. Continue until you have taken at least seven, only then can you proceed to the next step to save your measurement file—give it a name and click SAVE.

After I used the seven positions shown in Figure 1, I took a set of 11: the original seven plus two equally spaced behind the chair and two in front. When I measured the correction results, there was no noticeable difference between the two. It seems seven measurements were adequate.

Correction Module

I attempted to use the free Room EQ Wizard (REW) to verify the correction but this didn't

Photo 2: In the second measurement screen, you make your audio hardware selection. I selected my ESI Juli@ sound card and the ARC software found the ASIO driver and filled in the other data. Click NEXT to continue to screen 3.

Photo 3: In this third step, you set up the measurement sound level with a series of "beeps" from your speakers. I think you get the best results (better signal-to-noise ratio) by keeping your mic preamp's gain fairly low and using a fairly loud speaker volume without over driving. Keep adjusting it until the bar graph reading is in on the OK range.

Photo 4: Screen 4 is where you make the measurements. The small vertical arrow is under "Box 1." Make sure your mic is in location one and press the TAKE MEASUREMENT bar. If the measurement is okay, the arrow will move to "Box 2." Move your mic and continue the process. When you have completed the seventh step, you will have the option to save your data or you can continue taking measurements.

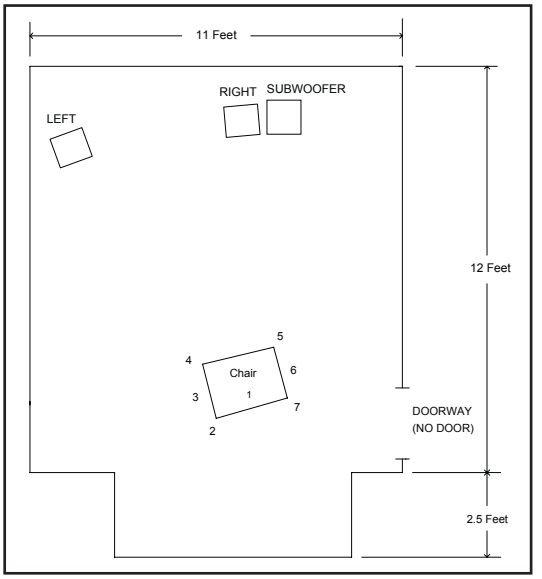

Figure 1: In this floor plan of my listening room, the numbers around the chair are the microphone locations for my first set of ARC correction measurements and the mic locations for measuring the correction as graphed in Figure 3.

succeed. Primarily, I use Reaper as my DAW and for it to work in pass-through mode, it must be installed with its optional ASIO driver. The sound card must also have an ASIO driver, which my ESI Juli@ card does, and REW must have an ASIO driver, which it does. Despite this apparent compatibility, my prior EQ measurements using REW through Reaper showed a 60 dB drop in level between 2 kHz and 22 kHz. I've measured this room many times and knew this was a gross error. I tried three different microphones, including the MEMS, different speaker volume levels with compensating changes in the mic preamp gain. All to no avail so I went to "Plan B"—white noise as diagrammed in **Figure 2**.

Using the free Audacity audio editor, I generated a 30-second segment of white noise and saved it as a 24-bit, 96 kHz wav file. Reaper

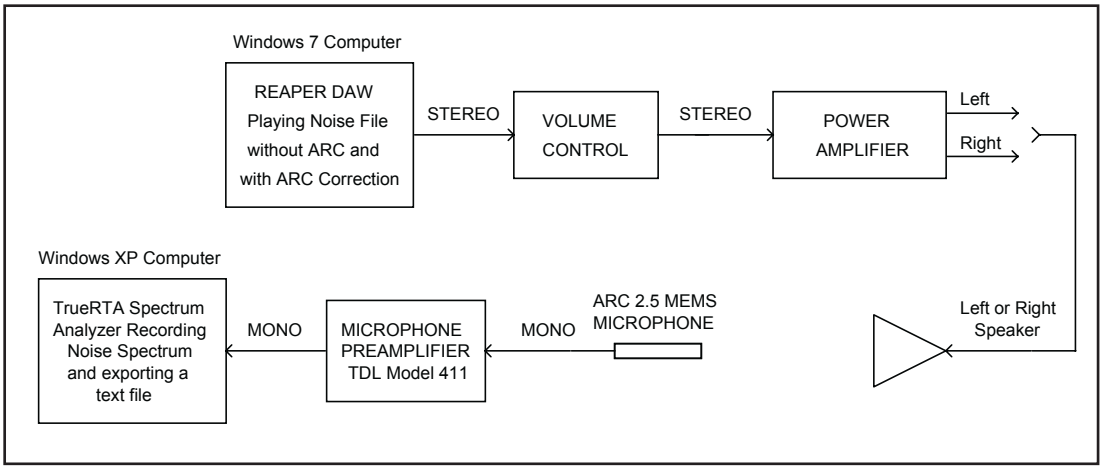

Figure 2: The block diagram details my before-and-after correction frequency response measurement setup. The Reaper DAW plays the white noise file directly into the sound card for the "before" data and through the ARC Correction plug-in for the "after" data. TrueRTA records the 1/6-octave frequency spectra and exports a text file for each.

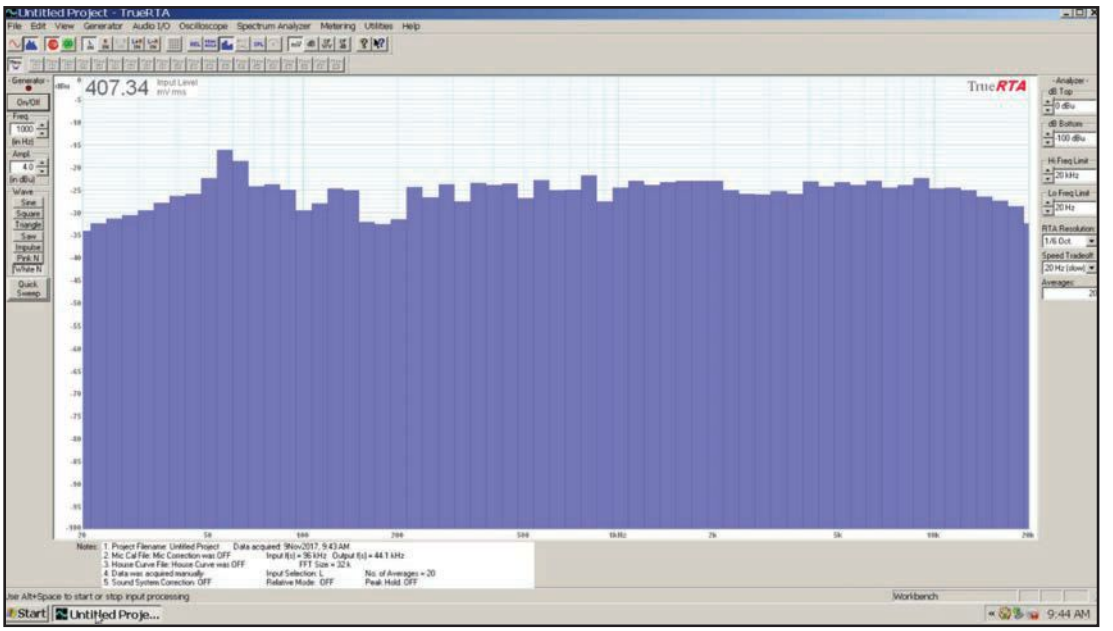

Photo 5: This TrueRTA screenshot shows a typical 1/6-octave display of a captured white noise signal as played by Reaper on the other computer. I exported each display as a text file for later averaging of the data from the seven mic locations. The two "before" and the two "after" averages are graphed in Figure 3.

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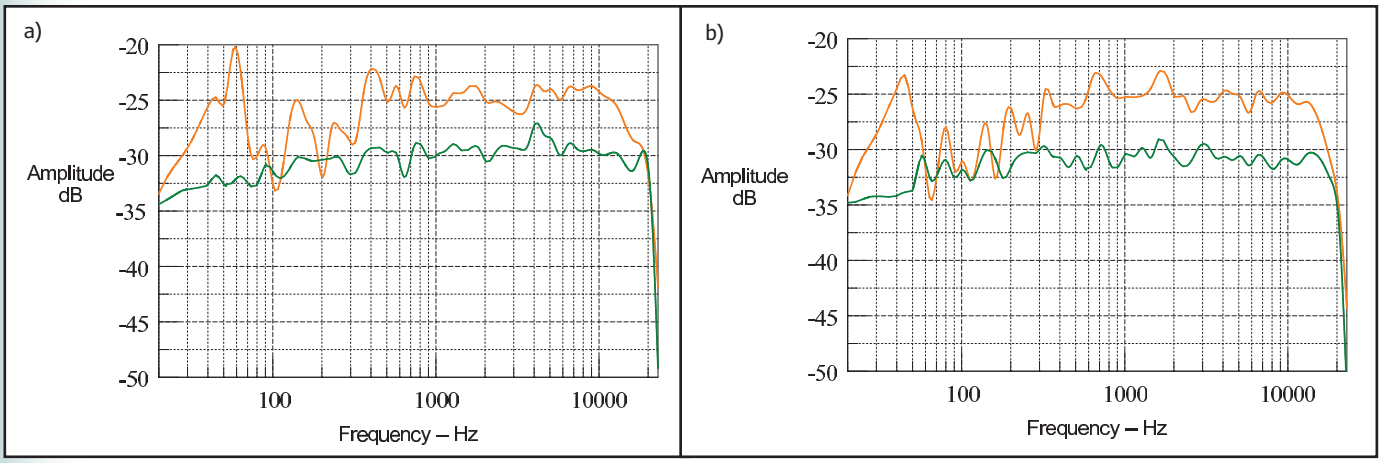

Figure 3: The left speaker uncorrected frequency response (orange line) and the green corrected response. Although not as flat at the target, the corrected response is much improved (a). The orange line is for the uncorrected right speaker with the corrected response shown in green (b).

Photo 6: This Correction screen, shown in the Reaper DAW, displays a portion of the Target Curve drop-down menu. Clicking on Virtual Monitoring displays a second menu as shown in Photo 7.

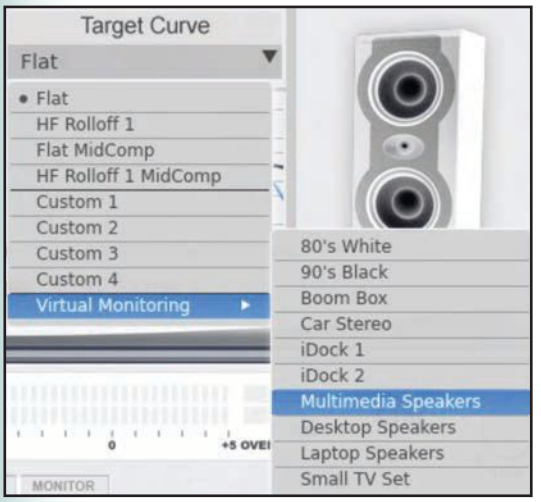

Photo 7: The detail of the Target Curves menu shows I selected Flat. Four Custom presets are available for programming any curve you might find useful. I reversed the colors to make this screen more readable.

Photo 8: The basic Correction screen shows the Left and Right measurements in orange with the target curve in white. The stereo bar-graph meter can be switched to the input signal by pressing the PRE button or to the output with the POST button. Levels above 0 dB are likely to produce clipping.

played this file directly into the sound card for the Before-EQ measurements and then through the ARC Correction plug-in for the EQ measurements. The mic's output was recorded by TrueRTA running on a second computer. **Photo 5** shows a typical recording. I exported each of the TrueRTA sixth-octave displays as text files and then imported them into a Microsoft Excel spreadsheet and averaged them. The averages are graphed in **Figure 3**.

The correction module has the setup screen (see **Photo 6**) with drop down menus for the correction file to be loaded and for the correction curve (see **Photo 7**). This menu includes a variety of presets including four custom targets. If you happen to be feeling wild and crazy, you could produce some interesting sounds. I just chose flat and sequentially loaded my two correction files.

The Left and Right orange lines (see **Photo 8**) are ARC's room measurements and the white lines are the calculated corrected responses. The green target line is hidden behind the white line. The bar-graph meters under the frequency response displays show the Left and Right levels either into

Photo 9: Pressing the MONITOR bar in Photo 8 loads this screen with LEVEL and MODE controls and a large VOLUME knob.

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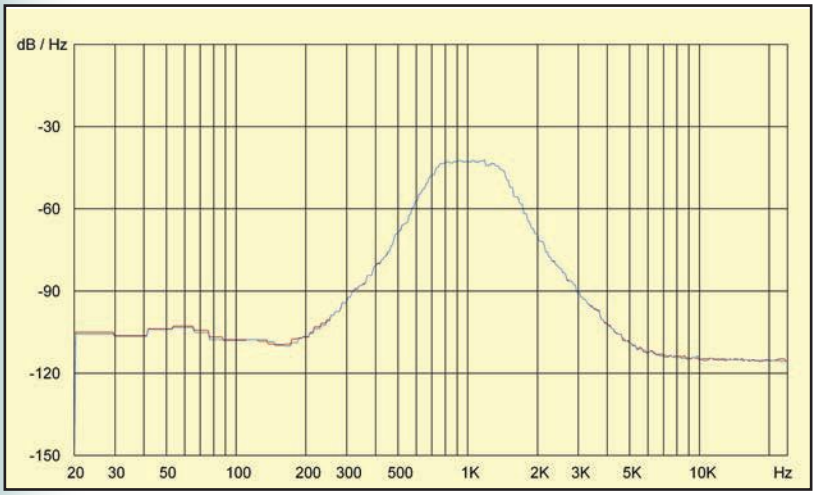

Photo 10: This graph shows the frequency response of a four-pole Butterworth band-pass filter with a 1-octave bandwidth as generated by the free Engineers Filter. Note the 60 dB dynamic range of this IIR digital filter. All eight filters I used for the RT-60 measurements have this shape and dynamic range.

Photo 11: Not having an omni-directional speaker system, I used my main WMTMW speakers for the RT-60 measurements with the MEMS microphone pointed toward the center of the wall behind the speakers. The volume control, active crossover filters, and power amplifiers are in the left-side equipment rack.

the module if the PRE button is depressed or the output level when the POST button is depressed. It's a good idea to keep the level below 0 dB to prevent clipping.

Figure 3 shows the Left and Right before and after correction frequency responses. The measured correction isn't as flat as the calculated white lines but they are a great improvement over the uncorrected.

Clicking the MONITOR bar near the screen's bottom displays the screen with its Level and Mode controls and large Volume control—attractive as well as functional (see **Photo 9**).

Time Response

As I mentioned earlier, ARC corrects the time response in addition to the frequency response. Before and after-correction waterfall plots would be useful, but not possible, because the needed phase information is not available because I couldn't use REW. So on to "Plan C" to measure reverberation time, RT-60, the time it takes an impulsive sound to decay to 60 dB below its initial value.

Getting a single measurement of a room's RT-60 is easily done with a balloon burst, a clapper, or gun shot. But measuring it as a function of frequency is more difficult without phase data. However, I found an informative video presentation by Prof. Clive Greated in which he explained how to do it—using band-pass filtered white noise.

It seems a four-pole Butterworth filter with a 1-octave bandwidth works well and may even be optimal. I searched online and found a free VST plug-in that's perfect for this job: Engineers Filter. Again, using Audacity, I generated and saved a two second white noise file. I filtered this file eight times saving the files as white-63.wav through white-8000.wav—each with center frequencies one octave apart. **Photo 10** shows a screenshot of one of the filters.

Ideally, I would use an omni-directional speaker and microphone but I used my main speaker system (see **Photo 11**) with the omni-directional MEMS mic pointed at the center of the wall behind the speakers. Using Reaper, I played each of the filtered files without ARC correction and recorded the mic's output with Audacity naming them NoEQ-63.wav through NoEQ-8000.wav. This was then repeated through the ARC correction plug-in.

Audacity is a good choice for analyzing these files because its amplitude scale can be easily

switched to decibel with the signal's peak at 0 dB and -60 dB at the center line. Measuring a 60 dB range is challenging so I normalized the wav file to -30 dB, measured the time and doubled it. But even working with a 30 dB range requires care. I found I got the best signal-to-noise ratio (SNR) by turning the mic preamp's gain down to 20 dB and increasing the speaker volume to LOUD. **Photo 12** clearly shows a 100 mSec decay time, which doubles to 200 mSec.

Last year I evaluated an actual "room correction system," the Primacoustic London 12 Kit of sound absorption panels. The article "Primacoustic London 12 Kit: Installation and Performance Tests" (*audioXpress*, February 2017) included my REW room measurement graphs. The after-correction, third- octave RT-60 data is included here in **Figure 4** along with my current before-and-after curves. Although the one-octave curves have less frequency detail, they are all very similar and well within the guidelines for a small room.

Even with the absorption panels the RT-60 times at the lower audio frequencies show

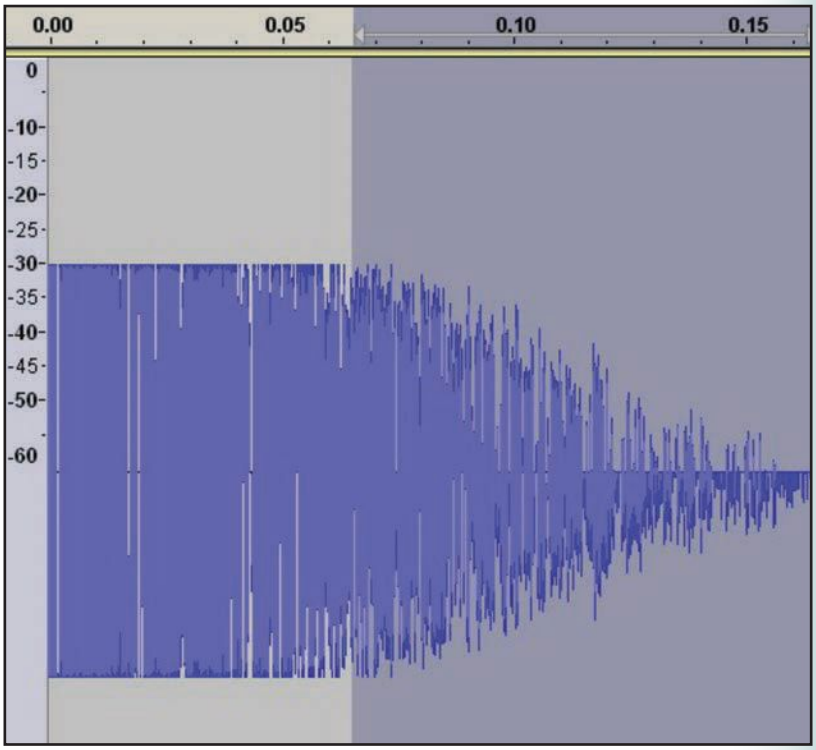

Photo 12: This is a typical ARC corrected RT-60 decay plot normalized to -30 dB in the free Audacity Audio Editor. The decay time as measured in the Audacity "Length" counter is 100 mSec, which doubles to 200 mSec for 60 dB decay.

a noticeable increase. The larger panels on the rear wall (see **Photo 13**) are only 2" thick which is minimally effective below 200 Hz even though they are separated from the wall by at least 3" (one is corner mounted) with mesh bags of Acousta-Stuf hung behind them. Thicker panels would have higher absorption below 200 Hz but were not included in this kit.

There is very little difference between the current before-and-after correction curves. It may be that the London 12 Kit provides sufficient time correction that the ARC system could not improve. Certainly, music sounds excellent through the ARC correction.

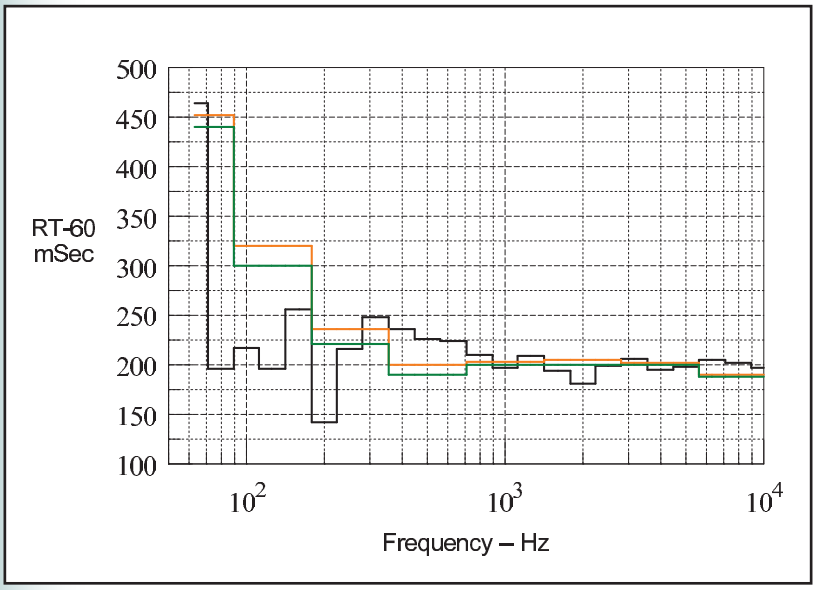

Figure 4: This RT-60 graph shows the corrected 1/3-octave data from my Primacoustic London 12 Kit review in black. The current uncorrected octave data is in orange and the green line is the corrected octave data. The current data lines are almost identical.

Photo 13: The two 2′ × 4′ absorption panels are designated as bass traps but at only 2" thick they are minimally effective below 200 Hz. The left panel is corner mounted and the other is 3" from the wall on special impaler brackets. Both have mesh bags of Acousta-Stuf hung behind them for some additional absorption.

Final Thoughts

The ARC system is very easy to use and quickly gives good correction. Measuring just the minimum seven locations around a single chair worked fine for me. But I expect if the listening area is larger, a sofa, or group of seats in a home theater, a larger number of locations would be better. I think it's a good buy at $199.99 and a real bargain if you have version 2.0. Buy the MEMS microphone for $69.99 and download the version 2.5 software from the IK Multimedia website. You can activate it with your 2.0 license.

Project Files

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

Resource

C. Greated, "Introduction to the Measurement of Reverberation," University of Edinburgh, Scotland.
www.youtube.com/watch?v=imHvRBOMg84

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Cockos, Inc. | www.reaper.fm

ARC 2.5 System
IK Multimedia |
www.ikmultimedia.com/products/arc2

Engineers Filter
KVR Audio, Inc. | www.kvraudio.com/product/engineersfilter_by_rs_met

Acousta-Stuf polyfill
Parts Express | www.parts-express.com

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Room EQ Wizard | www.roomeqwizard.com

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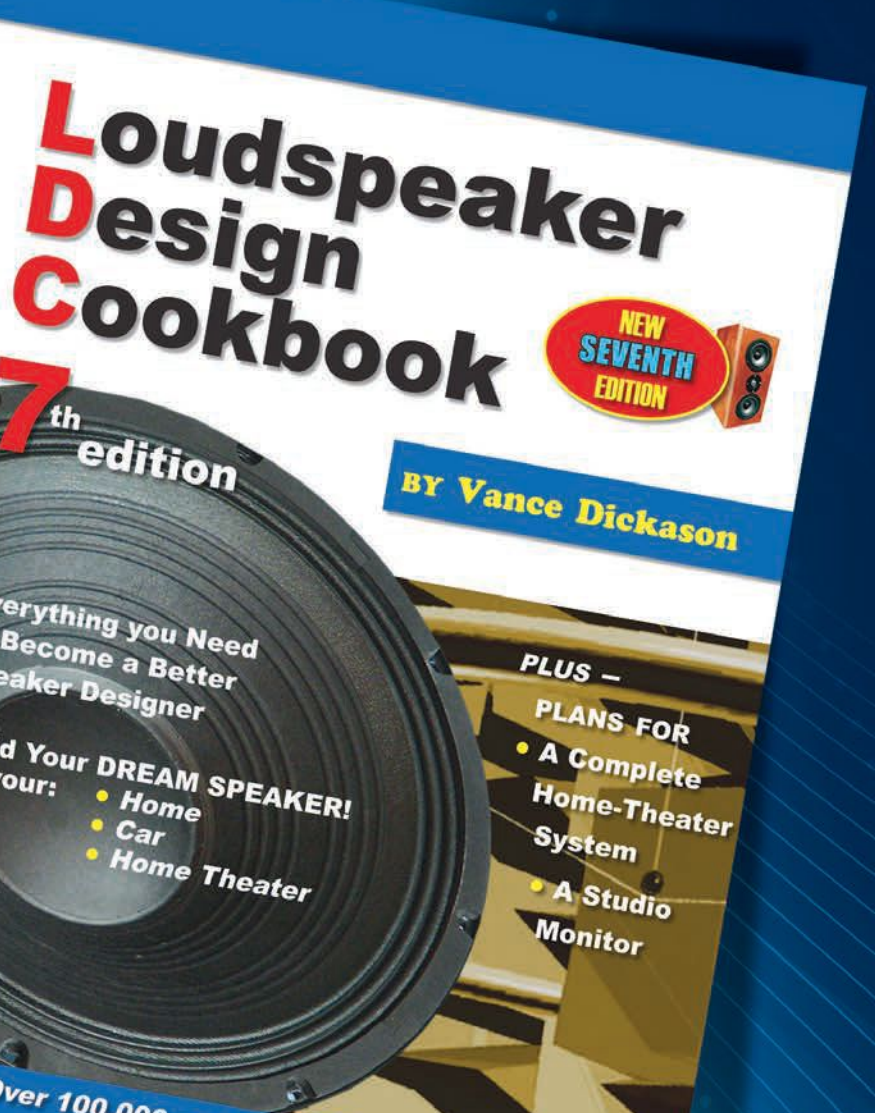

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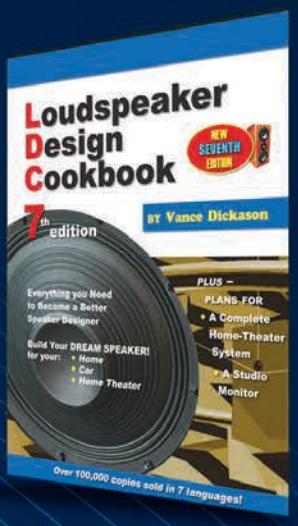

IK Multimedia ARC 2.5

The MEMS Test Microphone

By
Stuart Yaniger
(United States)

In this month's special Test & Measurement Focus, two of our authors tackle IK Multimedia's ARC System 2.5, which the company claims is the first acoustic correction system that combines a super accurate MEMS measurement microphone, measurement software, and a correction plug-in. In a separate article, Ron Tipton examines the system. Here Stuart Yaniger tests the new MEMS measurement microphone. The microphone is built around an omni-directional, high precision MEMS capsule to ensure maximum response accuracy and stability over time.

Photo 1: The MEMS microphone element used by IK Multimedia in what is, to the best of our knowledge, the first application of this type of component for measurement applications.

Smaller, cheaper, better: time's arrow for technology flies rapidly and inexorably. And so it is for microphones. Microphones based on Micro-Electronic Mechanical Systems (MEMS) technology are now made in the billions, and totally dominate the huge cellphone, tablet, and pad market. Compared with electret microphones, MEMS mics have a wide operating temperature range, good resistance to the stresses of soldering, and low current requirements. They are also physically small and light and mechanically rugged. The small size has enabled their use in arrays, which is a great aid to DSP noise cancelling in many devices.

MEMS transducers generally are constructed in a similar manner as microchips (i.e., by using a piece of high purity silicon as a base and performing a series of etching and deposition operations). This not only makes the resulting parts far more consistent than usual mechanical assembly, but makes the integration of signal conditioning much simpler. Most MEMS work on the condenser principle, that is, they have a diaphragm and a back plate with a constant charge, and the acoustic pressures cause the distance between them (and hence the voltage) to be modulated.

Until recently, the disadvantages of MEMS have included low dynamic range and uneven frequency response due to the Helmholtz cavity resonance from the front and/or back cavity. The small diaphragm size has also resulted in a significantly higher noise

floor. This is likely the reason that electrets have still dominated in lower cost measurement application. In the past few years, though, MEMS technology has improved to the point where it's a viable option for test and measurement applications.

The ARC 2.5 Room Correction System described by Ron Tipton in his review (also found in this edition) makes use of a MEMS-based measurement mic, which replaces an electret-based mic used in earlier versions of the system. Ron was kind enough to send me the old mic, which appeared to be configured for diffuse-field measurement. Running it through some basic tests, I quickly determined why IK Multimedia may have wanted to upgrade it—the top octave is severely rolled off (-10 dB at 20 kHz) and the distortion is relatively high. The current MEMS mic is said to have an omni-directional (free-field) polar pattern.

Microphone Setup

The MEMS mic is packaged in an injection-molded plastic case (see **Photo 1**), with the mic inlet on the top side when the mic is placed as directed (the tip pointed toward the speaker). The MEMS mic attaches via a standard XLR cable to the input of any microphone preamp that can supply phantom power. There's a small green LED on the top side as well to indicate when power is present.

Photo 2: The IK Multimedia MEMS microphone has an unusual geometry, with acoustic sensing done through the small hole on the top surface.

With the case removed, the MEMS element can be seen to be mounted on the top side of PCB and is a top-port sensor (see **Photo 2**). The use of a top-port MEMS and the top side mounting are necessary design elements to get a horizontal polar pattern that is omni-directional. Of course, the trade-off is that the vertical polar pattern will not be omni, which may be an issue for some measurement applications. Likewise, with this sort of mounting, the MEMS mic will not give accurate near-field results for woofer

measurements. The last point turns out to be moot, as we will soon see.

Although the MEMS element does not have a readable marking on it, it looks similar to the Knowles SPH1642HT5H-1, part of the SiSonic series, and referencing the datasheet for that part (see Resources) gives some insight into the operation of this family of elements (similar parts can be sourced from InvenSense and PUI, among others). The MEMS element includes the sensor itself, a buffer, and an output amplifier. The other parts on the PCB are used to convert phantom power into an operational voltage suitable for the MEMS element and buffering circuitry to drive cables and a mic preamp (see **Photo 3**).

Measurements

All my measurements were done using an Audio Precision APx515 analyzer and an APx1701 transducer interface. The reference microphone was a calibrated PCB Piezotronics 376A32 0.5" phantom powered pre-polarized condenser microphone (the subject of an upcoming review), which comprises a 377B02 capsule and a 426A14 preamplifier.

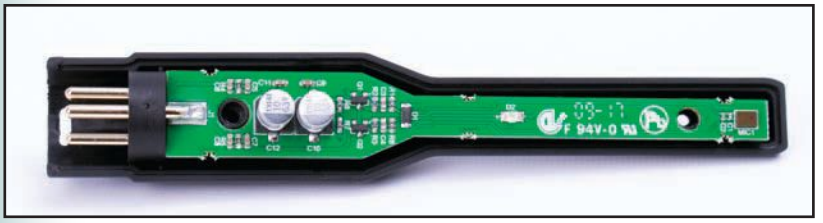

Photo 3: The circuitry is laid out on a double-sided PCB, with the MEMS element seen on the far right hand side (Photo courtesy of Cynthia Wenslow)

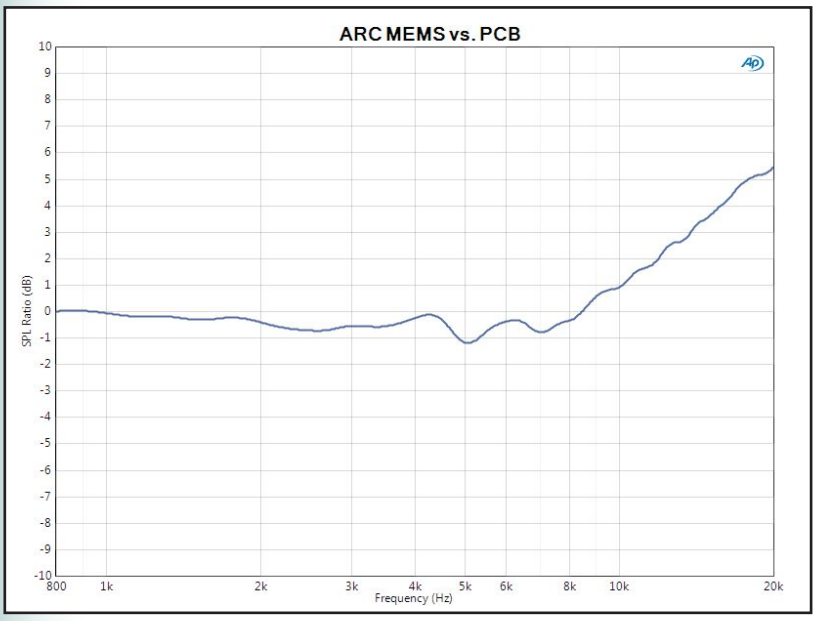

Figure 1: The frequency response of the IK Multimedia MEMS microphone is very even through the midrange, but shows a significant rise in the top octave.

The performance specifications provided for the IK Multimedia MEMS mic are somewhat rudimentary and no sensitivity is given. I measured it to be 12.4 mV at 94 dB SPL (or -38 dB). The quasi-anechoic on-axis frequency response of the MEMS microphone above 1 kHz is shown in **Figure 1**, and was obtained by comparison to the reference using a swept tone stimulus played through a Vanatoo Zero powered loudspeaker at a 1.5 m distance—taking the ratio of the two microphones' responses removes the loudspeaker's frequency response from the measurement. The impulse response derived from the swept tone was gated to remove room effects and yield the quasi-anechoic measurement.

Many MEMS elements show a peak at 21 to 25 kHz, which is used as a way of extending treble response. In this case, the resonance was at about 23 kHz, and the peaking caused a smooth response rise in the top audible octave, with the 20 kHz response being 6 dB elevated compared to the midrange. The response was otherwise smooth, and this rise was easy to equalize out in software (as IK Multimedia almost certainly did in the ARC 2.5 system). If you plan to use this mic for speaker measurements, you can either electrically equalize out the treble rise with some first-order RC filtering or compensate in your measurement software. If we assume that the MEMS element is the Knowles MEMS or something similar, this should be quite consistent from part to part, so a good approximation for correction is a single pole low-pass filter with a -3 dB point at about 15 kHz.

Figure 2 shows the bass response of the IK Multimedia MEMS mic, obtained by measurement of a woofer with both the reference PCB Piezotronics mic and the ARC MEMS, then taking the ratio of the results. From this graph, you can see that the -3 dB point for the MEMS mic is at about 65 Hz, which is high enough to rule out using this microphone for most woofer measurements. As mentioned before, the construction of this mic also prevents it from being used for most near-field woofer measurements.

I measured the IK Multimedia MEMS mic's self-noise by using the BLANK chamber I have described in previous articles (basically burying the microphone in a tub of sand and kitty litter). With a 20 kHz bandwidth and A-weighting, the noise was 29 dB SPL equivalent. This is 65 dB below the usual 94 dB SPL reference, so the mic exactly meets its 65 dB signal-to-noise ratio (SNR) specification.

When I started doing polar pattern measurements, I saw some rather complex lobing in the upper midrange and treble. **Figure 3** shows why this might be—this graph is the ratio between on-axis and 180° frequency response. For a perfectly omni-directional

microphone, this should be a flat line at 0 dB. For real-world omnis using traditional technologies (condenser or electret), one usually sees a treble rolloff of several decibels and a few small ripples from diffraction. The IK Multimedia MEMS mic has an effective diaphragm diameter of roughly a millimeter, so the usual treble rolloff is not evident—in that sense, this is the best I've ever seen in a mic measurement. Because of the unusual mounting of the element, diffraction effects become more significant, and this graph shows some pretty significant peaks and dips in the upper midrange at 180° (especially at 7 kHz, which may correspond to the distance between the MEMS port and the front of the microphone) that do not appear in on-axis response.

My usual method for measuring microphone distortion is to do a near-field measurement at low frequency on a woofer. The small distance from the woofer means that for high SPL, the excursion (and hence distortion) can be quite low. This method won't work for the MEMS mic because of the top-ported layout, but in the spirit of improvisation, I did some 1 m on-axis distortion measurements of a loudspeaker system and compared it to the same measurements acquired with the reference PCB Piezotronics 376A32

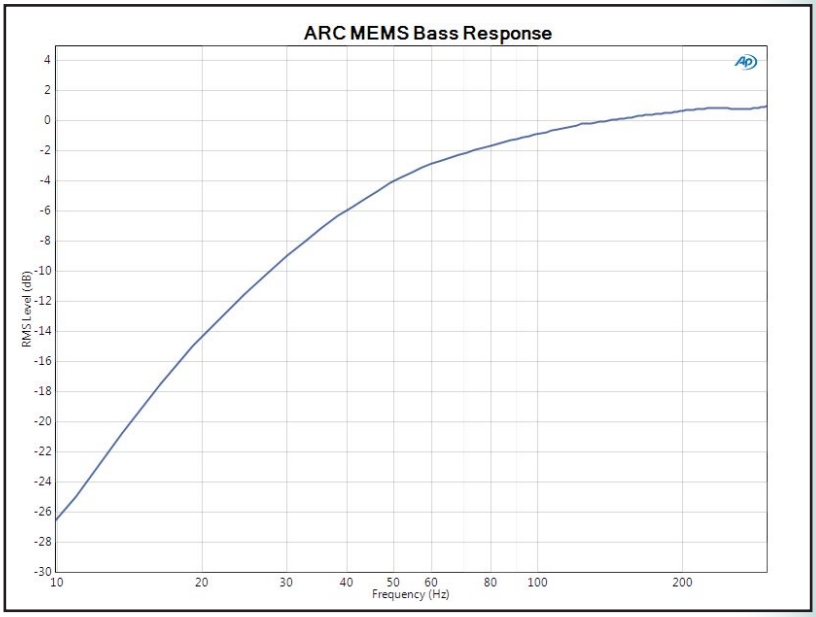

Figure 2: The bass response of the IK Multimedia MEMS microphone is rolled off, with the -3 dB point at about 65 Hz.

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.

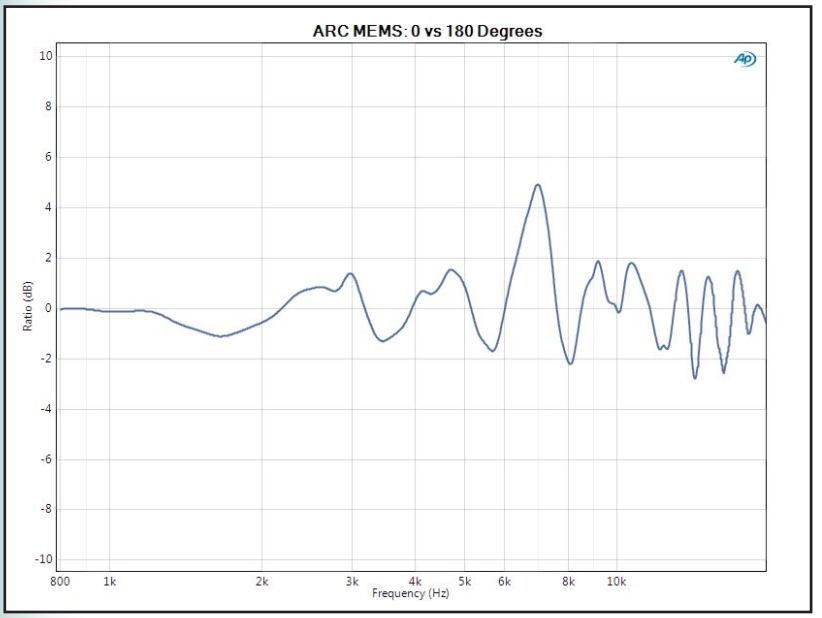

Figure 3: A comparison of the MEMS microphone response on axis and at 180° shows some diffraction-related ripple, but no rolloff in the top octave.

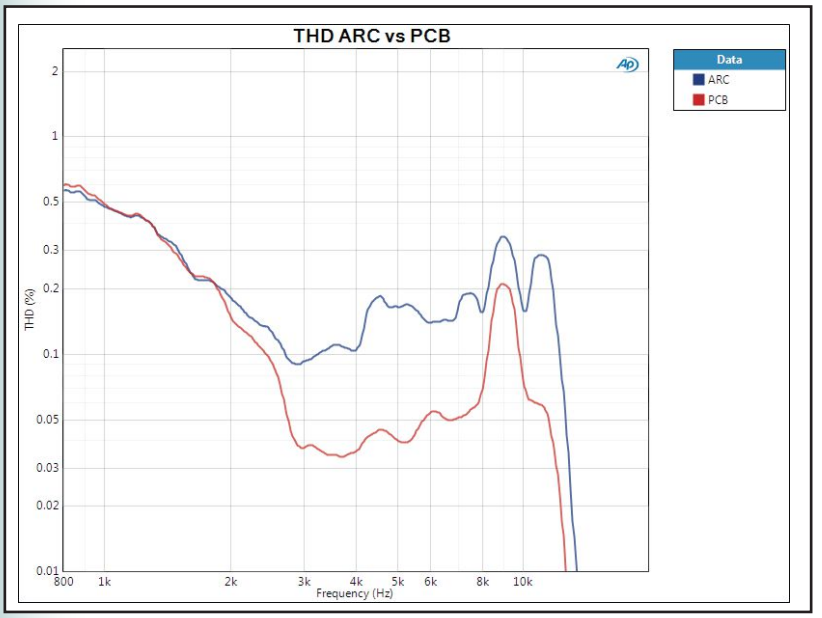

Figure 4: A comparison of distortion measurement of a loudspeaker between the IK Multimedia MEMS microphone and the reference mike shows the distortion level at moderate SPL is 0.1-0.2%.

Sources

IK Multimedia ARC System 2.5
IK Multimedia | www.ikmultimedia.com/products/arc2

Knowles MEMS microphone element
Knowles | www.knowles.com

microphone, which has the lowest distortion of any mic I've tested so far. **Figure 4** shows the distortion vs. frequency of the loudspeaker at 78 dB SPL measured with both microphones. At low frequencies, the loudspeaker distortion predominates, but at higher frequencies, the distortion drops below the MEMS mic's distortion floor, which is about 0.1 to 0.2% THD. At the same frequencies, the PCB mic shows about 0.04% distortion, which is likely limited by the loudspeaker, not the microphone. By comparison, the last generation microphone for the ARC 2.0 room correction software showed about 0.5% THD at the same SPL. The new MEMS mic is a significant improvement.

Bringing the MEMS mic in to about 1" away from the woofer cone, the 3% distortion point at 200 Hz was about 119 dB SPL. This is consistent with the Knowles MEMS microphone's element rating of 124 dB SPL for 10% distortion. Retesting at 1 kHz, the distortion measurement looked about the same, so it is apparently related to diaphragm displacement rather than turbulence in the mic element cavity.

Conclusions

I'm always fascinated when a new technology crashes the party. The low-cost, solderability, and environmental resistance of MEMS are compelling advantages, and the tiny diaphragm size suggests some interesting potential virtues. For this microphone using the latest generation of MEMS sensors, the disadvantages (noise and frequency response flatness) are demonstrably improved compared to earlier generations. The IK Multimedia MEMS mic works beautifully with its ARC 2.5 system, but the technology still has not quite reached the point where it can be used for general measurement because of the bass rolloff and relatively high distortion. The rear response at the top octave is impressive, though, as is the midrange flatness on axis.

For recording use, the 29 dB noise floor is higher than one might want, at least for use in a studio. For some outdoor or live concert applications, this might be good enough, and the excellent environmental stability and ruggedness of the MEMS sensor may be more important than the noise floor.

Overall, this MEMS microphone is a significant upgrade compared to the mic previously provided with the ARC system, and its virtues (ruggedness and stability) are considerable. The pairing with the ARC 2.5 room correction software made a significant improvement in the sound of my recording monitors. I'm looking forward to seeing what the next generation of MEMS sensors can do to expand the capability of that technology.

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By
David Mathew
(Audio Precision)

As with all technological devices, loudspeakers, headphones, ear buds, MEMS speakers, sonar emitters and police sirens need to be measured and tested, both in design and in production. Further, products that make any noise at all (motors, airplanes, wind turbines, coffee makers, HVAC) are often measured for safety or environmental impact, or are continuously monitored, listening for signature acoustic signals that indicate correct performance, or failure.

The sensors required to accurately acquire the acoustic signals for test and evaluation are not the rock'n'roll mics a drummer would have arrayed around his kit. A wide range of measurement microphones from a number of manufacturers has existed for many years, with a special solution for every acoustic test need (see **Photo 1**).

A measurement microphone is like an ordinary microphone in the superficial features: It is typically tubular, with a sensor at one end and a connector at the other, and the sensor itself is a lightweight diaphragm that is excited by changes in air pressure, responding in a way that can produce an electrical signal. But at this point the two microphone types diverge (see **Photo 2**). You won't see a singer's wireless mic measuring loudspeaker drivers in an anechoic chamber, and you won't see a comedian using a measurement microphone for the mic drop at the end of his routine.

Photo 1: Audio Precision offers a selection of measurement microphones designed to provide ready solutions for professionals working in acoustic test.

Characteristics

Measurement microphones are optimized for superior performance in one or more of these characteristics: frequency response, frequency range, self-noise, maximum level, and distortion. Further, some are designed to be robust in harsh environments, or to have characteristics that closely match in an array application. Microphone sensitivity and frequency response are very stable over time. A measurement microphone is typically shipped with a calibration table or chart documenting its performance (see **Figure 1**).

TEDS

A transducer electronic data sheet (TEDS) is a standardized method of storing transducer (sensors or actuators) identification, calibration, correction data, and manufacturer-related information. TEDS formats are defined in the Institute of Electrical and Electronics Engineers (IEEE) 1451 set of smart transducer interface standards developed by the IEEE Instrumentation and Measurement Society's Sensor Technology Technical Committee.

The standards describe a set of open, common, network-independent communication interfaces for connecting transducers to microprocessors, instrumentation systems, and control/field networks.

One of the key elements of the IEEE 1451 standards is the definition of TEDS for each transducer. The TEDS can be implemented as a memory device attached to the transducer and containing information needed by a measurement instrument or control system to interface with a transducer.

However, TEDS can be implemented in two ways. First, the TEDS can reside in embedded memory, typically an EEPROM, within the transducer itself which is connected to the measurement instrument or control system. Second, a virtual TEDS can exist as a data file accessible by the measurement instrument or control system.

A virtual TEDS extends the standardized TEDS to legacy sensors and applications where embedded memory may not be available. If you've been used to using sensors that are TEDS-enabled, you'll be pleased to know that most modern measurement microphones carry identification and calibration information as TEDS data.

Frequency Response

Typical measurement microphones are specified as ± 2 dB from 5 Hz to 20 kHz, but some models have usable response as low as 0.07 Hz, or as high as 140 kHz.

Low Noise

Most measurement microphones have a noise floor of about 20–40 dBA, but specialized 1" models can spec a noise floor as low as –2.5 dBA.

Maximum Level

For measurement microphones, 3% total harmonic distortion (THD) is considered overload. Typical measurement microphones might overload at 160 dB; specialized models will not overload until 184 dB or more.

Sensor Design

A number of methods have historically been used to convert sound pressure to an electrical signal:

- Piezoelectric, using a crystal attached to a diaphragm
- Variable resistance, using packed carbon granules in a small container, attached to a diaphragm
- Dynamic, using a magnet and a coil to convert diaphragm movement to a current
- Variable capacitance, where the diaphragm itself is one side of a capacitor, converting the movement of the diaphragm into a voltage

As it turns out, the capacitive method will, in most applications, provide the most sensitive microphones, largely due to the low diaphragm mass that this method makes possible. A survey

Photo 2: This type of microphone would be used by your average garage band (a) while measurement microphones are used to take precision measurements (b).

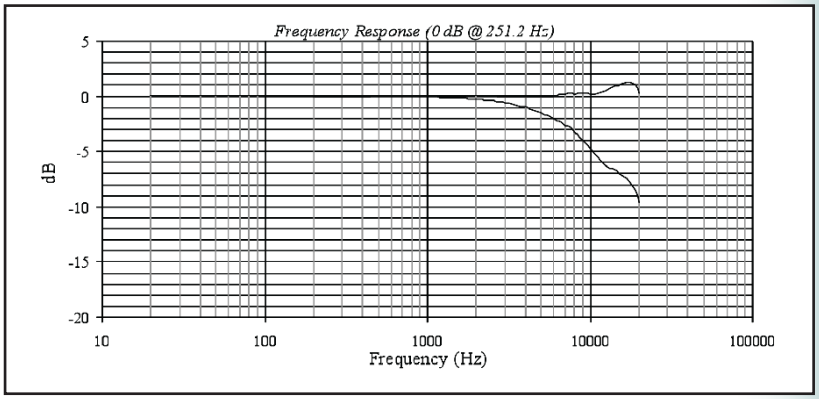


Figure 1: A measurement microphone is typically shipped with a calibration table or chart documenting its performance.

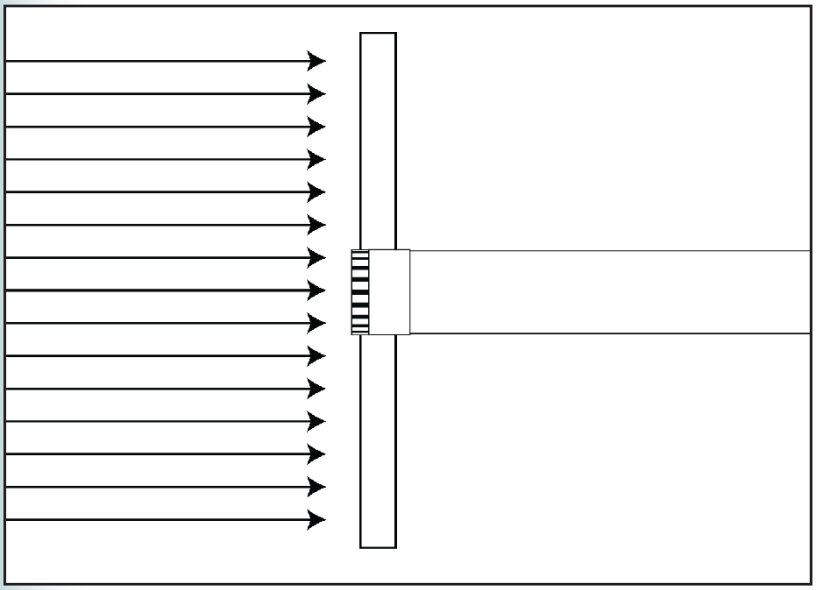


Figure 2: A microphone's pressure response is flat when its presence does not disrupt the pressure wave.

of measurement microphones over the past 50 years reveals wide use of capacitive microphones. In microphone circles, capacitive microphones are often called condenser microphones, and that is the term we will continue with here.

The one exception is an application where the sound levels are very high, such as near a blast or explosion. In this case, a piezoelectric measurement microphone is the correct choice.

Powering Condenser Microphones

A dynamic microphone can simply be connected via a shielded cable to an appropriate downstream amplifier and put to work. Condenser microphones, however, require more support:

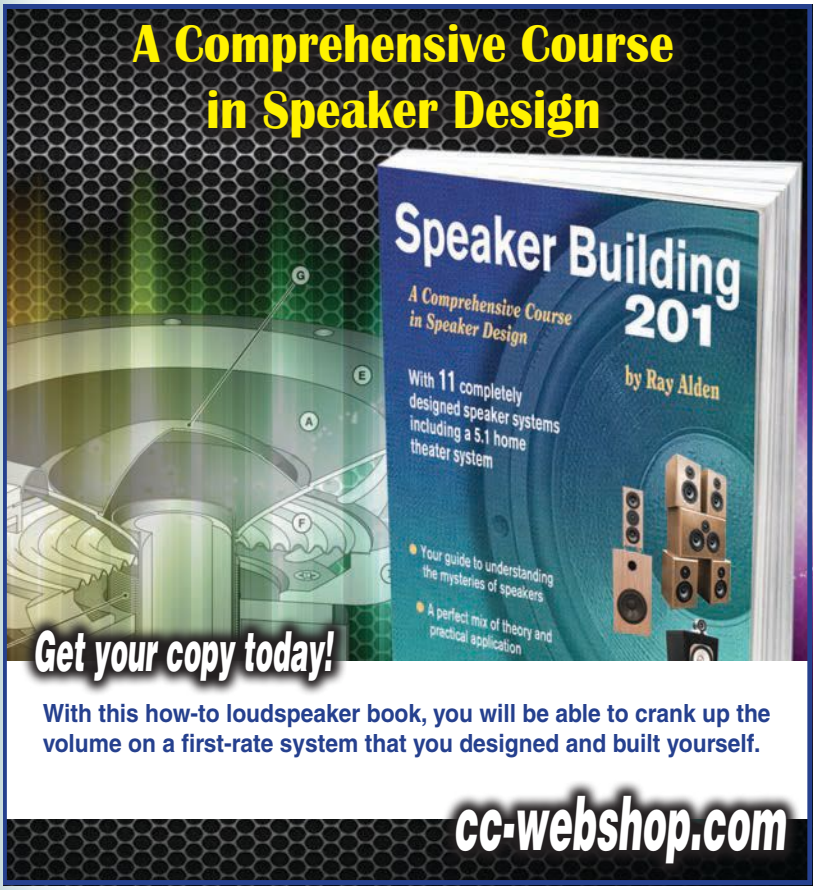
- The capacitive sensor element requires a polarizing voltage.
- The impedance of the sensor element is very high; consequently, the signal current is so small that it must be amplified at the source before it is swamped by noise.
- Condenser microphones always have a pre-amplifier either built into the microphone body, or connected directly to the microphone sensor capsule.

Prior to the introduction of solid-state amplifiers, the preamplifier in a condenser microphone was of a vacuum tube (valve) design. These microphones required custom power supplies and multi-conductor cables that provided the capacitor polarizing voltage and also plate voltage and filament current for the tube.

Today, measurement microphone preamplifiers are solid state and have modest power requirements. Depending upon applications, some microphones are externally polarized and require a 200 V polarizing voltage; many other designs are pre-polarized, with an electret capacitor as the sensor element, and require only preamplifier power. Early electrets were not suitable for high-performance applications, but modern electret microphones offer excellent specifications and long-term stability.

Diaphragm Size

Measurement microphones are offered in four nominal diaphragm sizes: 1", 0.5", 0.25", and 0.125". Generally speaking, the smaller the diaphragm, the greater the self-noise, the higher the frequency response, and the higher the maximum level. Most general applications are satisfied with 0.5" measurement microphones.



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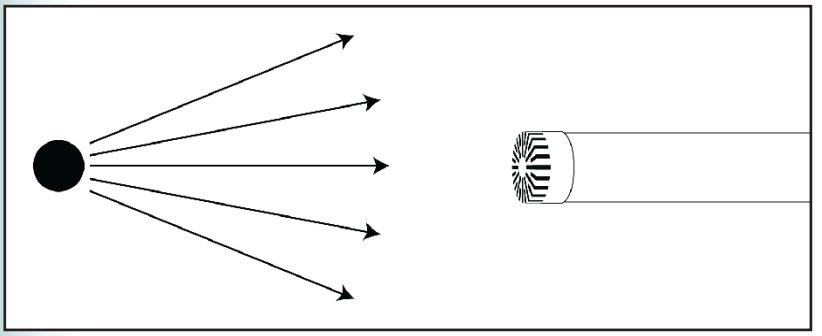


Figure 3: A free-field microphone is compensated to produce a flat response when used in an anechoic space where the sound waves are arriving from one direction.

Directional and Sound Field Characteristics

Engineers with some experience in sound amplification or recording might be familiar with microphone directional patterns such as cardiod, figure of eight, shotgun and so on. These characteristics are accomplished by modifications to the basic diaphragm element, such as acoustic ports, additional diaphragms, or interference tubes.

Measurement microphones, on the other hand, are omnidirectional, without modifications for directionality. Measurement microphones are optimized for one of three acoustic applications: measuring sound pressure, measuring incident sound from one direction in a free-field (anechoic) acoustic space, and measuring sound that may arrive from any direction (random incidence) in a diffuse-field acoustic space.

Effect of the Measurement Microphone on Incident Sound Waves

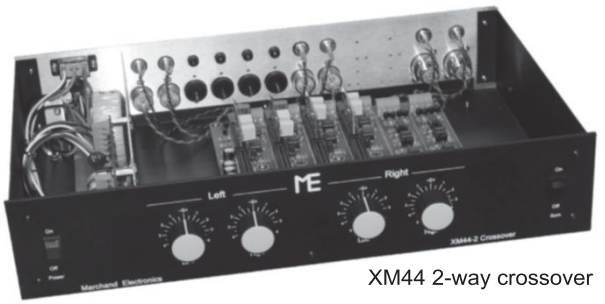
The mere presence of a microphone in an acoustic space disrupts the sound pressure wave as it encounters the microphone. The wave reflects from and diffracts around the sensor element to varying degrees, dependent upon the dimensions of the microphone and the frequency and angle of incidence of the sound wave. This effect is avoided in the first case below, the pressure microphone (see **Figure 2**).

A microphone's pressure response is flat when its presence does not disrupt the pressure wave. This occurs when the microphone is not in the

ME

Electronic Crossovers:

Solid State, Tube,
Passive Line Level
6,12,18,24,32,48 dB/octave
Butterworth, Linkwitz-Riley, etc.



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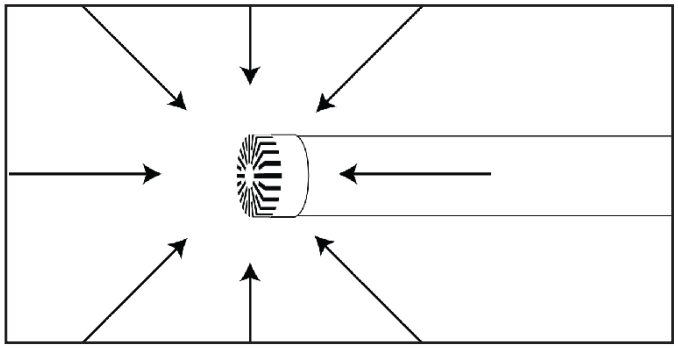

Figure 4: A diffuse-field microphone is compensated to produce a flat response when used in a reverberant space.

sound field, but is a component of the barrier containing the sound field. Applications include flush mounting within an acoustic coupler, or flush mounting on a wall or barrier.

A free-field microphone is compensated to produce a flat response when used in an anechoic space where the sound waves are arriving from one direction (see **Figure 3**). Applications include loudspeaker testing, microphone testing, evaluations and monitoring of sound-emitting equipment, and sound-level meters. The sound field must be free of reflections, such as an anechoic chamber or use out-of-doors.

A diffuse-field microphone is compensated to produce a flat response when used in a reverberant space such as a church, a concert hall, a room, or an aircraft or automobile cabin (see **Figure 4**). Applications include room tuning, impulse-response testing, and ambient industrial or environmental noise evaluation.

Microphone Arrays

Some applications require a geometric array of two or more (sometimes many more) matched microphones to capture temporal, directional and phase information for mathematical analysis. Array microphones are typically of the free-field type, with careful attention paid to close phase-matching among the microphones. Because a large number of microphones may be required, array microphones are usually of a general-purpose (and therefore, less expensive) design.

Resource

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About the Author

David Mathew is Technical Publications Manager and a senior technical writer at Audio Precision in Beaverton, Oregon. He has worked as both a mixing engineer and as a technical engineer in the recording and filmmaking industries, and was awarded an Emmy for his sound work in 1988.

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Perseverance and Innovation Leads to Success

An Interview with Steve Temme, President of Listen, Inc.

By
Shannon Becker
(United States)

Steve Temme founded Listen, Inc., in 1995 after teaching himself LabVIEW programming and working long hours to develop the coding for his company's flagship product, SoundCheck.

Listen, Inc. is a world leader in audio and electroacoustic test and measurement. Founded in 1995, when its flagship product SoundCheck was launched, the company has pioneered electroacoustic measurement techniques for more than 20 years. Listen has set the standard in the marketplace with powerful, innovative test methods and algorithms, and fast, flexible testing.

SHANNON BECKER: Tell us a little about your background.

STEVE TEMME: I have been fascinated with audio since the day that my father brought home a "close and play" record player when I was a child. The sound quality was terrible but it was better than nothing. After hearing some of my friends' high-end stereos, I quickly realized that there were better audio systems out there, and resolved to buy one myself to replace the "close and play" at home. My interest in audio continued in my teenage years, including going to Boston Symphony Hall with my father to listen to orchestral music and eventually I went to Tufts University to study Mechanical Engineering. While I was there, I started a business selling stereo equipment on campus, which financed my own audio equipment purchases as well as helping to pay my way through college.

SHANNON: How did you become interested in audio and electroacoustic test and measurement?

STEVE: Like many people in this industry, my real audio education began at Brüel & Kjær. Straight after college I began working there as a customer support engineer where I answered customer support phone calls and followed up with typed letters (actual hard copy—there was no email back then!). Eventually I moved into sales and then was offered a job at one of my loudspeaker manufacturing customers as they needed someone who knew how to operate the dual-channel Fast Fourier Transform (FFT) audio analyzer that I was trying to sell them. At Apogee acoustics,

in addition to audio test, I was involved in the design of their first active crossover for their flagship Diva loudspeakers, which I still own and regularly use at home. My time there cemented my love for high-end audio equipment, which has remained to this day. After that I moved to Denmark to again work for Brüel & Kjær, this time as applications engineer to make a sales video for the new 2012, one of the most famous acoustic analyzers of all time. It was here that I acquired a deep understanding of all things related to acoustic test and started writing Audio Engineering Society (AES) and other technical papers on the subject. I also started to think there must be a better and less expensive way of making audio measurements, and the idea for Listen, Inc. was born.

SHANNON: Tell us more about the start of your company, Listen, Inc.?

STEVE: In 1994, I returned to the US after living in Europe for four years. Having been a serial entrepreneur through my teenage and college years, I was determined to start my own business and was evaluating several ideas. At this point, Windows 95 had just been released, bringing with it the option to use multimedia sound cards. With my strong background in audio test and measurement, I realized that the flexibility offered by software, personal computers and sound cards could replace conventional hardware-based audio analyzers, offering less expensive, faster, more powerful and more flexible audio analysis. While I spent my evenings developing a business plan for a lower-cost PC-based audio test system, I was also working several audio consulting jobs in order to pay the bills.

SHANNON: How did SoundCheck come to be?

STEVE: The turning point for me came on a visit to Sonetronics, one of my consulting clients, and a manufacturer of military headsets. Sonetronics had purchased several 2012 analyzers from Brüel & Kjær for production line testing, and had hired me to assist with developing the tests and getting the systems installed on the production line. The first time that I met Sonetronics President Gary Kuskin, he asked me if their measurements could be done with a PC and a sound card, as the hardware systems were expensive, slow, and he would need additional systems to meet his production testing needs. I replied that yes, it was definitely possible. Although I couldn't persuade Gary to pay me to develop the systems, he quickly agreed to buy several systems at a negotiated price if I could deliver them quickly.

The Close 'n Play record player inspired Steve Temme's lifelong mission to improve audio quality.

Hoping that I could build the system and make money from it, I began developing the first version of the product that would become SoundCheck. I had already decided that the main weakness of most measurement systems on the market was that the operator needed to understand programming in order to configure tests, so I decided to build SoundCheck on the National Instruments (NI) LabVIEW platform. This enabled tests to be built by selecting from a range of pre-configured steps using a point and click interface which hugely simplified test setup as no programming knowledge was required

Next, I looked for a programmer to partner with to develop the system. However, with no money to pay anyone, I was looking for a consultant who would write the code in exchange for a share of the profits when the system was sold. Unsurprisingly, no one

The original SoundCheck system is pictured in use at Sonetronics (circa 1995).

was willing to work on this basis, but I was advised that LabVIEW was not difficult to learn, and my best option would be to teach myself. Armed with a book and some knowledge from a few NI users, I taught myself LabVIEW programming and started coding SoundCheck.

For six months I programmed day and night, upwards of 60 hours a week. In addition to programming LabVIEW, I also needed to learn a lot about testing to military specifications, which necessitated weekly 10 hour round trips to work with engineers at Sonetronics. The return trip by car was always done in a day, as hotels and flights were way out of my budget!

Six months after starting the project, I had a prototype ready for evaluation. For the most part, the system performed as expected, showing the same results as the more costly hardware system, and the customer was happy. Some minor tweaking and refining ensued, and the customer agreed to purchase the system. The system was named SoundCheck, Listen was formed, and I was in business!

SHANNON: What were some of the greatest challenges in creating SoundCheck?

STEVE: While there were many technical challenges, most of these were fun to overcome, as I was doing something I enjoyed. Perhaps the greatest technical challenge in the early days was developing the "Stweep" test signal and the corresponding HarmonicTrak algorithms. Most test systems back then (and even today), used stepped-sine sweeps followed by a tracking filter. When stepping from one frequency to the next, there was an audible transient caused by a phase discontinuity transitioning from the old to the new sine frequency. With the help of some former work colleagues, we came up with a phase continuous transition and a smart FFT that allowed us to play the "Stweep" in one go and analyze afterward, using an auto-delay algorithm to align the FFT with each frequency. This saved considerable time and captured all the harmonics simultaneously. The PC processors were getting so fast that the total test time was approaching the "Stweep" time.

More significant than the technical challenge was doing most of it alone, as it was hard to find partners who were willing or able to take a chance on investing a lot of time in a project with no guaranteed financial return. At the same time as spending long hours writing code, I was also continuing my consulting work to keep paying the bills, and managing huge amounts of credit card debt, moving it from one card to another every time the initial interest free period expired. Juggling all that, while working on a product for which I still had to find customers, rather than taking home a paycheck was probably the hardest part of all!

SHANNON: What made SoundCheck so different from the others on the market?

STEVE: This has changed over the years. In the beginning, it was a truly revolutionary idea using a high-end pro-audio sound card as the audio interface rather than an expensive hardware box or data acquisition card. We also designed our own measurement microphone power supply since existing ones were so expensive. This brought the cost of automated testing down to less than half its previous cost, making it a financially viable option for production lines. The early days also saw a focus on creating fast tests to meet the needs of high-throughput production line testing. I invested heavily in developing specialist algorithms which offer faster and more accurate measurements. In fact the STWEEP and Harmonictrak algorithms that I programmed into the first version of SoundCheck are still in use today.

Innovation has been a key selling point through Listen's 22-year history. We were the first company to implement the Farina sweep (Log TSR) in 2001 just months after his landmark paper on the subject—something that other companies have only introduced in the past 5 to 10 years. It helped that I was already considering it because of its speed

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SoundCheck is being used for driver testing on a production line in China.

benefits before the paper was presented! We were the first to offer algorithms for robust Rub & Buzz detection algorithm using higher order harmonics, loose particle detection, perceptual Rub & Buzz, non-coherent distortion measurement with music, and more.

I think the big advantage I had coming to it in 1995 was that with the latest technology and no legacy system, I was in a position to build a system for the computer age. This turned out to be hugely beneficial because, as devices have become more sophisticated and interfaces have changed, there has been no need to change the underlying architecture of the system. This means that with the evolution of the algorithms contained in the software, plus some additional low-cost interfaces, measurements that back in 1995 were made on a simple speaker driver can now be made on an internet-connected smart speaker, Bluetooth headphone or any other device.

SHANNON: Can you tell us, which of your products you feel have had the greatest impact on the industry.

STEVE: Without a doubt, out of all of our products, SoundCheck has had the biggest impact on the industry. I believe SoundCheck was largely responsible for the shift from human listeners to automated end of line production testing throughout the 2000's. Back in the beginning I used to have to try to convince people to invest in automated end-of-line testing because it was more reliable and cost-effective than human listeners. Now others have entered this marketplace and there is no doubt that automated systems are needed for any new production line - the question is 'which brand?' With the wide adoption of SoundCheck in recent years, and most of the Fortune 500 consumer electronics companies using SoundCheck, I think it's fair to say that most people own at least one product—probably more—that is tested (or its components tested) with SoundCheck.

SHANNON: How have your products evolved over the past 20 plus years?

STEVE: Some things stay the same and some things evolve! SoundCheck is still SoundCheck (and still costs the same as back in 1995), yet it is a very different product. It has hugely increased functionality, is equally popular for both R&D and production, and has expanded in line with new developments in the audio industry. For example, since its early days an equation editor was added which enabled calculation of Thiele-Small parameters, or any user-defined mathematical calculation, making it as suitable for the R&D lab as the production line. Many new algorithms have been added, for example, Perceptual Rub & Buzz measurement, Time-frequency analysis, and algorithms to implement the post-processing frequency shifts necessary to test Bluetooth devices and cloud-connected devices such as smart speakers. Additional hardware integration options have been added so that the hardware necessary to measure today's devices, such as Bluetooth, MEMS microphones, etc. can be controlled from within SoundCheck. While the sequence editor and noise immunity have always made it a great option for production line measurements, today SoundCheck also features the ability to incorporate it as part of a wider factory automation system with control via Python and TCP/IP and various saving to database options. It has also expanded from a two-channel system to a system that can simultaneously acquire and process as many channels as the audio interface could handle—we have customers successfully testing up to 64 channels simultaneously.

Hardware has also been an area of great innovation over the years. Although Listen was initially a software company, I realized fairly early on that although our software would work with any hardware, there was an advantage to creating our own designed specifically for audio measurement purposes. We first started creating hardware in 1997 with the SoundConnect microphone power supply. In more recent years, we launched the hugely-popular AmpConnect all-in-one audio interface which significantly streamlines the test setup, and products such as AudioConnect (audio interface with built-in microphone power supply), SoundConnect 2 (a two-channel microphone power supply), and SCAMP (an amplifier specifically designed for audio testing).

Through our partnership with Portland Tool and Die, we also now offer dedicated interfaces for MEMS microphone testing, Bluetooth testing, and more.

My personal area of interest is distortion measurement, and we have always been on the cutting-edge of this area of electroacoustic test. We have launched many innovative distortion measurement methodologies over the years including the first robust Rub & Buzz detection algorithm using higher order harmonics, loose particle detection, non-coherent distortion measurement with music, and most significantly, Perceptual Rub & Buzz. We continue to innovate in this area and will have some additional developments soon.

SoundCheck is used for headphone testing at Reviewed.com, a review website that objectively measures and compares hundreds of headphones.

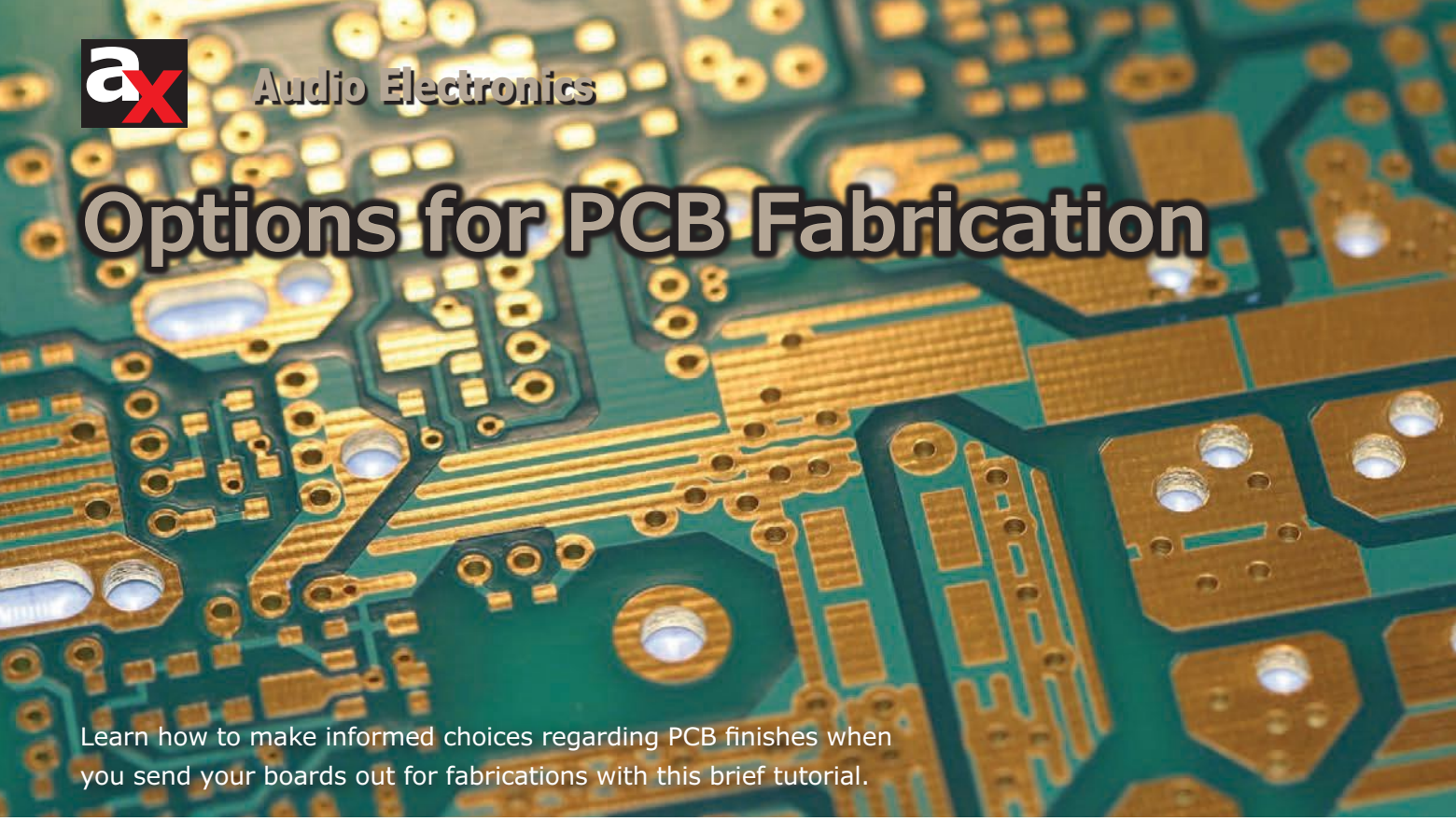
SHANNON: Can you share your personal views on market trends and where you see the audio industry headed in the next couple years?

STEVE: In the next couple of years, I believe mobile audio will continue to grow but I think the biggest explosion in growth will come from voice-controlled devices—in the home, in the car and mobile. The lines between different kinds of devices are becoming blurred as new virtual/augmented reality devices are launched, consumer electronics companies start manufacturing cars, televisions become voice-controlled, smart speakers become speakerphones, earphones double as 'hearables' and Amazon's Alexa becomes omnipresent in our lives. This presents some new and exciting challenges for audio testing, both in terms of what to measure as well as how to measure it, as audio signals transmit via the cloud as well as Bluetooth, USB, mini-USB, lightning and other interfaces. One of the real advantages about the flexible architecture of SoundCheck is that since we have been testing a wide variety of products since the early days of the company, we are already familiar with the test methodology used in these different areas, and with the addition of the algorithms that compensate for latency caused by various wireless interfaces, it's actually pretty straightforward for us to test most wireless and cloud based devices to the same standards as their wired counterparts.

SHANNON: What's next for Listen, Inc.?

STEVE: In terms of product and market development, we are continuing to develop our testing capabilities for voice control and smart speakers (and other IoT devices). In fact this year we are touring with our smart speaker testing seminar where we demonstrate how to measure smart speakers and other voice-controlled devices such as robots, thermostats, and more. We continue to add to the functionality of the software with new and improved algorithms, additional display options, and tighter software-hardware integration. We are also developing some exciting new products (which I am not ready to reveal yet) that will be launched early in 2019. From a business standpoint, this year, we plan to recruit additional staff in sales, software development, DSP, and business management to meet our increasing business needs.

Options for PCB Fabrication



Learn how to make informed choices regarding PCB finishes when you send your boards out for fabrications with this brief tutorial.

By
Scott Dorsey

Printed Circuit Board (PCB) fabrication is the assembly method for circuit boards used in electronic and computer devices. The layers of the board are put together along with the specific surface pattern so it can be used in electronics manufacturing. Often firms that fabricate PCBs also make related products for their customers. Manufacturers of electronic goods can either fabricate in-house or contract this operation out to third party specialists.

The design provides information about what the board needs to do and the environment in which it will be used. There are many of finishes and options available for PCB board manufacturing, and there's a lot of misinformation out in the audio community about them. Every day, as PC board manufacturing methods advance, we get more options, most of which are appropriate only for very specialized applications.

This is a quick introduction to many of the more common options that you will be presented with if you send boards out for fabrication, written with the expectation that you'll be doing small runs of boards for hand assembly for audio applications. Many of the statements that I make here are valid only under those conditions and may seem peculiar from someone with a large-scale manufacturing background. I have tried to explain some of the more common examples but may have missed some.

As you send boards out, the first main set of options you get is "board finish." That is, after the copper on the board has been etched and the holes plated through, something is done to ease further processing.

Back in the 1970s, you might have had an entire set of options about how you wanted the holes plated through, but after a while the manufacturing engineers figured out that most of those options didn't work (remember Tascam recorders with those riveted holes that all failed?) and so the hole plating is pretty much standardized. However, finishes aren't standardized, and I want to provide you with some of the more common choices.

HASL

Hot Air Solder Leveling (HASL, pronounced "hazzle") is a simple dip in a bath of molten solder with no additional plating, with air knives used on the hot solder to smooth the finish out. This is usually the least expensive board finish available and it's almost always the best one for small batch production.

The main downside of this is that the surfaces are not completely flat, which can be an issue for fine pitch surface mount work. Occasionally if the process is done poorly you can encounter plugged holes, which can be an issue for through-hole work. Both of these can be avoided if the factory is competent, but not all factories are.

HASL surfaces are the absolute easiest to perform rework on. Although the surface layer may be slightly uneven, it is typically pretty thick. There is no actual standard specification for thickness, but it's typically about 0.100 to 0.200 mils.

This surface is thin enough that the solder that gets dissolved into your joint is not enough to substantially change the composition of the joint. If 60/40 solder

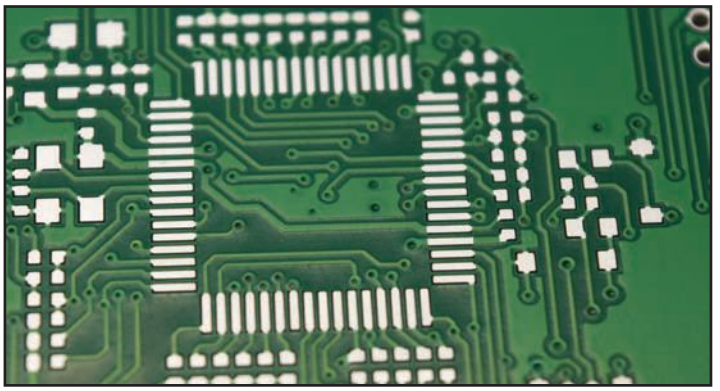


Photo 1: Immersion Tin is a non-electrolytic tin plating applied chemically over the copper surface.- (Image courtesy of Millennium Circuits Ltd.)

is used on the surface, when you install parts with 63/37 solder, so little of the 60/40 will dissolve in that your solder will remain eutectic.

Because the surface is thick, soft, and slightly irregular, HASL is not appropriate for any board with edge card connectors unless those connectors are processed separately.

Lead-Free HASL

Lead-Free HASL is an identical process except the dip is in a bath of some sort of lead-free solder, typically a tin-copper, tin-copper-nickel, or even tin-copper-germanium alloy. If you are forced by Restriction of Hazardous Substances (RoHS) requirements to use a lead-free solder, this is the way to go, but you need to make sure that the solder used on the board is compatible with the solder you are using for work.

This is normally the default process used by the Asian board fab houses, and if you are using lead solder it is workable. Using leaded solder on this material will dissolve enough into the joint to cause only moderate contamination, but the solder will not flow as well as on a leaded surface.

I strongly recommend avoiding lead-free HASL unless you are working with a lead-free solder and have made sure that the lead-free solder is compatible with the board finish. The board fab house should be able to help with this since RoHS work is sufficiently common today that they have it down to a science. Compatibility problems, however, may not show up for many years.

Immersion Tin

Immersion Tin is a non-electrolytic tin plating applied chemically over the copper surface (see **Photo 1**). It is inexpensive and lead-free. However, the tin surface is not terribly hard (approximately 0.020 to 0.015 mils thick) and it is not inert. It will corrode less easily than bare copper but is still quite delicate. Also it is prone to "tin whiskers," especially for high voltage applications, where thin crystals of tin can roll off the surface.

Immersion Tin is fine if you're going to wave solder boards, because all of the exposed area will be covered. However, if you are doing hand assembly you need to make sure the traces are covered with solder to prevent future damage.

I generally recommend avoiding Immersion Tin for small production

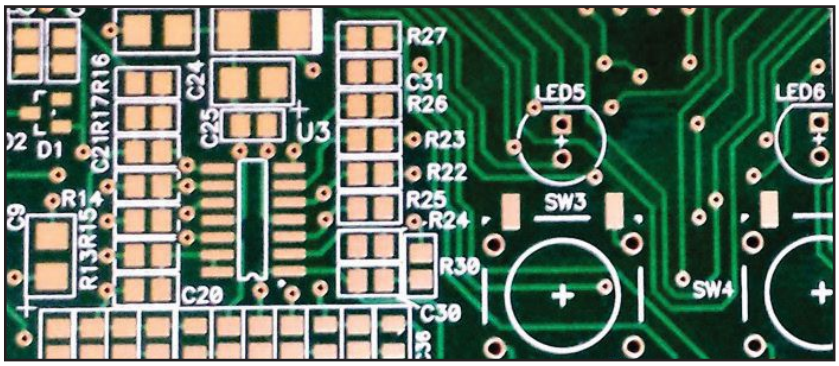


Photo 2: Electroless Nickel Immersion Gold (ENIG) involves a thick and hard coat of nickel a very thin flashing of gold on top.

work. However, if you are making your own boards, the tin plating process can be done in a tray in the kitchen and it's better than leaving bare copper finish.

Edge card connectors done with Immersion Tin will be reliable for a small number of cycles lugging in and out. I would not recommend this for any application where boards are going to be regularly changed out, the surface is much flatter than HASL and will permit edge card use.

Immersion Silver

This is the same basic concept as Immersion Tin, but with silver instead. The surface finish is even thinner (0.005 to 0.012 mils), there are no worries about tin whiskers, and it's lead free. However, if using leaded solder, you need to use a 2% silver-bearing solder on this surface to prevent the silver from leaching out around the solder joint and forming a location for corrosion to occur, unless you are wave soldering (where the traces are completely covered). Unlike with Immersion Tin, covering all traces is not essential as long as you use silver-bearing material. However, if the traces are not covered, they will discolor over time. The discoloration is not harmful but may be esthetically displeasing.

OSP

IBM came up with this concept in the 1970s. Organic Solderability Preservative (OSP) is a sort of wax that is sprayed over the bare copper board surface. The surface is very thin, only a couple thousands of a mil, but it provides fairly rugged protection for the copper surface. It harmlessly melts away when solder is applied. However, the OSP material can clog the interior of through-holes. It will rub off with a fingernail and is sensitive to rough handling but is remarkably effective over the long term for boards that are protected inside cabinets. I have never heard of anyone using OSP for home assembly work but I don't see why you couldn't. It's lead-free and metal-free in general and, therefore, eliminates any solder compatibility worries with any solder material.

ENIG

All audio enthusiasts love ENIG surfaces! Electroless Nickel Immersion Gold (ENIG) involves a thick and hard coat of nickel that is 0.100 to 0.200 mils thick with a very thin flashing of gold on top (see **Photo 2**).

If this is properly made, the gold provides a protective finish to the nickel but there isn't enough gold to contaminate solder joints and cause gold embrittlement of solder joints. Too much gold and joints fail, too little gold and the traces corrode.

The surfaces look lovely, and they have a long shelf life. You can leave boards for decades on the shelf before soldering them. The surface is hard and flat. However, the process is complex and expensive.

Running heavy currents through nickel can cause small amounts of distortion due to metal effects. This should not be an issue under normal cases since most of the current is flowing through the copper trace below it, but it's still a possible worry in high current contacts.

Using ENIG surfaces for edge card connectors is kind of tricky because the gold flashing wears off the first time you insert the card, and then you have a nickel contact. This may not actually be a bad thing, and ENIG is likely the least expensive way to get a reliable edge card connection.

I can't personally recommend ENIG for audio circuits overall; although aesthetically pleasing there are no advantages over HASL in typical designs and the added cost is substantial.

Hard Gold

Hard gold (also called electrolytic gold) provides a hard and durable surface that does not corrode. It consists of about 0.125 to 0.150 mils of nickel with 0.025 to 0.050 mils of gold. Because it has so much gold, you cannot solder to this surface because it will leach enough gold into the joint to make the solder brittle.

Hard gold is almost exclusively used for edge

About the Author

Scott Dorsey has a degree in electrical engineering, during the pursuit of which he worked in the broadcast and recording industries. After several years working at a major studio, he took a job with a defense contractor. This left him time to do live concert recording for acoustical music and to design and build audio devices for personal use and on contract to several audio manufacturers and importers. Scott is a regular contributor to several audio magazines. He has been publishing equipment reviews and DIY projects since the mid-1980s. He is probably best known in the general audio community for his retrofit electronics designs in inexpensive Oktava, AKG, and Feilo microphones.

card connectors, where it is a very good choice. Many facilities will allow you to process the board as standard HASL or tin with selective plating of the edge card connector as hard gold.

It may also be useful for other contacts, such as for large computer-grade capacitors or buss-bars, which are bolted directly onto the board. It can be useful if you have large power supply leads that are bolted onto a large hole through the board and are carrying considerable contact current.

Gold Flashing

There are some Asian facilities that can produce a gold surface that is 0.05 to 0.010 mils worth of gold on an edge card connector. This looks good but is not really appropriate for boards that will be plugged in and out. Adding proper hard gold-edge card connectors is likely to double or triple the cost of board processing but is worth every penny in the long run. I encourage you to have them hard gold plated and make absolutely sure you are getting is the real thing if you are doing boards that plug into consoles or 500 series racks or otherwise use edge card connectors.

Wire-Bondable Gold

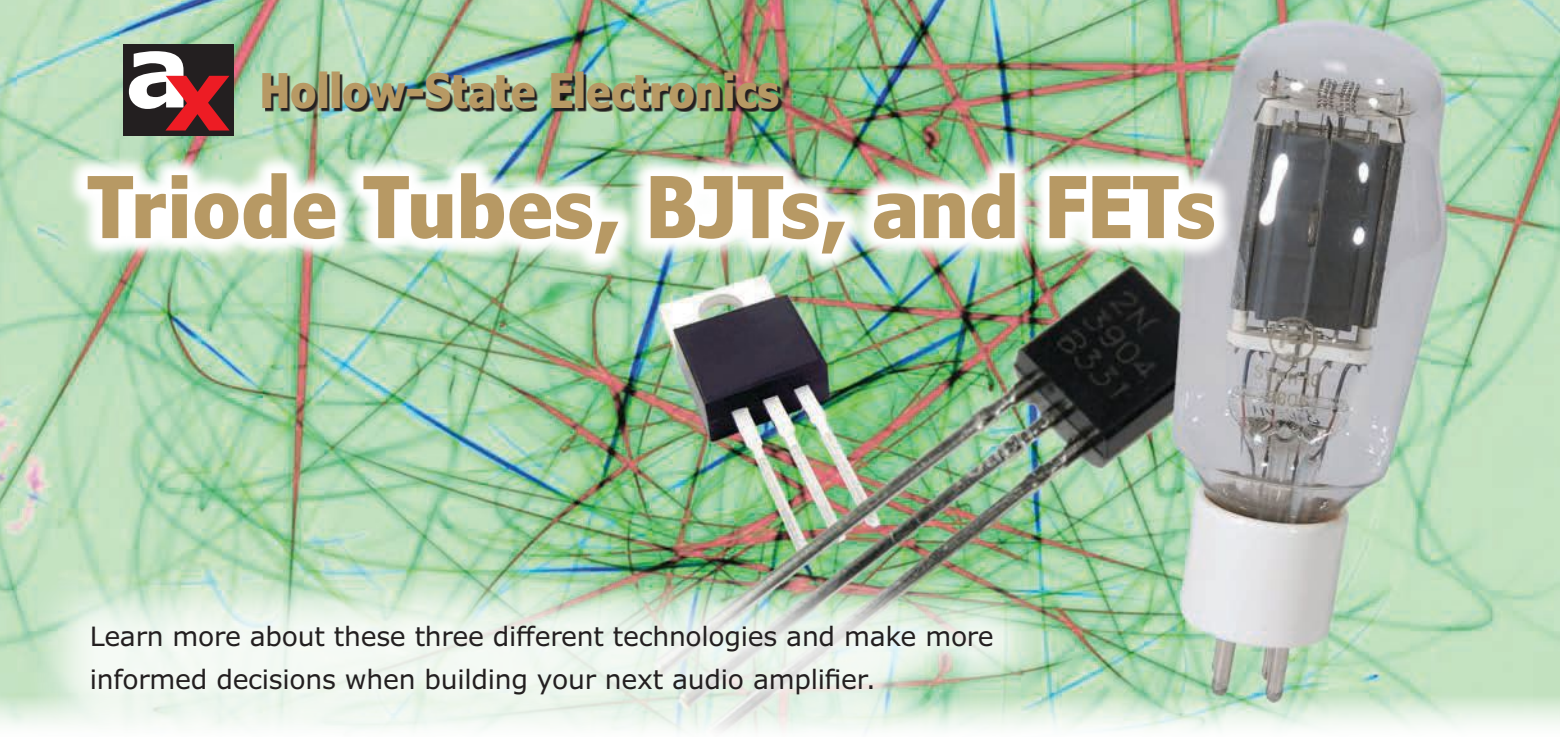
This is an interesting oddity, it's a gold finish intended to allow fine gold-plated wires to be welded to the surface of the board. It's used for thick film hybrid modules, and for chip-on-board modules where the die is directly attached to the PC board, wires tack-welded on, and a glob of epoxy dropped on top to seal the assembly.

This is actually a variant of ENIG with a much thicker gold surface, which makes soldering to it problematic. I cannot imagine any application for this in small production audio work, but likely you'll have the fab house offer it to you so you should know what it is.

Board Finishes

If you're having boards made for you by an outside factory or by any of the many wonderful services that have popped up to panelize small boards from hobbyists and have them made in bulk, it's good to know what the various options available as well as their advantages and disadvantages. The odds are that you'll be wanting simple HASL surfaces, but if not you should be familiar with some of the options.

Triode Tubes, BJTs, and FETs



Learn more about these three different technologies and make more informed decisions when building your next audio amplifier.

By
Richard Honeycutt
(United States)

The Age of Electronics began with Thomas Edison's discovery of the Edison Effect in 1875 (which was refined and patented in 1883). Most historians would date the beginning of the Age to 1904, and the invention of the Fleming Valve or vacuum diode.

As described in my February 2012 *audioXpress* Hollow-State Electronics article, "The Birth of Hollow-State Electronics," this device was used initially as a detector for AM radio. In an attempt to create an even more sensitive radio detector, Lee de Forest invented the vacuum triode, which he dubbed the Audion, in 1906. In 1926, Gilles Holst and Bernhard D.H. Tellegen invented the vacuum pentode, which could provide much higher voltage gain than a triode.

In 1926, Julius Lilienfeld proposed a solid-state device to perform the functions of a vacuum triode: the Field-Effect Transistor (FET). In 1934, Oskar Heil patented another version of the FET. Neither of these devices was commercialized at that time. Japanese engineers Jun-ichi Nishizawa and Yukio Watanabe invented the first working FET in 1950. This device used a reverse-biased PN semiconductor junction to establish an electric field in a conducting channel of N-doped silicon, making it a Junction FET (JFET). They called it a Static Induction Transistor (SIT).

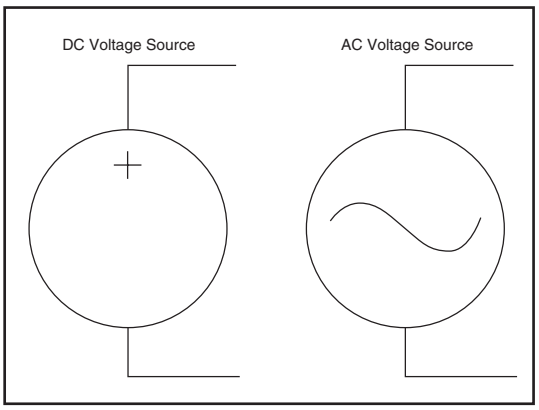


Figure 1: These symbols represent ideal voltage sources.

William Shockley, Walter Brattain, and John Bardeen at Bell Laboratories were working with operational models of the FET when they invented the point-contact transistor in December 1947, just after the Lilienfeld patent had expired. Problems with noise, reliability, and moisture sensitivity plagued the point-contact transistor. The invention of the Bipolar Junction Transistor (BJT) in 1951 overcame these issues.

William Shockley published his theory of the operation of the FET in 1952, but it was not until 1960 that materials processing technology had advanced to the point that John M. Atalla of Bell Laboratories introduced a commercially viable device. His FET used a metal-oxide layer to prevent current flow from the gate into the channel, and was called a Metal-Oxide-Semiconductor Field-Effect Transistor (MOSFET). The MOS layer kept leakage current at an insignificant level, giving a much higher input impedance than the reverse-biased PN junction of the JFET could provide.

During the late 1960s and early 1970s, JFETs became widely available, and were mostly used as small-signal devices, although at least one Japanese manufacturer produced and sold high-power SITs. MOSFETs came to be used for small-signal, logic, and power applications.

Differences Among the Valves

Fast-forward to today: We now have audio amplifiers using vacuum triodes and pentodes,

BJTs, JFETs, and MOSFETs. So what's the difference among these various "electronic valves?" Probably the best way to answer this question is to employ the controlled-source theory.

Figure 1 shows the schematic symbols for a DC and an AC ideal voltage source. These sources are called "ideal" because they provide the same voltage into any load, meaning that they have zero output resistance and impedance. Of course, no voltage source is really ideal, but the concept of an ideal voltage source is useful for analysis and conceptualization.

Figure 2 shows the schematic symbol for an ideal current source. There is no separate symbol for an ideal AC current source. The DC symbol is usually used instead, and the reader must understand from context when an ideal AC source is intended. An ideal current source produces the same current into any load. This implies that the output resistance and impedance are so much larger than the resistance/impedance of any load, that the effects of the load are negligible in determining the current. This is often thought of as meaning that the output resistance/impedance of an ideal current source is infinite. If you think too deeply about this, you are bound to wonder how any current at all can pass through an infinite output resistance. Discussing this quandary further leads into rabbit chases that have nothing to do with our topic, so let's just agree that the output resistance/impedance of an ideal current source is very high.

Controlled Sources

If a voltage or current source is a battery or electrical power source, the voltage or current it produces is determined by its construction. But it is useful to introduce the concept of a controlled source (or dependent source) whose voltage or current is determined by a voltage applied to an input terminal. There are four types of controlled sources:

- A voltage-controlled voltage source (VCVS) is a voltage source whose output is determined by a voltage applied to the control input.
- A current-controlled voltage source (CCVS) is a voltage source whose output is determined by a current applied to the control input.
- A voltage-controlled current source (VCCS) is a current source whose output is determined by a voltage applied at the control input.
- A current-controlled current source (CCCS) is a current source whose output is determined by a current applied at the control input.

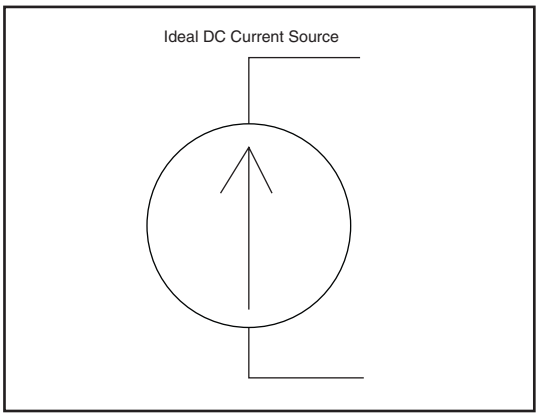


Figure 2: This symbol represents an ideal DC current source.

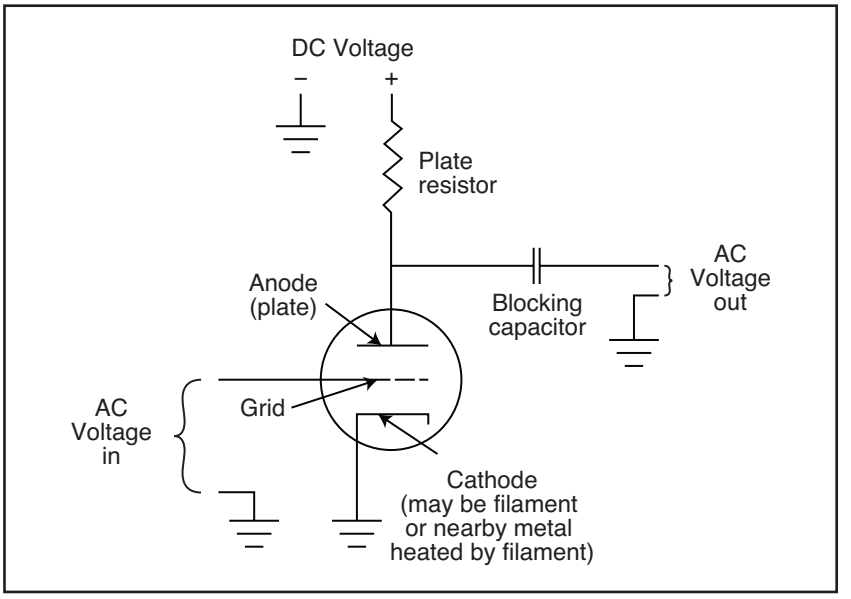


Figure 3: This is the schematic of a basic vacuum triode amplifier stage.

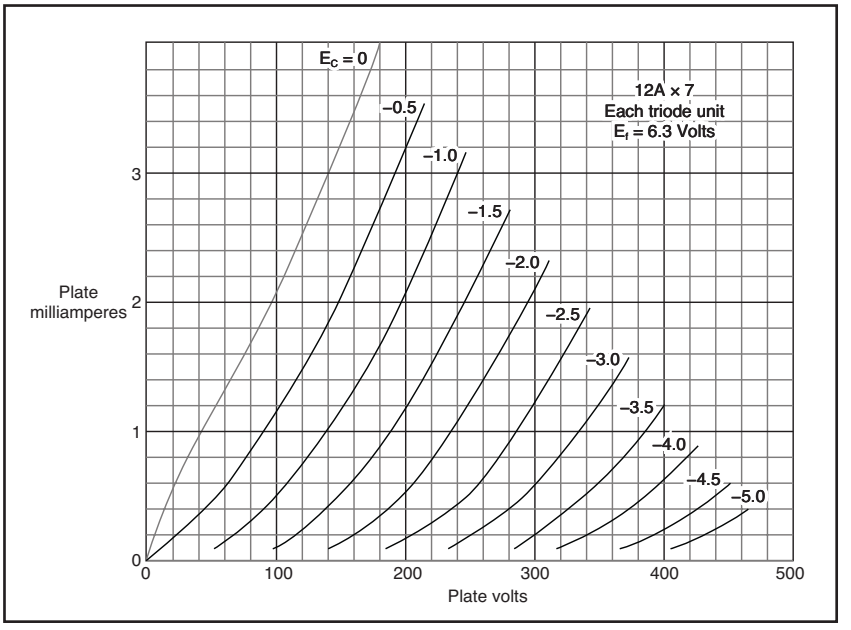


Figure 4: The characteristic curves for one triode of the 12AX7 show that the plate current depends upon both the grid-to-cathode voltage E_c and the plate voltage.

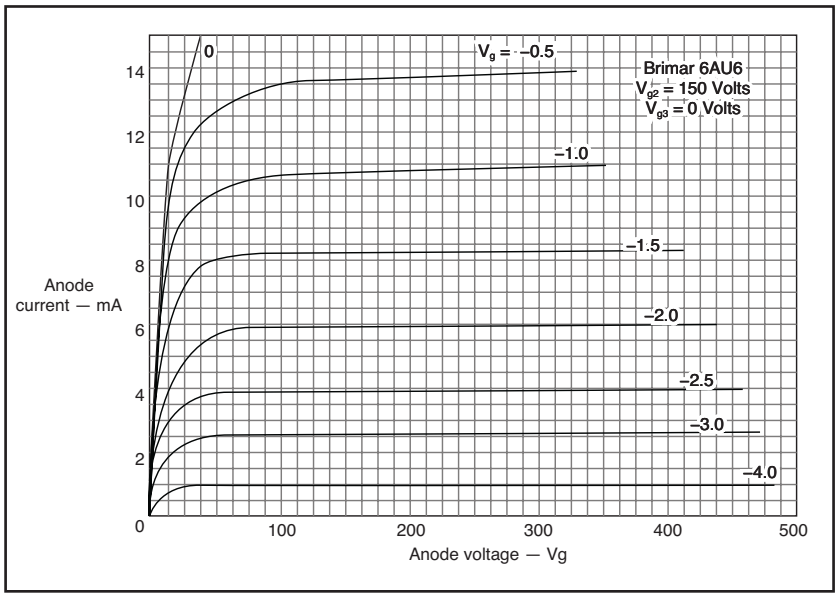


Figure 5: The characteristic curves for a 6AU6 vacuum pentode show little dependence of plate current upon plate voltage.

Actual Functions

With these idealized elements in mind, let's look at how tubes, BJTs, and FETs operate. **Figure 3** shows a schematic of a vacuum triode connected as an amplifier. The hot cathode frees electrons from its surface by a process called thermionic emission. These electrons form a cloud, or "space charge" near the cathode. Since there is a DC voltage on the plate, some of the space-charge electrons are pulled toward the plate. The control grid is a screen, mesh, or coil of fine wire that is usually biased with a voltage that is negative with respect to the cathode, repelling electrons back toward the space charge region. Depending upon the plate and grid voltages, a certain fraction of the space-charge electrons do make it through the grid to the plate. Thus, electron current flows from the cathode, through the grid, to the plate, through the plate resistor, to the plate voltage supply, and then to ground and back to the cathode, completing the circuit. The plate current is controlled by the grid-to-cathode voltage, making a vacuum triode act as a VCCS.

Actually, a vacuum triode is not a very ideal current source, since its plate current depends not only upon grid-to-cathode voltage, but also upon plate voltage, as shown in the characteristic curves of **Figure 4**. The instantaneous plate voltage is the product of the plate current and the plate resistor value, subtracted from the supply (B+) voltage. Thus in operation, a triode's plate current is affected by its instantaneous plate voltage.

This behavior is characterized by the plate resistance of the triode, which is specified as about 60 to 80 kΩ for a 12AX7, but also depends upon the grid-to-cathode voltage and plate current. If the triode acted as an ideal current source, the plate resistance would be much higher. (A triode is not an ideal voltage source either; that would require a plate resistance of zero ohms. But we usually speak of a triode as a VCCS.)

Vacuum pentodes behave more nearly like ideal current sources, since an effect of the screen grid is to make the plate current less dependent upon plate voltage. **Figure 5** illustrates the plate characteristics of a 6AU6 vacuum pentode. Notice that over most of the operating range (plate voltage above about 100 V), the plate current does not vary much as plate voltage changes. This is reflected in the higher plate resistance of 0.5 to 1.5 MΩ. The pentode is also modeled as a VCCS.

An FET (JFET or MOSFET) has electrons injected into the "source" terminal of a "channel" of N-type silicon. The electrons flow through the channel, attracted by a positive voltage applied at the "drain terminal" (other end from where the electrons are injected). The effective width of the channel—and thus, the effective drain-to-source resistance—is controlled by the strength of a negative electric field resulting from a negative voltage at the control ("gate") terminal. This voltage itself produces very little current (in a JFET) or essentially no current (in a MOSFET), but exerts its control on the drain current

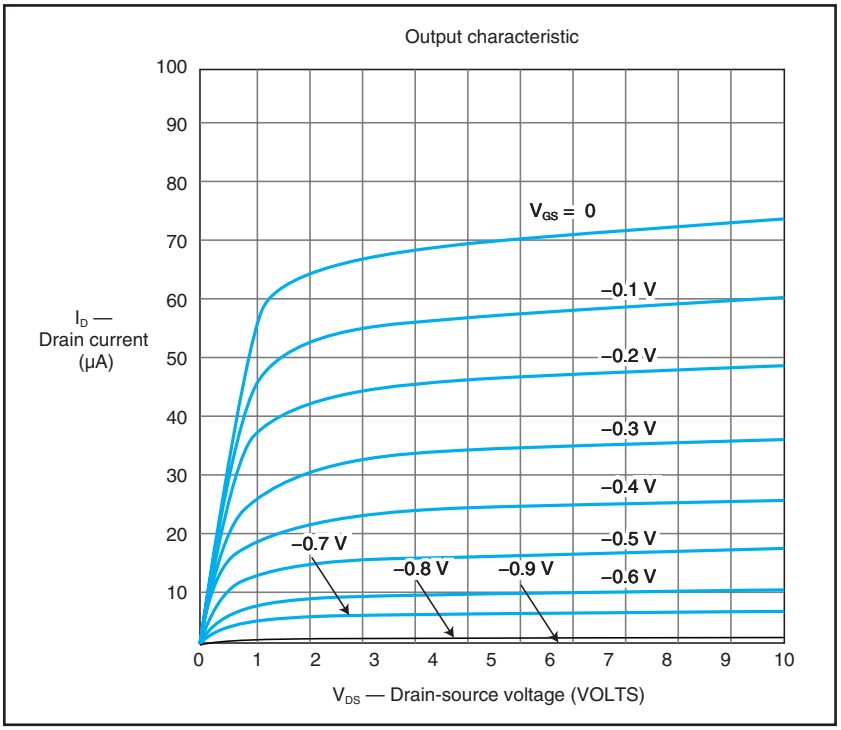


Figure 6: The drain characteristic curves for a 2N4117 JFET are very similar in shape to the plate curves of a vacuum pentode.

only via the electric field. (This description applies to an N-channel JFET, or NJFET. There are also PJFETs which operate with opposite polarity. The charge carriers in a PJFET are "holes" rather than electrons.)

Figure 6 shows that the JFET performs similarly to a vacuum pentode: It is modeled as a VCCS that is more nearly ideal than a vacuum triode is. The curves for a MOSFET are pretty much the same, except that in JFETs, as in vacuum tubes, the control voltage is usually negative. For FETs, this behavior is referred to as depletion mode. MOSFETs are available in enhancement mode (sometimes called enhancement/depletion mode) as well. In this mode, some drain current flows when the gate-to-source voltage is zero, drain current increases as the gate is biased more positively, and drain current decreases as the gate is biased more negatively. The effect on the drain characteristic curves is just that the individual curves would be labeled differently: (e. g., the curve labeled V_{GS} = -3 V in Figure 6 could be the V_{GS} = 0 curve for an enhancement mode MOSFET.

A BJT is a different animal. It is made of a "semiconductor sandwich" (usually silicon, but germanium, gallium arsenide, and other semiconductors can be used). The "bread" in the sandwich will be either N-type semiconductor (for an NPN transistor) or P-type semiconductor (for a PNP transistor). The "cheese" in the sandwich is the opposite type of semiconductor. Remember that if a PN junction is biased with the P material positive with respect to the N material, the junction is said to be forward-biased, and current flows. The forward-bias voltage required to establish conduction depends upon the semiconductor material and the level of P- and N-type impurities.

Silicon transistor junctions typically require about 0.6 V to make forward current flow. Almost no current flows when the junction is reverse-biased. In N-type material, the majority of electric charge carriers are electrons. In P-type materials, the majority current carriers are orbitals in the P-type crystals' atoms that have apparent "holes," or energy levels that would be occupied by electrons in a crystal of pure silicon. Current in this P-type material occurs as electrons "hop-scotch" from hole to hole. So the majority carriers in P-type material are holes.

If we consider an NPN transistor, external dc voltages are connected so as to make the electrons enter at the emitter (N-type). The base (P-type) has a small positive voltage, so that a small current flows through the emitter/base junction. The collector (other N-type section) is biased more positively than the base, causing the collector/base junction to be reverse-biased.

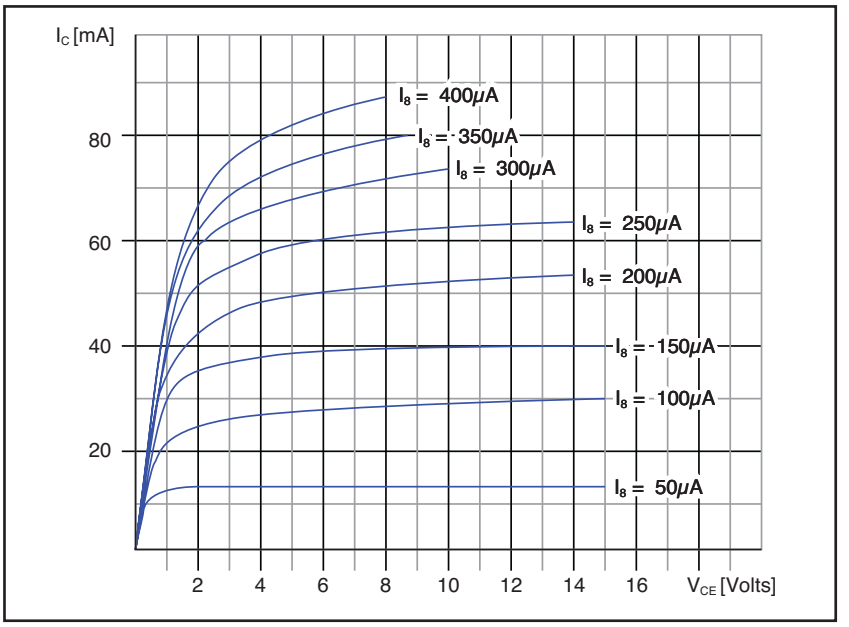


Figure 7: The collector characteristic curves for a BJT are also similar in appearance to those for a vacuum pentode.

The base region has few impurities and the base region is very thin, so the current in it is limited. Most of the electrons approaching the emitter-base junction from the emitter do not recombine with holes and create base current. Instead, they proceed—as minority carriers—through the base, being attracted by the positively-biased collector region, and thus cross the reverse-biased base/collector junction.

This happens because the reverse-biased junction prevents the flow of recombination current made up of majority carriers, but does not block the flow of minority carriers; namely, the electrons in the base region. Therefore, most of the emitter current becomes collector current. Controlling the base current allows us to control the collector current.

Figure 7 shows the collector characteristic curves of a common low-noise small-signal NPN silicon BJT. Like a vacuum pentode and an FET, these curves show little effect of voltage on current. The BJT is modeled as a CCCS, since it is actually the base current that controls the collector current. However, since base current is affected by the base-to-emitter voltage, some engineers choose to consider a BJT as a VCCS.

Three Different Technologies

So we have three very different technologies for building audio amplifiers. It is commonly believed that "tube amps sound good," "BJT amps sound bad," and "FET amps sound better than BJT amps, but not as good as tube amps." Another thing that "everybody knows" is that the triodes' distortion signature has proportionately more even harmonics than does the BJTs', and FETs are somewhere in between. In a future article, we'll look at the output waveforms of each type of amplifier, using a sine-wave input, and see just what the differences look like. Then we can also investigate the intermodulation (IM) distortion performance across a range of output voltages.

Industry Calendar

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

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

Mobile World Congress is the world's largest gathering for the mobile industry, organized by the GSMA and held in the Mobile World Capital, Barcelona. The event will once again take place at Fira Gran Via with selected events, including 4YFN, taking place at Fira Montjuïc. With more than 108,000 attendees and 2,300 exhibitors representing 208 countries. It is an event not to be missed. Mobile World Congress 2018 will showcase the latest in mobile technology, from 5G to artificial intelligence, connected cars to virtual reality, drones to robots, and everything in between.

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

electronica China is the leading trade fair for the electronics industry in Asia. It's the industry's most important gathering and brings the future to the world, a place that stands out as best for business promotions and the latest trends in the electronics industry. electronica China is now one of the leading platforms for electronic components, systems and applications. The fair has been one of the most important events in Asia since 2002. In 2017, electronica China saw 68,215 attendees, an increase of 11% over 2016, and 1,230 exhibitors, which was an increase of 13%.

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

Audio Salon de Montréal celebrates 31 years. The Audio Salon de Montréal attracts more than 5,000 visitors, who wish to listen, see and touch more than 300 audio product brands under one roof. The largest gathering of manufacturers, distributors and retailers in the audio-video industry in Canada. The 2018 edition has already received confirmations from more than 100 exhibitors, all eager to share their passion for music, by demonstrating some of the world's best sound systems from $3,000 to $1,000,000.

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

CanJam Global is the world's premier series of headphone audio shows and is produced by Head-Fi.org, the world's largest audio community. Since 2006, CanJam events have been a community showcase for Head-Fi.org, and an industry platform that enables headphone audio manufacturers, distributors, and dealers interact with their customers, peers, and the audio, lifestyle, and technology press.

In March CanJam Global returns to the Pan Pacific Hotel in Marina Square for it's third edition of CanJam Singapore. Featuring more than 100 leading brands in over 15,000 ft² of exhibition space, CanJam Singapore 2018 will showcase the latest products and innovations in headphones and personal audio electronics in all price ranges—from $50 earbuds to the world's finest headphone audio systems at $50,000.

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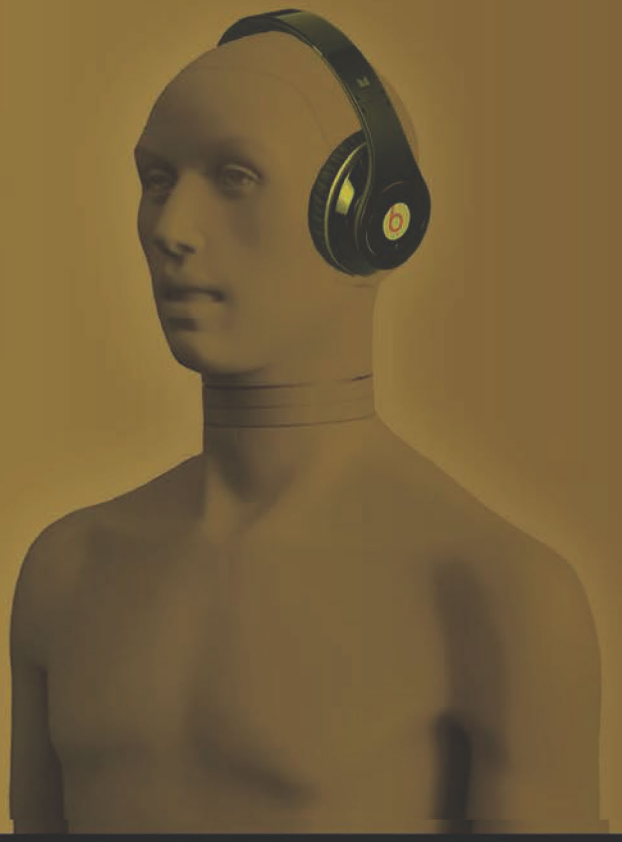

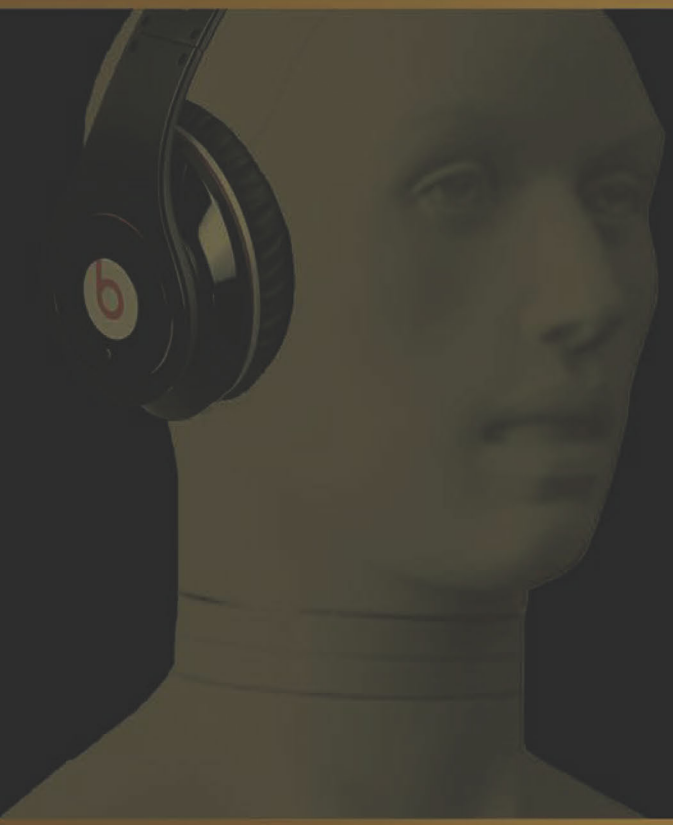

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