INNOVATIONS IN AUDIO • AUDIO ELECTRONICS • THE BEST IN DIY AUDIO

www.audioxpress.com

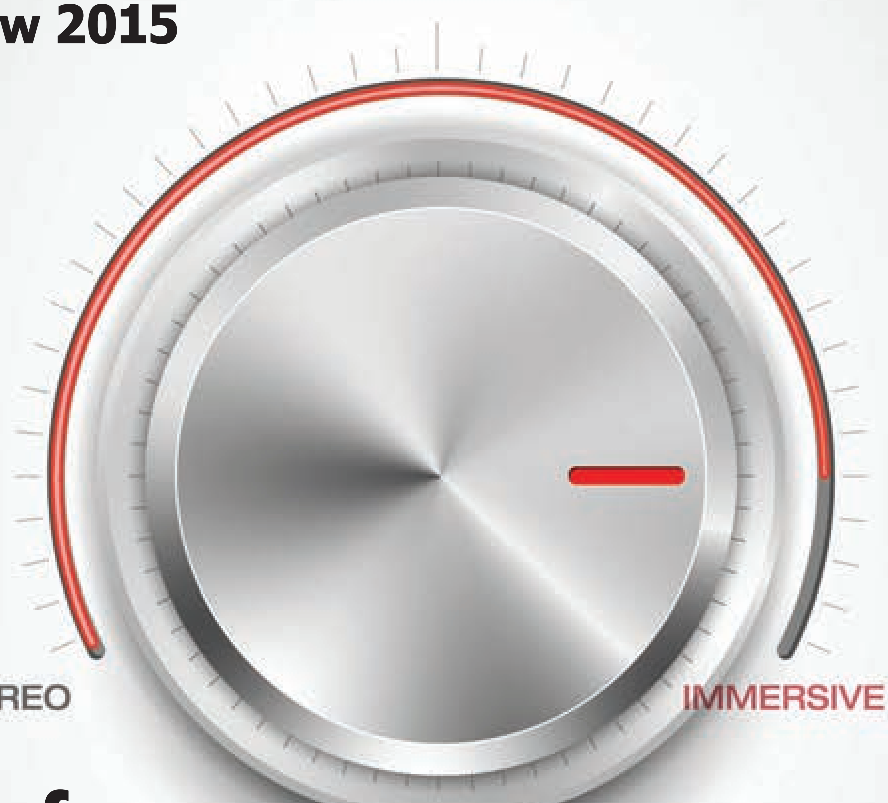
audioxpress

ADVANCING THE EVOLUTION OF AUDIO TECHNOLOGY

Show Review
Munich High End Show 2015
By Ward Maas

Audio Praxis
Upmixing – Stereo to Surround
By Jon Schorah

Headsets
In-Ear Monitor Design Challenges
By Mike Klasco and Steve Tatarunis



Standards Review
Considerations for the MPEG-H Audio Standard
By João Martins and Robert Bleidt

Sound Control
The EASERA Electro-Acoustical Measurement Platform
By Richard Honeycutt

Audio Electronics
The Right Filter (Part 3) Closer to the Ideal Crossover
By Vincent Thiernesse

Questions & Answers
Interview with Kvart & Bølge Founder Bjørn Johannesen

$9.00US $10.00CAN

AUGUST 2015


HAMMOND
MANUFACTURING®
MANUFACTURING & DESIGNING
TRANSFORMERS & ENCLOSURES
SINCE 1917
Chassis, Plates & Covers
Tube Output Transformers
Plate & Filament Transformers
Open & Enclosed Chokes
USA
Tel: (716) 630-7030
Fax: (716) 630-7042
Canada
Tel: (519) 822-2960
Fax: (519) 822-0715
UK
Tel: 01256 812812
Fax: 01256 332249
Australia
Tel: 8-8240-2244
Fax: 8-8240-2255
www.hammondmfg.com

August 2015 **ISSN 1548-0628**

www.audioxpress.com

audioxpress (US ISSN 1548-0628) is published monthly, at $50 per year for the US, at $65 per year for Canada, and at $75 per year Foreign/ROW, by Segment LLC, Hugo Van haecke, publisher, at 111 Founders Plaza, Suite 904, East Hartford, CT 06108, US Periodical Postage paid at East Hartford, CT, and additional offices.

Head Office:
Segment LLC
111 Founders Plaza, Suite 904
East Hartford, CT 06108, US
Phone: 860-289-0800
Fax: 860-461-0450

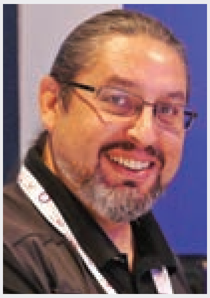
Subscription Management:
audioxpress
P.O. Box 462256
Escondido, CA 92046, US
Phone: 800-269-6301
E-mail: audioxpress@pcspublink.com
Internet: www.audioxpress.com

Postmaster: send address changes to:
audioxpress
P.O. Box 462256
Escondido, CA 92046, US

US Advertising:
Strategic Media Marketing, Inc.
2 Main Street
Gloucester, MA 01930, US
Phone: 978-281-7708
Fax: 978-281-7706
E-mail: audioxpress@smmarketing.us

Advertising rates and terms available on request.

Editorial Inquiries:
Send editorial correspondence and manuscripts to:
audioxpress, Editorial Department
111 Founders Plaza, Suite 904
East Hartford, CT 06108, US
E-mail: editor@audioxpress.com



The Promise of Better Audio Experiences

In 2014, *audioXpress* selected immersive audio as the year's most important technology. That year Dolby Atmos actually gained momentum—both in commercial cinemas and residential installations—while making an entrance into the consumer market. Also, some of the most impressive demos at the Consumer Electronics Show (CES), Integrated Systems Europe (ISE), InfoComm, Custom Electronic Design & Installation Association (CEDIA) Expo, the National Association of Broadcasters (NAB) Show, and IBC (among others) included some sort of immersive audio experience.

That's the reason why, when planning our 2015 editorial calendar, we decided to dedicate this issue to the topic. We selected the term "spatial audio" because it is one of the generic designations that currently bridges interactive applications in gaming, future digital television formats, and effectively, all sorts of enhanced reproduction formats currently being promoted for headphones, such as binaural and some proprietary algorithms, from DTS's Headphone:X to the Fraunhofer Institute, Cingo algorithm. We also could have called it "3-D audio."

The fundamental differences between "immersive," "spatial," and "3-D" sound are not that important. The most important new terminology in this domain is that of "object-based" audio in contrast to channel-based audio formats.

Object-based audio (also referred to as granular or particle concepts) converts mixing decisions to metadata. It is why Dolby grabbed the attention of all major Hollywood studios so quickly, and why, in the production communities, everyone adopted the concept. Basically many of the sound for film studios and post-production facilities already have increasing amounts of sound work for computer games. That's how they became familiar with creating virtual spaces using convolution reverb algorithms—separated from the audio sources (tracks)—so that the acoustics space could be "rendered" in sync with the image each gamer sees in every moment of the game.

From that "virtual space" concept, the next step would be to change the traditional audio mixing techniques and the way panoramic information is distributed for two, five, seven, or more channels in a "fixed" way. With current automated mixing techniques, it became possible to think about sounds independently of channels, instead thinking about objects manipulated in "virtual spaces."

But basically, even gamers use traditional stereo, 2.1, or 5.1 speaker systems, so all the work of converting audio "objects" into metadata, which could be rendered in real time by the game consoles, couldn't achieve that much impact. The same happened with traditional home-theater setups.

Enter Dolby Atmos, Auro-3D, and other solutions and the idea of placing speakers in the ceiling or up-firing drivers, creating another level of space around the gamer/spectator, which could finally be considered 3-D or immersive audio. Problem is, there is no guaranteed way to ensure that object-based audio metadata is efficiently distributed in consumer media and that reproduction from different formats/sources can be correctly translated.

DTS unveiled details about DTS:X—its own object-based, immersive audio technology—that is in direct competition with Dolby Atmos. However, it promises extended levels of interoperability, including with binaural headphone experiences. The foundation of DTS:X is Multi-Dimensional Audio (MDA)—an open platform for creation of object-based immersive audio that was submitted to the Society of Motion Picture and Television Engineers (SMPTE) for standardization.

Unfortunately, DTS has not disclosed any more information and the only area we could focus on was the new MPEG-H audio emerging standard, a codec already proposed as an open format currently being considered for adoption in the US as part of the new ATSC 3.0 television standard.

I am certain by the year's end we will hear about more fascinating developments in this area and we look forward to CES 2016, when all those technology proposals and standards are expected to be in place.

João Martins
Editor-in-Chief

OUR NETWORK

audioxpress | VOICE COIL | circuit cellar | LIS Loudspeaker Industry Sourcebook

SUPPORTING COMPANIES

NOT A SUPPORTING COMPANY YET?

Contact Peter Wostrel (audioxpress@smmarketing.us, Phone 978-281-7708, Fax 978-281-7706) to reserve your own space for the next edition of our magazine.

COLUMNISTS

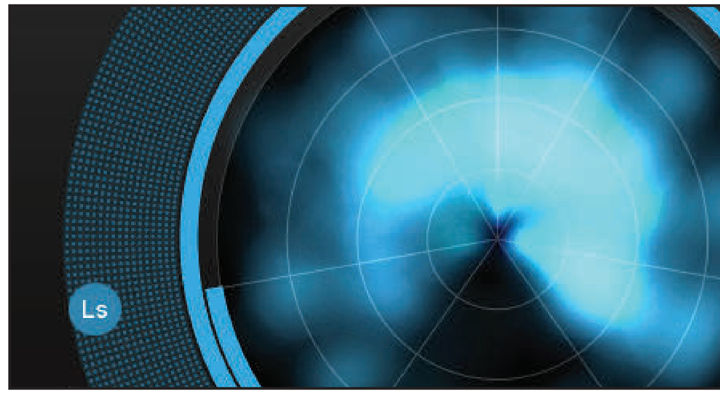
Vance Dickason has been working as a professional in the loudspeaker industry since 1974. He is the author of *Loudspeaker Design Cookbook*—which is now in its seventh edition and published in English, French, German, Dutch, Italian, Spanish, and Portuguese—and *The Loudspeaker Recipes*. Vance is the 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.

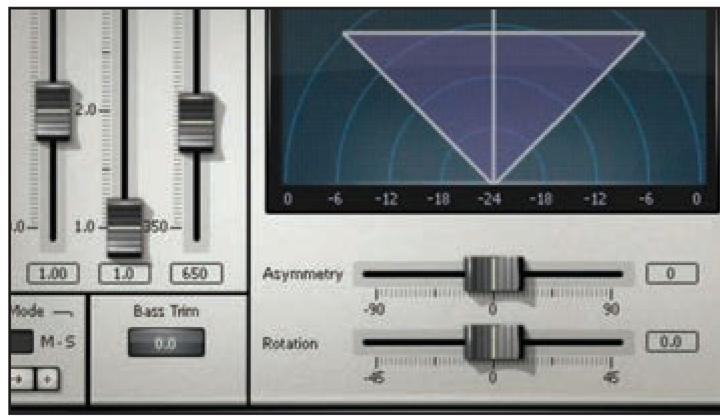
Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject while Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical 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.

Contents





Features

Volume 46 – No. 8 August 2015

Columns

Departments

Websites

audioxpress.com
voicecoilmagazine.com
cc-webshop.com

@audioxp_editor audioxpresscommunity

Book Review

Exploring *Vanderveen Trans Tube Amplifiers*

What should a reader expect from *Vanderveen Trans Tube Amplifiers* by Menno van der Veen? Elektor.com describes it as an "adventure story" recounting van der Veen's discoveries during his research into transconductance (trans) tube amplifiers, and implies the book will present designs "suitable for hobbyist construction." The back cover declares that in *Trans Tube Amplifiers*, van der Veen describes how—while researching transconductance amplifier design—he developed the "Transie 1" and "Transie 2," which are both sweet-sounding, DC-coupled single-ended amplifiers, and eventually "struck gold" with the Vanderveen Trans 30.

Review By
Bill Reeve
(United States)

Before we crack open *Vanderveen Trans Tube Amplifiers* and decide for ourselves, let's talk a little about Menno van der Veen's background and his accomplishments to date. More than 30 years ago, van der Veen combined the technical background he gained from his university degree in physics and electronics with his love of high-fidelity audio to develop high-performance toroidal output transformers for vacuum tube amplifiers. These have been available from several manufacturers for many years, and they truly are of the highest quality—both in manufacture and performance.

He has studied, developed, and built vacuum tube audio amplifiers for even longer, and has published hundreds of papers on the subject. Probably his best-known work is his book, *Modern High End Valve Amplifiers* (Elektor, 1999). In 1986, van der Veen formed his research and development consulting company, and in 2006 he established the Dutch TubeSociety school to teach audio amplifier design. Clearly, van der Veen has the knowledge and experience to be an authority on performance improvements to vacuum tube (and hybrid solid-state/vacuum tube) audio amplifiers.

His latest book, *Vanderveen Trans Tube Amplifiers* is a crisp, silver and grey paperback, about a quarter-of-an-inch thick, with a beautifully constructed tube amplifier's top deck artfully displayed on its cover.

The book delivers in many areas. It is engagingly written, and generally easy to read. *Vanderveen Trans Tube Amplifiers* moves quickly through an explanation of the author's specific transimpedance amplifier application to his simple "Transie 1" and "Transie 2" low-power, single-ended common-cathode audio amplifiers that are easy to understand, and relatively simple to build. These amplifier designs clearly illustrate the implementation of the transimpedance concept.

By driving a current (rather than voltage) signal into the local feedback loop around an amplifier's output power tube, and thus lowering the amplifier's output impedance, van der Veen effectively applies high-gain local feedback around the output power

Vanderveen Trans Tube Amplifiers
By Menno van der Veen
Publication Date: March 2015
Format: Soft cover
Pages: 108
ISBN: 978-1-907920-34-9
Elektor: www.elektor.com/books/audio

About the Reviewer

Bill Reeve is a technical program manager at Google in Mountain View, CA. He has graduate engineering degrees from the Colorado School of Mines and the University of California, Berkeley. He also earned his Master's degree in Electrical Engineering from Santa Clara University.

tube to reduce distortion and improves low-frequency performance (both in the tube and the output transformer that it is driving), all without the stability concerns associated with a global feedback loop.

On the downside, van der Veen's initial explanation of what he means by a transconductance amplifier is too succinct for most people to understand as they read the text. I had to get out a pencil and paper and re-derive an inverting amplifier's output impedance, and then replace the input resistor with a voltage-controlled-current source to understand how the current source's high-output impedance reduces the amplifier's output impedance by increasing the loop gain. From there, it was a simple step to model and derive the feedback equation for a common-cathode tube amplifier driven in the same way.

As a suggestion for a second edition, replacing the existing appendices (which seem only tangentially related to the book's technical theme) with some step-by-step derivations of the transconductance amplifier theory as well as some analyses of the transconductance designs presented in the book would further empower readers to apply the transconductance concept to their own original designs.

Just a technical nit: the back cover states, "...transconductance...means converting current into voltage or voltage into current." This isn't true. Transconductance is the conversion of voltage into current. Transimpedance is the conversion of current into voltage. Current flowing through an impedance develops a voltage. A resistor (which represents the real—that is, phase independent, part of impedance) is the simplest transimpedance amplifier. Conductance is the inverse of resistance, and transconductance amplifiers (which convert voltage into current) are active devices.

The literature and history of vacuum tube audio amplifiers is extensive, and multifaceted, and *Vanderveen Trans Tube Amplifiers* is a worthy addition to that body of work.

Quite often, modern articles and books on high-performance audio amplifiers are exercises in egotism and elitism, but van der Veen avoids these pitfalls. His presentation is factual and humble. In addition to transfer functions, he includes many harmonic distortion and linearity measurement results, and these quantitative evaluations of amplifier performance are greatly appreciated. His distortion vs. frequency plots are difficult to interpret because they were probably originally generated in color but printed in black and white. However, they get the point across. His subjective evaluations of the tonal quality, dynamic range, and clarity of his amplifier designs are...well... subjective, but that cannot be avoided. However, van der Veen's objectivity is strengthened by his criticism of—based upon his own subjective listening to—one of his own designs. I was glad he included the example of an amplifier he developed using the "trans" concept that did not sound good. That unsuccessful experiment—which put too much high-gain, solid-state circuitry directly in the audio sign path—will save other experimenters time and guide their creative developments.

Consistent with the literature (and actually tube amplifiers themselves), this book's topics and designs are esoteric. The techniques presented for lowering tube-driven output transformer distortion solve a problem that most commercial manufacturers avoid by using solid-state power drivers—but those drivers are not for everyone. The beautiful sound, the signal path simplicity, and the challenge of designing and building our own audio amplifiers pull us toward the emotional comfort of glowing vacuum tubes. Menno van der Veen's books and papers present new ideas and new implementations of old ideas to provoke creativity and stimulate experimentation.

In summary, *Vanderveen Trans Tube Amplifiers* contains technical content worthy of a paper, or a chapter in a larger book—but van der Veen expanded the material to include valuable descriptions of multiple amplifier designs that actually are "suitable for hobbyist construction" and the experimental paths that were dead ends (to save others from having to learn those lessons themselves). This expansion to book format is valuable for a few reasons. First, books typically reach a larger audience than technical papers. Second, the broader exploration of successful, and unsuccessful, amplifier designs makes the trans topology much more accessible to amateur experimenters, designers, and constructors.

Vanderveen Trans Tube Amplifiers is a worthy read, and it provides provocative ideas that readers can experiment with and expand upon to create their own sweet-sounding vacuum tube amplifiers.

In-Ear Monitor Design Challenges

An Introduction

This article provides a brief introduction to in-ear monitor (IEM) earphones and details the key components. The second article in this series will discuss some of the manufacturers and detail the current status of armature driver design.

By
Mike Klasco and Steve Tatarunis
(United States)

The practice of using a two-way system (woofer and tweeter) to provide full-range sound has been around since Fred Flintstone (his rock speaker is still commonly used in outdoor gardens). Bass reproduction requires a relatively heavy and large voice coil, a cone, and a surround along with some decent excursion capability. Higher frequencies need a small diaphragm for wide even coverage of a room. A light voice coil and a diaphragm ensure the moving mass is low enough to reach the uppermost range.

At first glance, the two-way approach might seem overly complicated for earphones and headphones, as the headphone driver is small enough that dispersion is not an issue. Since the driver faces into a duct (the ear canal), the low-end response remains flat below resonance. Yet as the market demand for high resolution wideband sound and higher acoustic output with low distortion increases, the benefits of woofers and tweeters in earphones and headphones has become more appealing to the design engineers.

Today, there are many in-ear monitor (IEM) earphones that are two-way or three-way designs, or more. Until now two-way headphones have been an obscure category but several innovative two-way headphones are reaching the mass market and we will discuss some of them in the next few months. With the "high-resolution" craze, there is also a growing expectation for headphones with ultra-wide-bandwidth and achieving extended top-end response to 50 kHz and beyond is a lot easier with a two-way configuration.

IEM History

The evolution of today's multi-way earphones really began with IEMs for on-stage use. During the 1980s, there was a steady escalation of on-stage concert sound levels that often resulted in microphone feedback. Just the leakage from the stage monitors getting into the microphone pickup degraded audio quality and the band members could not hear each other, which did not help their overall performances. By the 1990s, many famous musicians such as Pete Townsend were suffering from hearing damage and inner ear balance problems. Some musicians had to use hearing aids and hope for partial hearing recovery. This led to the development of in-ear monitors, which initially contained modified hearing aid components.

In 1995, Jerry Harvey, a touring monitor engineer, worked with drummer Alex Van Halen because he had difficulty hearing his other band members over all the stage noise. Harvey created a custom-molded earpiece for Van Halen to block some of the stage noise. The device was intended to deliver a more isolated monitor mix directly to his ears. This was the first legitimate IEM prototype earpiece. It contained two tiny speaker drivers, one for low frequencies and one for high frequencies. The frequencies were split by a passive crossover. The high-frequency driver was a stock Japanese component but a suitable low-frequency driver was difficult to find. The only driver that survived strong kick drum signals was a Knowles-made pacemaker part, which was a balanced armature transducer intended to warn the pacemaker wearer of internal problems.

Other Van Halen band members became interested, additional earpieces were crafted, and the business known as Harvey's Ultimate Ears was born. Since IEMs are essentially high-performance hearing aids, Ultimate Ears originally partnered with Westone, a custom ear molder for hearing aid manufacturers, to produce Harvey's designs. After a few years, the popularity of iPods and other MP3 players prompted a consumer model IEM for audiophiles. Offshore manufacturing followed and Altec Lansing and others launched licensed and rebranded consumer versions of the Ultimate Ears two-way IEMs. Logitech acquired Ultimate Ears in August 2008 and multi-way armature earphones have been in the upper echelon of mainstream earphones ever since.

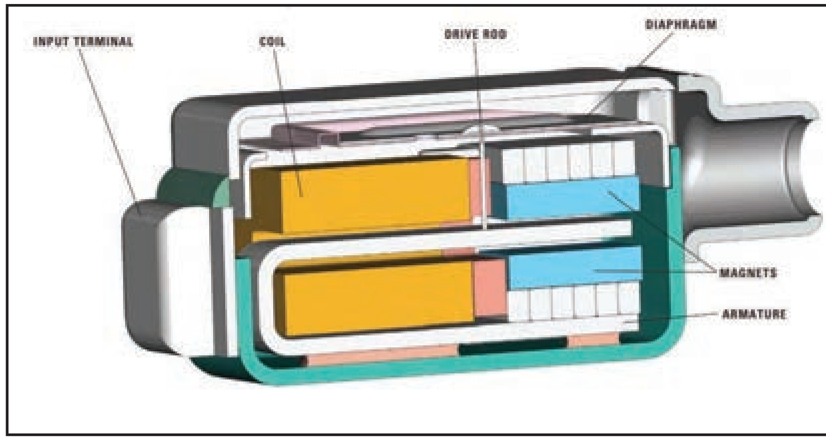


Photo 1: This armature driver cutaway details the inner workings of an in-ear monitor. (Photo courtesy of Knowles Electronics)

Inside an IEM

An IEM can have a single full-range driver, but most devices typically consist of at least one bass

Photo 2: A new brand, Campfire Audio by ALO Audio, has just released its Lyra armature-driver earphone. (Photo courtesy of Campfire Audio by ALO Audio)

driver and one “tweeter.” For the top end, armature drivers are preferred. Next month, we will examine the armature driver’s inner workings, but for now we will just provide a brief overview.

A balanced armature driver employs a voice coil similar to a dynamic driver. However, unlike the coil in the dynamic driver, the coil in the balanced armature transducer is in a fixed position as is the magnet. The moving part is a long arm (the armature) that is mounted in the coil’s center. The armature moves inside the coil essentially like a pendulum and a drive rod connects the armature to the diaphragm (see **Photo 1**).

Armature drivers were first used in hearing aids due to their small size and form factor, high sensitivity, and clear sound quality. These are the same reasons that make armature drivers attractive for IEM earphones. This armature assembly is integrated into a tiny rectangular casing with terminals and a sound port that feeds through ducts to the eartip.

IEM System Design Challenges

IEM multi-way drivers can be constructed in many ways. While the top-end response is handled by an armature, the bass might be two armatures or a double armature (two armatures in the same case). There are also some IEMs that are ”hybrids,” which combine armature drivers with a small moving-coil driver to handle the bass.

Dividing frequency ranges between drivers is achieved by using a passive crossover network. One of the biggest challenges is getting the drivers in time alignment so there is not a cancellation notch due to differing phase at the crossover frequency.

Specmanship is the name of the game with IEMs. Dual armatures, three, four, five, and more crossover points are gaining popularity. Some audio engineers might question the rationale of so many drivers and the point of diminishing returns (i.e., the results getting worse rather than better). Trying to get all the transducers to line up and have the same acoustic center—and still have the unit fit into the ear and not stick out or be too heavy—takes talent and patience and maybe even a little bit of luck!

Article co-author Mike Klasco spoke with Ken Ball, owner and IEM designer at the boutique brand Campfire by ALO. Ball and Klasco agreed that while cramming as many drivers as possible into a shell is all the rage now, more sometimes does not mean better. The additional crossovers and time alignment configurations create more problems and cause more complicated issues. If it is not done just right, the results will end up sounding too colored and unnatural. Despite marketing hoopla to the contrary, there is still something to be said for a thoughtful, elegant, and simple IEM design (see **Photo 2**).

IEM Earshell Construction

On a commodity earphone, you would expect a cheap ABS injection-molded plastic shell that tends to add a buzzy “aftertaste” to the sound. Higher-grade and less-resonant types of construction for the body include glass-filed nylon, wood nano-fiber composites, cast ceramics, machined wood, and metal-injection molding. At the top-end are CNC-machined bodies. Ball said when he puts put the same receiver system into a plastic shell and into a CNC-aluminum shell, the aluminum shell always sounds better and exhibits a perceived larger sound than its polycarbonate twin. Perhaps this is real or it may be due to some psycho-acoustical effect.

Photo 3: Since 1959, Westone has specialized in custom earplugs and earpieces as well as other products to protect and enhance hearing and facilitate communication. (Photo courtesy of Westone)

The Eartip

One of the original IEM hallmarks was custom eartips matched to a mold made from the user's ears. This was critical because a precise fit with custom-molded silicon eartips would ensure high passive attenuation. Performers needed to block the intense sound levels on stage so they could hear the monitor mix in their IEMs. Bass reproduction also benefits from a tight seal on the eartips. Custom eartips require a skilled audiologist to take the mold and prepare the eartips.

There has been some recent development work on 3-D scanners used in smartphones to create the eartip contour. People can email the scans as 3-D CAD files to the tip manufacturer and custom eartips are delivered to their doors in a week or two. For standard eartips, there are several different materials and configurations (e.g., foam, silicon, thermoplastic elastomers, along with stabilizer wings, ear-hooks, etc.). Usually three sizes are included, which covers most ears but sometimes comfort might take a back seat to absolute passive sound attenuation (see **Photo 3**).

Ball observed that the universal (non-custom mold) ear-tip is sorely underdeveloped and many improvements have yet to be realized. The fit is the IEM's most critical part. Regardless of how your IEM is measured, if you do not have good fit, then your $1,200 IEM will sound like a $50 IEM and probably will not be comfortable (see **Photo 4**).

Photo 4: Comply earphone tips are designed to provide comfortable, secure in-ear fit, and noise isolation. (Photo courtesy of Comply Foam)

The Cable

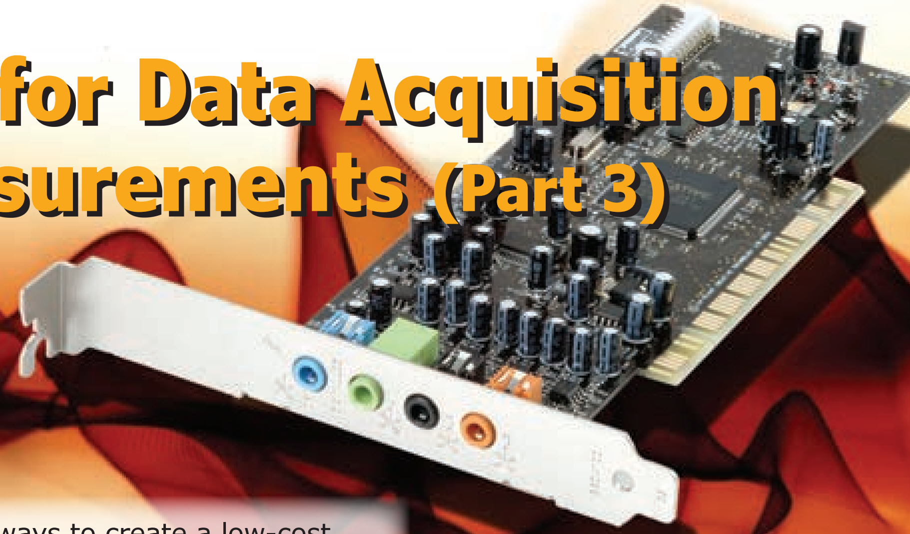
While not every audio engineer agrees on the speaker cable's contribution to the overall loudspeaker performance, earphone cables are a different story. There are a half-dozen considerations for earphone cables that are not relevant for speaker wire. For example, for earphone cables to be flexible, they must be thin and woven. Thin and woven cables can translate to high direct current resistance (DCR), parasitic capacitance, and inductance that impact the frequency response. Cables with non-tangling cross-sections also are a factor in cable design, damping aramid fibers interwoven with the conductors increase break resistant strength and reduce microphonics (i.e., the noise of the cable brushing against your shirt and transmitting through the cable to the shell). Connectors are not just for plugging into your smartphone or earphone amplifier. They can also connect to the shells. If the cable breaks, you can simply use a replacement cable without sending the whole earphone to be repaired (see **Photo 5**).

Coming Soon

Initially, there were only a couple of Scandinavian and US armature driver manufacturers. Interestingly, these incumbents have relocated their production facilities so many times that a small army of "armature engineering refugees," who were left behind after each factory closure, have established their own small armature production facilities in Asia. Not all of these newcomers are copy cats, as several Japanese companies have developed innovative variants of armature drivers. Next month, we'll explore some of the longtime manufacturers and some of the new companies, and examine the status of armature driver design.

Photo 5: High-end replacement cables can cost in excess of $500. (Photo courtesy of ALO Audio)

Sound Cards for Data Acquisition in Audio Measurements (Part 3)



Selecting the Right Hardware

In this article series, we are examining ways to create a low-cost system for lab-grade audio electronics measurements. Thus far, I hope the first two articles in this series have whet your interest in inexpensively extending your measurement capabilities. The pieces of a powerful modern measurement system comprise a sound card, software, and a signal conditioning interface. Next, you must choose the right computer.

By
Stuart Yaniger
(United States)

This month, we will consider the bigger part of the hardware equation: the computer itself. The options for Apple computers in audio measurement are limited. Despite Apple's popularity for music production, nearly all the measurement software is designed for Windows computers. Over the years, I've used about half-a-dozen different computers with different Windows-based operating systems and have come to several conclusions.

Windows-Based Systems

XP, although commercially a dead issue, seems to be favored by people who have a good knowledge of operating systems. For those of us who don't, it's easy to spend hours wrestling with different menus to get the sound from where we want to generate it to where we want to measure it without molesting it too badly. As an OS, the support by different software and sound card packages is uneven. I don't recommend it. At the moment, I have Windows 7 running on two different laptops and a desktop, and it's been easy for even a computer novice such as me to get it to do what I want. Reputedly, Windows 8.1 operates nearly identically and is probably easier to find in a new computer. Windows 10 will be released shortly (as of the time of this writing) and it will be interesting to see if it makes the audio amateur's experience easier or more difficult.

Audio Stream Input/Output (ASIO) drivers go a long way in helping you obtain top performance. They bypass the Windows layers between the sound card and the measurement software. Ensure that whatever software/hardware you use takes advantage of this!

Have a Separate Computer

A dedicated computer is a good idea. Having no interrupts or interference from wireless or antivirus or other unnecessary programs running in the background can save a lot of measurement frustration. If your computer needs to be used for other tasks, it may help to set up a dedicated boot profile which will disable any wireless or LAN adapters, and mail or antivirus programs on start-up.

If you're using a laptop, the battery is your friend. Much effort in reducing noise and ground-loops can be avoided if the computer has no connection to the mains. Get a good charge then unplug your power brick.

Avoid Time Delays

"Latency" is a big deal for music production. Think of it as a time delay while the computer is

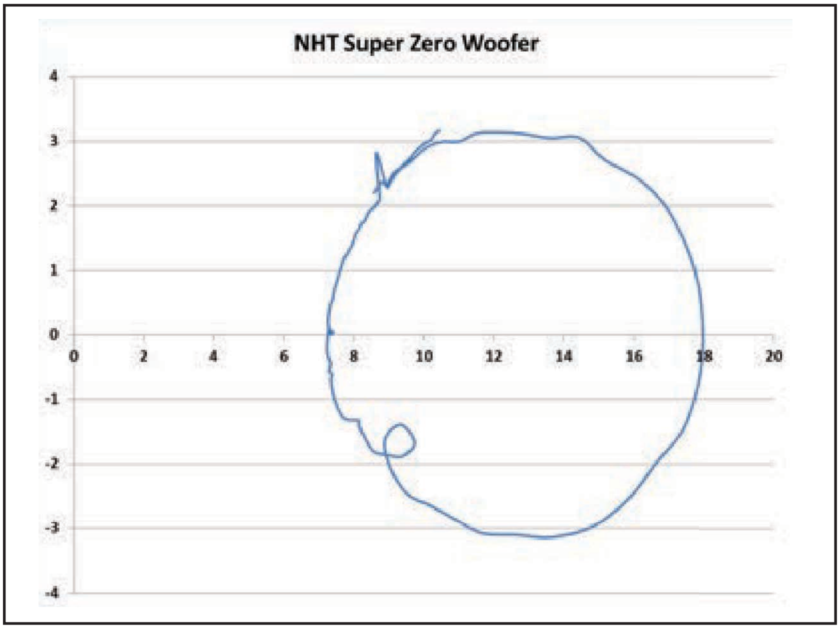


Figure 1: For this Heyser plot of complex impedance, I obtained the data using AudioTester, then exported it into Excel for plotting. The vertical axis is reactance (imaginary) in ohms, the horizontal axis is resistance (real) in ohms.

passing information from one place to another. That can certainly disrupt timing in overdubbing, for example. It's a function of the computer's speed and memory, as well as the specific software and hardware package used. It is generally not a big issue in most audio measurements, but if you have a measurement where synchronous operation is critical, you should at least be aware of it. There are various freeware latency checkers—I have been happy with LatencyMon.

Software Options

Choice of software is likewise broad, with a wide range of trade-offs between flexibility, user-friendliness, and cost. As before, this isn't a survey of all the available sound card-compatible software, but rather a sample of a few options that are representative of what's out there. Full reviews of each of these could fill entire articles (and perhaps will!), but I'll give you some brief impressions after spending some time with each of these particular packages.

Rightmark Audio Analyzer (RMAA) is a popular freeware package, originally designed to benchmark sound card performance. It has a limited range of test signals, limited flexibility, and is not easy to customize. However, for minimal effort and expense, the user can perform a predetermined test suite—noise, total harmonic distortion (THD), intermodulation (IM), swept THD and IM, and crosstalk—in a few seconds with the press of a button. Because of its popularity, there's lots of support on various audio and home theater forums.

RMAA freeware can be upgraded to a pro version that includes a mono mode (highly useful for those tests of circuits where only one channel is built up), variable Fast Fourier Transform (FFT) size, ASIO compatibility, and better control of display and plotting parameters. With the current favorable dollar/ruble exchange rate, the Pro version can be bought for less than $30. All in all, the freeware version is a nice choice for someone wanting to get started in audio electronics measurement. The trade-off for simplicity and ease of use is limited flexibility, and the inability to use ASIO drivers (which can lead to some wrestling with Windows MME to get good results). Any measurement of loudspeakers is pretty much off the table.

AudioTester is a much more versatile package, encompassing impedance, distortion, noise, speaker parameters, frequency response, impulse response, and spectral decay. It is easy to set up and use (though more complex than RMAA) and has good versatility. The various versions have had different degrees of bugginess, so if you're trying to do a measurement that just won't cooperate, you might try using a different version (the legacy versions are available on AudioTester's website).

True to Murphy's Law, in the midst of writing this article, I updated to the latest build and it is buggier on my system than the previous version, making data acquisition a real chore. No doubt this will be fixed in the next build, but it can be frustrating. My advice is that if you get it working well, don't be tempted to update! When it works, though, it works well, and the low price (about $47 at the time of this writing) is certainly attractive!

ARTA is probably the gold standard for measurement software packages, and seems to be the most popular option. It is actually a bundle of three programs: ARTA, which is an FFT-based spectrum analyzer; STEPS, which as the name implies uses frequency-stepped sine waves to perform measurements; and LIMP, which is a package directed toward loudspeaker measurement and has capabilities for impedance, frequency response, impulse response, and polar pattern. It's quite a bit more expensive than AudioTester (about $90 at the time of this writing), but I haven't actually screamed at my computer when using it. It only took an hour or two to figure out the features I needed for my basic measurements, and it has run smoothly thereafter. Given the large base of satisfied users, it's hard to go wrong with this choice.

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.

Virtins Multi-Instrument sports an amazing range of capabilities. Of all the software packages I tried for this article, it has the most features, the widest capability, the greatest flexibility, and the steepest learning curve. It is configured as a group of virtual instruments (thus the name), including a signal generator, a spectrum analyzer, a digital oscilloscope, and a multimeter. Add-on modules include a data logger and a LCR meter, as well as 3-D graphing capability.

One of the interesting features is the library of windowing functions (which I will discuss in next month's article) in .wav format so that you can customize or add your own window if you're so inclined. In the month or so that I've had the Virtins package, I don't think I've exploited even a tenth of its capability. The price for the standard edition is about the same as ARTA ($99.95). It is a tough choice but I bet that a year from now, after I've gotten a greater comfort level, this will be my go-to choice for software.

All of these general measurement packages (except RMAA) are capable of measuring both electronics and loudspeakers. Some might prefer software dedicated to loudspeaker measurement, so I'll mention a couple of those. For speaker and room measurement freeware, the outstanding examples are Room Equalization Wizard (REW) and HOLM Impulse.

REW is a remarkable piece of software, with acoustic measurement capability that includes (besides the usual frequency response and

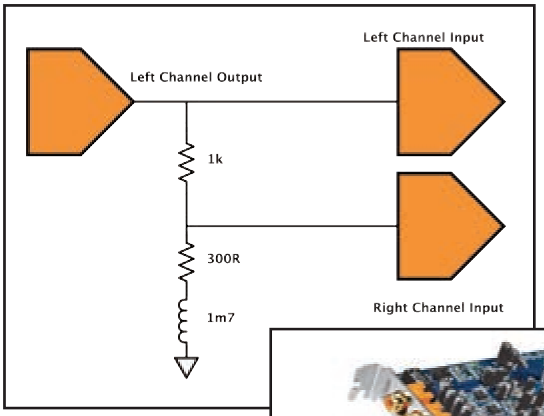


Figure 2: I used this test setup using the M-Audio 192 PCI sound card to measure complex impedance.

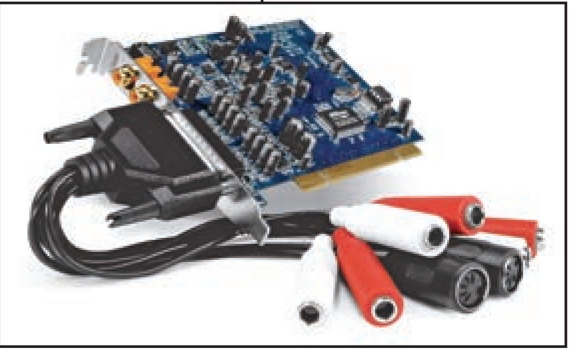


Photo 1: The M-Audio 192 PCI sound card features high-definition 192 kHz sampling rate, digital I/O, balanced analog I/O, and high-quality signal-to-noise ratio.

impedance) waterfall plots, spectrograms, polar plots, real-time analysis, SPL metering with selectable weighting, equalization, distortion, phase measurement and unwrapping. Basically, it has everything you need to perform comprehensive loudspeaker and room acoustics measurements. There is an active support forum available to help you past the rough patches and assist you in getting the most out of this remarkable software. Did I mention that REW is free?

HOLM Impulse is a nifty piece of speaker measurement freeware capable of measuring frequency response using frequency swept sin waves, chirp signals (basically a sine wave with a rapid change in frequency), or MLS (a method equivalent to impulse response, but with better signal-to-noise ratio). While not as versatile as REW, it is exceptionally easy to set up and use. If I were approaching speaker measurement for the first time, I'd start here, then move on to a more sophisticated package as my needs and abilities warranted.

Anyone who wants to obtain and sensibly interpret useful loudspeaker measurements is strongly urged to read Joe D'Appolito's superb book *Testing Loudspeakers* (Audio Amateur Press, 1998). An overview was given a few years ago in *audioXpress* and is available online. Likewise, Texas Instruments has helpfully outlined the basics of speaker measurement, featuring REW, and this review is also available free online.

Exporting Data

Most of the software packages enable users to export the data into a .txt file, suitable for importing into spreadsheet programs (e.g., Excel). Once the data are in Excel, you have a lot of flexibility in manipulating and graphing them. As an example, an interesting way of plotting impedance data is the Heyser plot. Heyser plots were a regular feature in speaker reviews in the late, lamented *Audio* magazine (examples are shown in figures 2 and 3 of the review reprint, see Resources), and I wish that other reviewers had continued this useful tradition. The impedance, usually of a loudspeaker, comprises a real (resistive) and imaginary (reactive) part.

If you plot the points of a frequency sweep on the complex impedance plane (resistance along the x axis, reactance along the y axis), you get a series of loops, each representing a resonance. In a Heyser plot, the large loops represent fundamental resonances (e.g., the bass resonances and crossover resonances). The small loops generally arise from things such as panel resonances, resonances in cavities inside cabinets, acoustic reflections, and flexure of cones. It's a nice way of visualizing impedance data, and Excel will allow you to use your acquired data to do this with just a few minutes of spreadsheet manipulation.

After acquiring the complex impedance data, exporting to Excel, and plotting on the complex plane, I made Heyser plots for a woofer test (see **Figure 1**). The woofer in this example came from an NHT Super Zero loudspeaker and was mounted in a sealed cardboard test box. The large loop in this plot is the bass resonance (F_c) and the little pigtail loop results from the flexure of the cardboard. **Figure 2** shows the test setup I used to acquire the complex impedance data. **Photo 1** shows the actual M-Audio 192 PCI Sound Card. I'll go into more detail about how to set up and perform these measurements in a later article of this series.

In next month's installment, I'll discuss the interface between the sound card and the device or system being tested.

Resources

J. D'Appolito, *Testing Loudspeakers,* Audio Amateur Press, 1998.

———, "Testing Loudspeakers: Which Measurements Matter, Part 1," *audioXpress* 2009, http://audioxpress.com/files/dappolito2959.pdf.

———, "Testing Loudspeakers: Which Measurements Matter, Part 1,"*audioXpress* 2009, http://audioxpress.com/files/dappolito2960.pdf.

R. Heyser, "Klipschorn Loudspeaker," *Audio*, November 1986, https://community.klipsch.com/forums/storage/3/1037187/Richard%20C%20Heyser%20KHorn%20Review.pdf.

T. Pate, "Audio Characterization Primer," Texas Instruments Application Report SLAA641, 2014, www.ti.com/lit/an/slaa641/slaa641.pdf.

Home Theater Shack, REW Forum, www.hometheatershack.com/forums/rew-forum.

Sources

Audio Measurement Analysis Software
ARTA | www.artalabs.hr

Rightmark Audio Analyzer (RMAA)
Audio RightMark | audio.rightmark.org/index_new.shtml

AudioTester
Audio Tester | www.audiotester.de

HOLM Impulse
HOLM Acoustics | www.holmacoustics.com/holmimpulse.php

LatencyMon
Resplendence Software Projects | www.resplendence.com/latencymon

Room Equalization Wizard (REW)
Room EQ Wizard | www.roomeqwizard.com

Audio Stream I/O (ASIO) drivers
Steinberg Gmbh | www.steinberg.net.

Multi-Instrument
Virtins Technology | www.virtins.com/multi-instrument.shtml


audioXpress
2014 Digital Archive
Just when you thought it couldn't get any easier...you can now access a full year of *audioXpress* from any device connected to the Internet! (2014: 12 issues)
You get all the benefits of a printed copy—bookmark pages, make annotations, and write in the margins—combined with the digital advantages of easy storage, zoom, links, and search features.
Only $25
Item #: AX-DA-2014
audioXpress archive CDs
The CD includes all 12 issues for a given year of *audioXpress* in PDF format.
2014
BACK ISSUES
2013
BACK ISSUES
Order yours today
cc-webshop.com

Enhanced Stereo (Part 2)

Exploring Enhanced Stereo Solutions

The goal of stereo enhancement is to artificially create sound-stage widening by "processing" the input. Continuing this series on realistic sound reproduction, this month we look at some additional Enhanced Stereo solutions.

By
Ron Tipton
(United States)

While searching for enhanced stereo solutions, I found some interesting VST plug-ins and even a hardware IC solution. I have included information on them and provided some feedback after testing them.

Stereoizer Pro

Stereoizer Pro from Hbasm is a free and low-cost Virtual Studio Technology (VST) plug-in. The free version seems better suited for turning a mono file into stereo. My evaluation of it in Audio Studio showed minimal benefit could be achieved with an already good stereo file.

However, the $19 version adds several controls that make it more useful (see **Photo 1**). I have included three short MP3 clips from Hbasm's website in the Supplementary Material with names starting with hbasm. Each clip is about 23 seconds long. The first half is unprocessed and then the plug-in is on during the second half. The documentation is minimal and no presets are supplied so I experimented using lady-S-4dB.wav (the stereo file attenuated 4 dB) as the input. I heard noticeable widening with these control settings: treble cut at −6 dB, bass cut at −6 dB, rate at 3 Hz, depth at 0.8, pre-delay at 0.7, feedback at 0.5, and the right-side vertical slider at the mid-point between green and red. The output file, lady-hbasm.wav, is included in the supplemental material.

Photo 1: The Stereoizer Pro licensed version activates all the controls and shows the setting number while the control is being adjusted.

S1 Stereo Imager

This $49 plug-in from Waves, Inc., is actually a package of three plug-ins: the Imager, the Shuffler, and the Converter (see **Photo 2**). I find that licensing and installing a Waves product is a bother but the results are worth the effort. The installation process builds a Window's tree seven-folders deep. To get your VST host to find the plug-ins, you must find the four dll files (for either 32- or 64-bit Windows) and copy them into your VST folder, usually C:\program files\vstplugins. After I did this, all my VST hosts found and properly loaded them.

First, I evaluated the Imager using lady-stereo.wav as the input. This is a simplified plug-in without the Shuffling and the Frequency sliders shown in **Photo 2** (more on these controls later). The 24-page manual is comprehensive but I didn't read it, I experimented instead: playing the input file, moving the sliders, and listening. When I liked the result, I ran the plug-in and saved the output file as lady-waves.wav, which is included in the Supplementary Material. My settings were: gain at −4 dB to prevent output clipping; width at 3, which is the maximum; and asymmetry and rotation at zero. The output file sounds at least as good as any of those I accomplished with the free plug-ins.

The Shuffling plug-in was next—but what is shuffling? The manual says, "shuffling increases the stereo width at bass frequencies to help compensate for the fact that the ears hear stereo effects as being more narrow than in the treble. Subjectively, it has the effect of making stereo images more spacious."

The other added slider "controls the frequency below which the shuffling effect is increased. This frequency may be adjusted between 350 and 1,400 Hz… a frequency between 600 and 700 Hz usually sounds best…" Note that these are not the "bass" frequencies I described in a previously published *audioXpress* article "Enhance Bass (Part 1): The Basics," as not aiding the stereo image. These are really low mid-range frequencies and are well above the Schroeder frequency.

I again used lady-stereo.wav as the input and listened as I adjusted the sliders. I finally settled on these settings: gain at −4 dB, width at 2.5, shuffle at 2, and frequency at 650 Hz. These may not be optimum but the output file, lady-shuffle.wav, is probably the best sounding stereo width enhancement I have achieved with the various plug-ins.

The Converter plug-in is for converting a left-right input to a mid-side output, or vice versa. These operations weren't applicable to this month's topic so I didn't evaluate the plug-in.

PSP Stereo Enhancer

The Stereo Enhancer plug-in is included in the PSP Stereo Pack ($49) with three other plug-ins: the Pseudo Stereo, the Stereo Controller, and the Stereo Analyzer (see **Photo 3**). The downloaded files will run in full-featured demo mode for 14 days. An operation manual is included with the pack, which is both interesting and helpful.

Often, documentation does not include the theory of operation but this manual says the Stereo Enhancer uses a comb filter to increase the differential signal content that is heard as a widening of the stereo image. The filter frequency

Photo 2: Although the name is odd, the Shuffling feature in the Waves S1 plug-in adds a touch of realism that I did not hear with the other plug-ins I evaluated.

is set using the Freq slider control. The filter resonates at all frequencies that are multiples of the set frequency so it is basically a stereo "width" control because the lower the set frequency, the wider the image. But a very low frequency, such as 20 Hz, introduces enough time delay that the right and left channels are heard as two separate sounds. A setting between 80 and 250 Hz usually produces the most pleasing result because our enjoyment of the increased width also depends on the music's primary frequency content.

In Mode 1, the other control sliders are labeled Depth and Width. The Depth slider sets the amount of processed signal present in the output, that is, the mix. A zero setting means none of the processed signal is used and 100% is processed signal only.

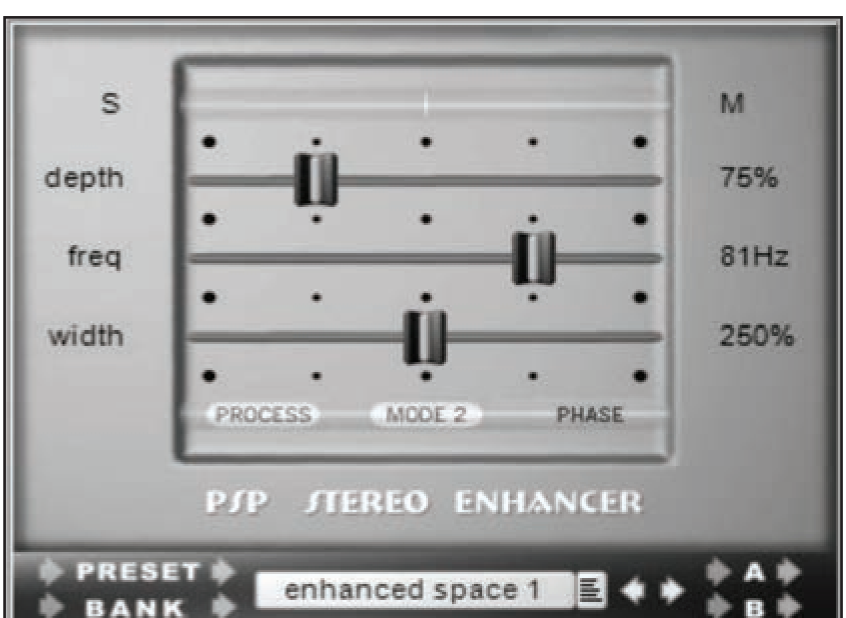


Photo 3: The 29 presets for various rooms and musical instruments in the PSP Stereo Enhancer attest to its versatility. I think it could be useful to liven up individual tracks before mixing.

A setting of about 50% is a good starting place.

The Width slider controls the amount of the differential signal from the comb filter that is applied to the input. The useful range can vary widely, from 0 to 400% in looking at the presets.

There are 29 presets selectable with the button near the bottom of the screen. During my testing, I tried several of them. I again used the "lady" file, specifically the lady-S-5dB.wav file, which is the stereo file attenuated by 5 dB. After trying various presets, I settled on depth at 50%, frequency at 252 Hz, and width at 250%. The attenuated input was needed because of the gain produced by the processing. You will find the processed file clip in the Supplementary Material as lady-psp-enhanced.wav.

I didn't evaluate the other plug-ins in the pack, but you may find them interesting. They are fully described in the operation manual, included in the Supplementary Material.

QSound

QSound Technologies has developed several products for sound enhancement. One of them is QXpander Stereo Expansion, which "creates a more enjoyable listening experience from existing stereo source material by increasing the width, immersiveness—and thus realism—of the sound

Project Files

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

Resources

Hbasm, www.hbasm.com.

QSound Technologies, www.qsound.com.

R. Tipton, "Enhanced Stereo (Part 1): The Basics," *audioXpress*, July 2015.

Sources

Stereoizer Pro VST plug-in
Hbasm | hbasm.com/stereoizer.html

PSP Stereo Enhancer
PSP Audioware | www.pspaudioware.com/plugins

SP-TFB-2 in-ear binaural microphone
Sound Professionals | Sound Professionals.com/cgi-bin/gold/item/sp-tfb-2

Tascam DR-05 portable handheld recorder
TASCAM | tascam.com/product/dr-05

S1 Stereo Imager
Waves Audio, Ltd. | www.waves.com

Photo 4: The front panel of the author's demo stereo enhancer, using a QSound QX2130S IC, is shown here. The stereo input and output connectors and the 24 VDC power supply connector are on the rear panel.

stage." This product is available as software written in C++ for a variety of computer platforms. It is also implemented as a 22-pin SDIP IC, the QX2130S. This IC is apparently no longer available from QSound Technologies but as of this writing you can find them online. I have included a copy of the nine-page pdf datasheet–application note in the Supplementary Material. File touroflondon.mp3 is a demo from QSound Technologies' website created with QXpander. The company does have some suggestions for listening to the demo. The company says you should be centered between the stereo speakers at a distance equal to or greater than the distance between the speakers. I heard very good expansion with a center-to-center speaker separation of 24" at a listening distance of about 6'.

Photo 4 shows the demo unit I built using the QX2130S. It has a nominal unity gain and can be placed between the output of your phono preamp or CD player and the input to your stereo power amplifier. The single control is a four-position rotary switch for Bypass, Wide, Wide2, and Wide3—the stereo stage width. With a common speaker separation of 8' to 10', Wide produces a pleasing expansion but Wide2 and Wide3 add little or nothing to the stage width. Their effects are more noticeable as you move the speakers closer together. Complete construction details, including a circuit board layout, are included in the Supplementary Material in QX2130S.zip.

I put several of the test files on a CD so I could play them through the QX box. I tried connecting the box's output to the sound card's line input, but there was no widening of the recorded music. So I tried a pair of Sound Professionals binaural microphones in my ears, recording the stereo output on my Tascam DR-05 digital recorder. These are Head Related Transfer Function (HRTF) recordings so listen with headphones. I left the switching transients in the two clips with "qx" in their names to make it easy to hear the change from Bypass to Wide.

I was sitting only a few feet from the loudspeakers, so there is some variation in the sound because it was difficult to keep my head perfectly still. It would have been better to have used a HRTF head on a tripod so I designed one around a styrofoam head model that you can build. I have included the details in the DIY-HRTF.pdf in the Supplementary Material. ax

About the Author

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 (now TDL Technology) to develop audio electronics. He is still the TDL president and principal designer.

Show Report

Munich High End Show 2015

Looking for the Great Product Designs

From May 14–17, 2015, more than 500 exhibitors from 42 countries, representing more than 900 brands, attended the Munich High End show. Event organizers claim Munich High End is the largest and most important event for high-quality consumer electronics, and that supposition was certainly underlined this year. The show, organized by the High End Society, occupied four halls and three Atria floors with 140 listening and conference rooms of the Munich Order Center (MOC) convention center in Munich, Germany. If last year's show was considered huge, they topped it this year.

By
Ward Maas
(Netherlands)

Photo 1: Realhorns put its "the most interesting sound system in the world" on display at Munich High End 2015.

As the show is about high-quality audio, it was easy to get overwhelmed. Where to start? Let's begin with a look at the work of a company called Realhorns (www.realhorns.de). **Photo 1** shows what the company means—no compromises when it comes to genuine, true authentic horns. The system that this monster fits in is for rent. Luckily, Realhorns has a large truck to transport it. They call it "the most interesting sound system in the world." True, you can talk about it or be completely speechless!

Amazing Loudspeakers

Moving on to real world loudspeakers that can fit in a normal house, there was the introduction of the Sopra loudspeaker line from French company Focal (www.focal.com). The Sopra Nr. 1 is a compact, two-way Bass-reflex loudspeaker that introduces well-thought out improvements compared to previous

Photo 2: The Sopra Nr. 2 is a three-way floor standing loudspeaker from Focal.

Photo 3: Devialet's Phantom loudspeaker can be used solo, duo, or multiroom.

products. The Sopra Nr. 2, its larger brother, is a three-way floor standing loudspeaker that features all the same improvements but is clearly better equipped for the bass region (see **Photo 2**). Focal already had a name for being research-driven. But it has once again improved its R&D, and that was clearly audible in these new products.

Another French company, Devialet (www.devialet.com), showed its Phantom loudspeaker (sometimes called a connected speaker). It can be

Photo 4a: Kii Audio introduced its new loudspeaker, the Kii THREE system—which is its first-ever product. b: The Kii Audio founding team is, from left to right, Wim Weijers, Sales Manager; Chris Reichardt, CEO; Bruno Putzeys, CTO; Bart van der Laan, COO; and Thomas Jansen, Product Manager.

Photo 5: Kyron Audio presented the Kronos, a nice boxless dipole system, which sounded great.

used solo, duo, or multiroom—wireless that is 6 liter units going from 16 Hz to 25 kHz, 3,000 W/105 dB (see **Photo 3**). It is really beautifully engineered and protected by 88 patents. How it sounds is difficult to describe—certainly very well and impressive. But there is something with this product that makes me say that if you are an Apple aficionado you must have it.

DSPs on Display

Since the introduction of digital signal processing (DSP), there have been many attempts to correct loudspeakers to a more ideal behavior. However, digital signal processing itself is not without debate: "meddling with the most important part of the audio system" has been subject to fierce debates. The better sounding systems were always quite moderate with corrections. The use of good components results in fewer issues that may need to be corrected.

A new company, Kii Audio (www.kiiaudio.com), promoted an exclusive first audition at MSM Studios, in the center of Munich, to introduce its new loudspeaker, the Kii THREE system—which is, in fact, its first-ever product. Knowing that Bruno Putzeys, one of the founders of modern Class-D amplification, is the CTO of Kii Audio resulted in high expectations. The system consists of four woofers, a midrange and a wave-guided tweeter, driven by six Class-D amplifiers and ground-breaking massive digital signal processing. How does it sound? To put it simply, very detailed but mostly impressively pleasant. This loudspeaker is undoubtedly one that indicates the shape of things to come (see **Photo 4**).

Another company not shying away from digital signal processing and room correction is Kyron Audio (www.kyronaudio.com.au). It presented a nice dipole system, which sounded great. The Kronos is a classical boxless design with modern electronic support (see **Photo 5**).

Walking into a demo room, I found the Enigmacoustics (www.enigmacoustics.com) Mythology M1, a two-way loudspeaker with an augmented electrostatic super tweeter (see **Photo 6**). It is always a pleasure to be surprised by an unexpected high-quality

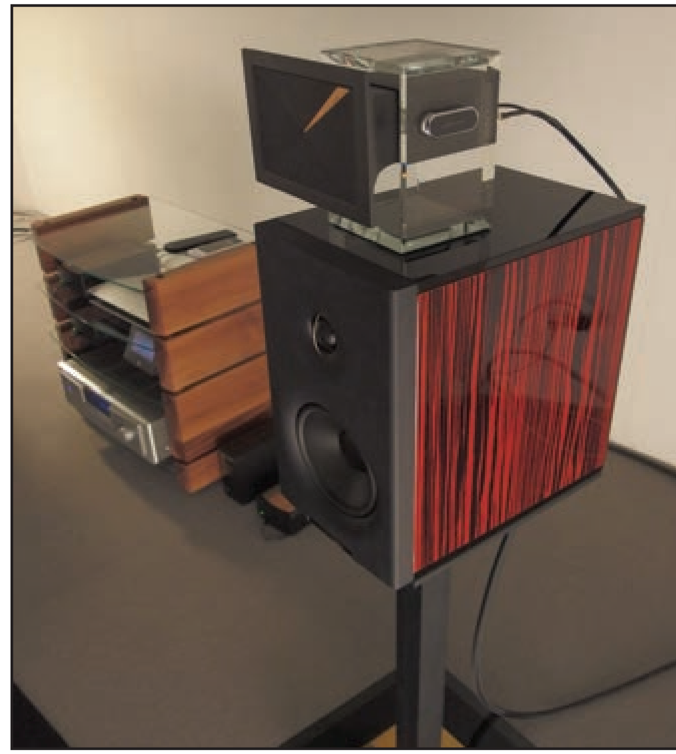


Photo 6: Enigmacoustics' Mythology M1 is a two-way loudspeaker with an augmented electrostatic super tweeter.

Photo 7: IsoAcoustics showcased its Aperta speaker isolation speaker stands.

system. Apparently it is not widely available, but I expect that to change.

Array of Accessories

Now during a show like this, you are also bombarded with all kinds of accessories imaginable. I have to admit I normally do not pay too much attention to them. But this year, two accessories are worth mentioning. First, there are the Aperta speaker isolation speaker stands from IsoAcoustics (www.isoacoustics.com). Their simple demo convinced me much can be gained in stiffness of bass and sound stage improvement using speaker decoupling (see **Photo 7**). With a recognizable name in the pro-audio world, I expect IsoAcoustics to do well in the consumer-audio world as well.

The second accessory is the Staywired zipper cable management system (www.staywired.de). Of course, there are always cable messes that are not so easy to handle. But I expect that a true DIYer will do whatever it takes.

Amps and Preamps

Yes, I did put the accessories and loudspeakers before the amplifiers, preamps, and streaming products. The reason for this is that is you could see them all—from small to large, cheap to extremely expensive, and ugly to beautiful.

In fact, the number of products that were not okay was very limited. So, I restricted myself to only mentioning a few products that stood out in one way or another.

First is Naim (www.naimaudio.com) with its simply understated well-performing audio products. While you may not use the amplifier in this fashion, I love the wing doors style cover (see **Photo 8**).

Second, from the newcomer's section is Westminsterlab (www.westminsterlab.co.uk). The company presented its UNUM mono amplifier. This amplifier is true to its rather extensive list of beliefs of how an amplifier should be built and what practices should be avoided. This amplifier is milled from one solid block of aluminum. Instead of PCBs, Westminsterlab uses sheets of polyether ether ketone (PEEK), hardwired all the components on a through-hole grid, and used aircraft industry titanium screws, carbon fiber parts, and so forth. It is certainly, an amplifier with an attitude (see **Photo 9**).

Densen Audio Technologies (www.densen.com) also offered another range of well-received products. Densen Audio showcased a complete line of high-quality products from the typical Nordic school of design—sleek and understated shapes. One remarkable feature is their upgrading philosophy,

Photo 8: Naim (www.naimaudio.com) with its simply understated well-performing audio products offered this amplifier with a wing doors style cover.

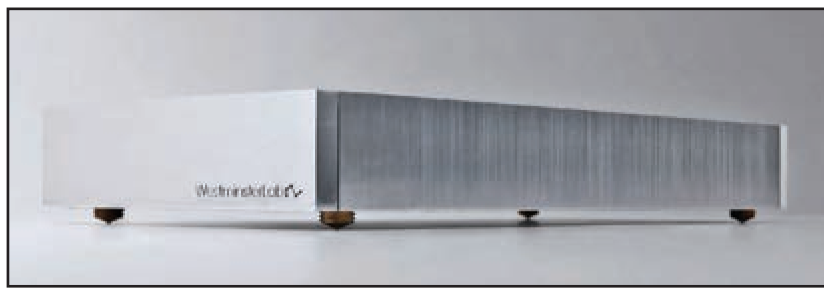


Photo 9: Westminsterlab presented its UNUM mono amplifier.

Photo 10: The Reed Company was one of several companies wanting to bring back the turntable, with its Muse 3C version.

About the Author

Ward Maas is the owner of Pilgham Audio. He studied electronics, marketing, and amplifier design. During his long career in consumer electronics, Ward has worked in areas ranging from CD standardization to radio and television to personal GPS navigation. Ward has worked on an extreme low-noise magnetic cartridge preamplifier and several special amplifier products. As the CTO of "Witchworld," a theme park near Amsterdam, he also works with animatronics. He lives in Almere, Netherlands, with his wife and son.

Photo 11: Vandenhul introduced its first turntable, "The Point One."

even older products remain upgradeable. That, combined with a lifetime guarantee, makes it a brand to check out.

Turntables Galore

And then there were turntables, lots of them. It seems no one remembered that only recently they were almost obsolete. The Reed Company from Lithuania (www.reed.lt) presented its Muse 3C turntable (see **Photo 10**). It is a really beautiful design and one of its special features is the possibility to convert it from a friction drive to a belt drive in minutes. It is a clever design.

Walking into the Vandenhul (www.vandenhul.com) demo room, I realized that it, too, was introducing its first turntable "The Point One." The source of this specific name comes from the fact that the spindle stands on a single ball and the platter (which rests on top of the spindle) is centered and stabilized by a magnetic field (see **Photo 11**).

Photo 12: The Auralic Aries Mini (shown at right) is the little brother of the Aries.

Streaming Products

Regarding streaming, there was the really interesting introduction of the Auralic Aries Mini (www.auralic.com). This is the little brother of the Aries. Comparing specs and prices, I would not be surprised if this could be one of the year's top sellers. The pricing ($399) compared to the Aries ($1599) is very promising (see **Photo 12**).

In just a few years, streaming has gone from new to mature. Although there is still discussion regarding formats, I have the feeling that it is just PC hardware, codecs, software, and apps. In my opinion, it is an unstable base for sustainable competitive market advantages. So, fasten your seat belts because this promises to be a bumpy ride.

Audio in Automobiles

A special section of the show was called "High End on Wheels." There, we found a group of high-end cars equipped with high-end audio systems. Think European luxury brands (e.g., BMW, Porsche, Audi, Mercedes, etc.) and about 25% of German car buyers want the music upgrade package installed. For a lot of people, their car is the only place where they are able to listen to music. So the ability to test how music could sound in a luxury car was highly appreciated. The reason not to buy a Rolls Royce certainly has nothing to do with the sound system.

High End 2015 Final Thoughts

All the new products and the show's growth can be seen as a positive. However, I believe there is more going on. In many countries, high-end audio is not doing so well. There are many reasons for this, but one of them is that the Internet is making the world a lot smaller. This means the traditional value chain is changing. If you are a distributor or a sales outlet, you have to justify your role in the selling chain from manufacturer to final consumer. If not, you find the world is rapidly changing.

Nubert (www.nubert.de), a German manufacturer, is already directly supplying to end-consumers, a change which resulted in it receiving the "Best Brand" award in 2014/2015. The value for the money is stunning and suddenly "high end" doesn't mean "high price" anymore… I expect more manufacturers will follow this example.

After four busy days, at the end of the show, I found myself once more in a Munich beer garden with a perfect beer in front of me. At least for the short time, the future looked splendid. For more information, visit www.highendsociety.de

Considerations for the MPEG-H Audio Standard

This article discusses the plan for the new MPEG-H Audio standard, a real-time 3-D audio encoder system with applications for broadcast and streaming distribution, developed in large part by the Fraunhofer Institute in Germany, and recently finalized and submitted as an ISO/MPEG standard.

By
João Martins
(Editor-in-Chief)

MPEG-H is a group of standards under development by the ISO/IEC Moving Picture Experts Group (MPEG) for a digital container standard, a video compression standard, an audio compression standard, and two conformance-testing standards. The group of standards is formally known as ISO/IEC 23008—high-efficiency coding and media delivery in heterogeneous environments. MPEG-H Audio is just the third part of 13 document chapters and basically covers the codec for 3-D audio, which supports: an object-audio codec for "interactive sound mixing;" a realistic audio experience with additional front- and rear-height speaker channels to traditional surround sound setups; and higher order ambisonics (HOA), a technique for reproducing a complete spherical soundfield to provide a fully immersive live sound experience.

The Applications

The basic MPEG-H audio technology, which is under discussion for publication as an ISO/IEC international standard, results basically from cooperation between the Fraunhofer Institute for Integrated Circuits IIS, Technicolor and Qualcomm.

Much of the research work followed demonstrations of new emerging immersive formats (e.g., Dolby Atmos and Auro-3D), and the direct commitment from several organizations including the European Broadcast Union (EBU/UER) and the Society of Motion Picture and Television Engineers (SMPTE). The organizations are promoting new "enhanced" audio experiences and technologies, including live rendering of audio objects for multichannel sound and binaural reproduction.

Since 2011, those efforts have also involved several prestigious European institutions, from the British Broadcasting Corp. (BBC) to the Fraunhofer IIS in Germany, which is also the provider of the HE-AAC multichannel audio codec powering half the world's TV surround sound today.

While Dolby Atmos is a format directly promoted for cinema production and exhibition, that later became also used in home-theater application, the before mentioned institutions are looking at consumer delivery of immersive audio in broadcast and streaming media on any type of distribution network—from direct Internet streaming to cable/IPTV operator platforms. To promote the MPEG-H standard, Fraunhofer IIS, Qualcomm Technologies, and Technicolor created the MPEG-H Audio Alliance.

In the MPEG-H Audio Alliance perspective, the MPEG-H Audio standard enables live broadcasts with object-based 3-D audio across all devices, providing viewers the ability to tailor the audio to suit their personal listening preferences. The system encodes elements of the audio as interactive objects so viewers at home can adjust the sound to their preference while also enhancing today's traditional multichannel surround sound broadcasts to create a more realistic audio experience.

From Object-Based to Speakers

In this context, it is important to understand how to differentiate the various channel-based

Pictured from left to right are Robert Bleidt, Division General Manager at Fraunhofer USA Digital Media Technologies and Claude Gagnon, Technicolor's SVP, Content Solutions & Industry Relations.

audio delivery solutions from the object-based audio concept. According to Nuno Fonseca (Instituto Politécnico de Leiria, Portugal), one of the leading researchers in the area of object-based audio, "Immersive sound is the designation that was given to the existence of sound in a 3 dimensional world, including a height dimension." Immersive audio can follow the channel-based and object-based approaches. In channel-based audio "all sound sources are mixed to output channels, where each output channel corresponds to a speaker (or an array of speakers) that will exist at a predefined position. Object-based audio uses a different approach, using metadata. Instead of using channels with predefined positions, each audio object will have the desired audio stream and metadata with time varying information on how that audio stream should be played (position in space, apparent size, etc.)."

Thanks mostly to Dolby's efforts in the promotion of its Dolby Atmos format for cinemas and the home, new delivery solutions with support for the

Delivering Multi-Platform Immersive Audio with the New MPEG-H Audio Standard

By
Robert Bleidt
(Fraunhofer USA)

New standards for TV audio and new content offerings from streaming media providers will soon provide consumers with immersive sound in their living rooms. Immersive sound is a step beyond today's surround sound, offering sound from above to complete the auditory illusion of being in the audience or at the performance instead of at home.

Immersive sound is being deployed in cinemas today, with Dolby's Atmos and Auro's Auro-3-D systems the most popular, while DTS has indicated it will re-enter the cinema market with the DTS:X system. Derivatives of these systems are also finding their way into Blu-ray releases as extensions to existing codecs in the disc's specification.

However, even the Blu-ray versions of these codecs require several megabits of data to deliver immersive audio. This is not practical for TV broadcast or Internet streaming, and for these purposes next-generation audio codecs must be used.

One of these codecs is the recently approved MPEG-H Audio Standard from the MPEG audio group that developed the MP3 and the AAC family of codecs. An audio system built around this codec is being developed by the MPEG-H Audio Alliance of Fraunhofer, Technicolor, and Qualcomm. It is now being proposed for new streaming services as well as being evaluated for the new ATSC 3.0 TV broadcast standard for the US.

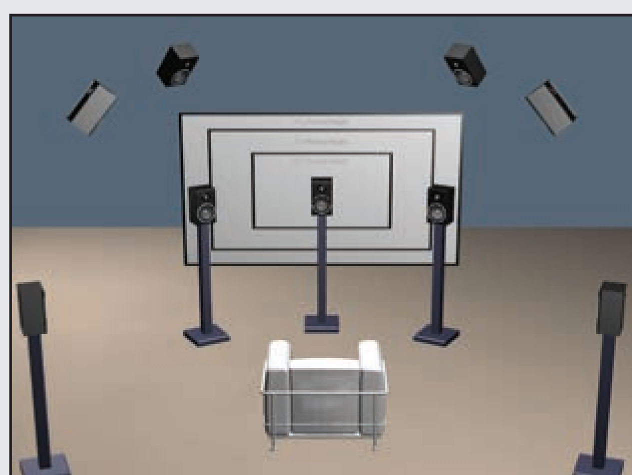


Photo 1: The 5.1 + 4 H speaker configuration is shown with four overhead speakers. A 7.1 + 4 H configuration is also possible, and for limited hardware only two front channels may be considered.

Although the standard is capable of carrying up to 128 channels of audio, and loudspeaker configurations are defined up to 22.2 channels, in practical use in the Alliance system, the use of today's 5.1 or 7.1 surround speaker layouts is proposed, with four overhead corner speakers. Listening tests at Fraunhofer have shown the 5.1 + 4 overhead speaker arrangement can make a substantial improvement in the perceived realism of sound, almost as great as the step from stereo to 5.1 surround (see **Photo 1**).

Systems for playback of Blu-ray immersive audio were shown at CES 2015 and have begun to enter the market. Most of them use additional drivers in the speaker enclosure aimed to reflect sound for the height channels off the ceiling of the listening room. Add-on bounce speakers for existing 5.1 or 7.1 installations have also appeared. These systems will provide immersive, though diffuse, overhead sound in many rooms, but they represent a compromise that still requires additional wiring and setup.

While ceiling-mounted loudspeakers offer the ultimate immersive sound quality, the consumer trend toward flat-screen displays have led them to abandon discrete speakers for soundbars. This has actually reduced the possible sound quality, as almost all soundbars are stereo devices. Fraunhofer has shown a potential solution to this problem with 3-D soundbars that have additional speakers and acoustic processing to enable an array of speakers surrounding the TV to produce realistic immersive sound (see **Photo 2**). This offers the mainstream consumer a way experience good immersive sound without any additional wiring or configuration—a décor-friendly, hang it and hear it experience.

Another trend is toward consumption on tablets and other mobile devices, even within the home. This represents a difficult challenge for audio reproduction given the typical 16-mm speakers in tablets, likely with a resonant frequency in the 600-to-800-Hz range, and often only exposed to the listener through rear-firing grilles or even internal ducts for mechanical convenience. Fraunhofer has had experience in addressing these challenges through psychoacoustic processing, perhaps applied in combination with specialized speaker driver circuits that extract the maximum possible excursion from the speaker.

One example is the Fraunhofer Cingo technology that appears in Google's Nexus tablets and now in the Gear VR virtual reality headset adapter for Samsung phones. Fraunhofer has extended this technology, originally developed for rendering surround sound on tablet speakers and headphones, to be able to recreate a realistic sound image for immersive audio as well. Another technology is the binaural rendering function built into the MPEG-H codec itself for headphone reproduction.

Thus, with these delivery modes—speakers, soundbars, and tablet software—MPEG-H can deliver immersive sound to the universe of devices consumers may use to experience new TV and video programming. Additionally, we have included functions in the system to enable the dynamic range to be tailored to the listening environment, moving beyond the "night mode" setting of today's codecs, to multiple dynamic range settings with different target levels for different use cases.

These different target levels are needed to account for the fact that, on mobile devices, the same volume control is used to set the loudness for all content. At home in the living room, a consumer may decide to play music from his phone,

Photo 2: This is an early concept prototype of a 3-D soundbar. Commercial products would be smaller and perhaps shaped more like a conventional soundbar.

Internet radio, or media server, but with a different volume setting for each. On a phone or tablet, the operating system does not distinguish between TV programming recorded with normal headroom, and music which is now uniformly compressed, limited, and clipped to maximize loudness, typically 10 to 15 dB louder. A compromise solution to this difference is to boost the level of video programming on mobile devices to match that of music.

Both of these issues—reducing dynamic range and matching video to music loudness—are addressed in the MPEG-H system through additional sets of dynamic range control coefficients sent in the bitstream. The system also offers special processing of the dialog to increase the intelligibility in noisy environments.

MPEG-H also offers three ways to carry immersive sound—traditional channels, audio objects, and higher-order ambisonics. More information on the MPEG-H Audio Alliance system is available at www.MPEGHAA.com.

About the Author

Robert Bleidt is Division General Manager of Fraunhofer USA Digital Media Technologies. He is the inventor of the award-winning Sonnox/Fraunhofer codec plug-in, widely used in music mastering, led the extension of Fraunhofer's codec business to an open-source model by inclusion in Android, and developed Fraunhofer's Symphoria automotive audio business. Before joining Fraunhofer, he was president of Streamcrest Associates, a product and business strategy consulting firm in new media technologies. Previously, he was Director of Product Management and Business Strategy for the MPEG-4 business of Philips Digital Networks and managed the development of Philips' Emmy-winning asset management system for television broadcasting. He also worked for Sarnoff Real Time Corp. and was president of Image Circuits, a consulting engineering firm and manufacturer of HDTV research equipment.

Fraunhofer's 3-D TV Audio System enables viewers to personalize sound broadcasts to suit their personal preferences. With a click of the remote, they can create their favorite mix of sports broadcasts. (©Getty Images/Fraunhofer USA)

MPEG-H audio format will be able to translate the additional height element, using overhead speakers or up-firing additional drivers. Of course, Dolby Atmos is more than simply adding ceiling speakers. It effectively combines complete authoring, distribution, and playback tools, including improved audio quality and timbre matching, as well as greater spatial control and resolution with which, the immersive or "3-D" perception of audio can benefit from developments in the area of real-time encoding of audio objects.

Previously, several other sound reproduction systems explored additional dimensions in sound reproductions, from ambisonics to 3-D Vector Base Amplitude Panning (VBAP) and Auro-3D. The MPEG-H 3-D audio standard could effectively have a lot to gain from the consumer's adoption of Dolby Atmos solutions (or the new DTS:X format) in helping to introduce elevated loudspeakers to create a real 3-D audio impression, since those loudspeaker setups are able to deliver higher spatial fidelity than the established 5.1 setup.

Of course, the MPEG-H audio project is also intended to solve additional challenges, which were not addressed by Dolby Atmos or other "immersive audio" solutions. Considerations such as production compatibility between different immersive 3-D audio formats, or efficient distribution to existing channels and delivery systems, including a single "3D" loudspeaker setup.

The MPEG-H Audio Effort

From earlier work dating back to 2011, with initial presentations on 3-D audio at the MPEG meetings, discussions around 3-D audio lead to a "Call for Proposals" for such new 3-D audio technologies in January 2013.

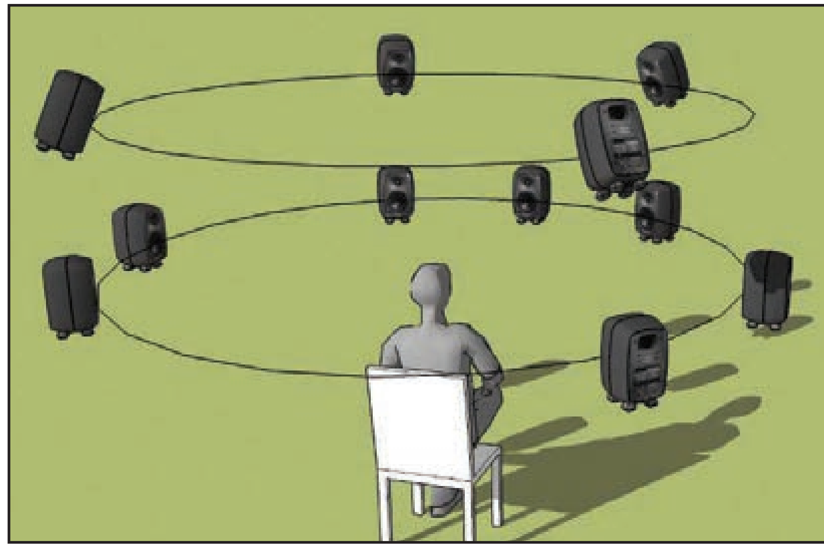
The requirements and application scenarios specified for the new technology included bit rate parameters and compatibility of rendered audio on various loudspeaker setups from 22.2 down to 5.1, plus binauralized rendering for virtualized headphone playback. The document also specified that submissions would be independently conducted for "channel and object (CO) based input" and "Higher Order Ambisonics (HOA) input content types." At the 105th MPEG meeting in July/August 2013, the Reference Model technology was selected from the received submissions (four for CO and three for HOA) based on their technical merits, to serve as the baseline for further collaborative technical refinement of the specification. The winning technology came from Fraunhofer IIS (the CO part) and Technicolor/Orange Labs (the HOA part). In a subsequent step, both companies agreed to merge the proposals into a single harmonized system and work together in areas such as binaural rendering. The final stage of the specification, which is the submission of a document for approval as an international ISO/IEC standard, was issued at the 111th MPEG meeting in Geneva, Switzerland, in February 2015. The documents have been available for public discussion on The MPEG website, http://mpeg.chiariglione.org/standards/mpeg-h.

Trials

With the MPEG-H standard complete and undergoing field tests and deployment, there is a practical solution to translate object-based audio productions into existing delivery multichannel systems, from broadcast chains to home-theater setups.

The MPEG-H standard enables high-efficiency coding and media delivery of high-quality 3-D audio content, independently of the number of loudspeaker channels used for reproduction using an appropriate coding and rendering algorithm. Based on the AAC codec family and new technologies, this system enables an easy upgrade path of existing infrastructures when adopting MPEG-H without major equipment changes.

At IBC 2014, the Fraunhofer/Technicolor demonstration showed the world's first real-time encoder for interactive and immersive TV audio. The real-time hardware featured the ability to encode audio for live broadcasts from stereo up to 3-D



MPEG-H has defined speaker configurations from 1.0 (mono) to 22.2, including 7.1 and 5.1 plus 4 height speakers above corner as shown. (Image courtesy Fraunhofer IIS)

sound in 7.1+4H format with additional tracks for interactive objects including commentary in several languages, ambient sound, or sound effects.

The solution includes a professional decoder to recover the uncompressed audio for further editing and mixing in the studio; a real-time encoder for delivery to consumers—over the Internet for new media use or for upcoming over-the-air broadcast systems (such as ATSC 3.0); and a professional decoder used to monitor the transmitted encoder's output.

In all the IBC 2014 and the National Association of Broadcasters (NAB) 2015 demonstrations were supported by Qualcomm, while Technicolor supervised the production services for content creation. Fraunhofer also introduced a new "surround soundbar frame" for television sets, a concept that delivers 3-D audio without using multiple external speakers.

As Robert Bleidt, General Manager of the Audio and Multimedia Division for Fraunhofer USA Digital Media Technologies, explained, "Future UHDTVs might build similar technology into the TV itself, offering consumers an 'un-box, plug-in, enjoy' experience with immersive sound. That will be much better than the soundbars of today, with no wires or external components at all. While high-end home theater enthusiasts will likely still prefer nine or 11 separate speakers for the ultimate 3-D sound quality, this new soundbar concept will greatly enhance the audio experience for a broad consumer base without complex installation and setup," he added.

Resources

Dolby, "Dolby® Atmos® Next-Generation Audio for Cinema," White Paper, 2014, www.dolby.com/uploadedFiles/Assets/US/Doc/Professional/Dolby-Atmos-Next-Generation-Audio-for-Cinema.pdf.

J. Herre, J. Hilpert, A. Kuntz, and J. Plogsties, "MPEG-H Audio—The New Standard for Universal Spatial/3D Audio Coding," Audio Engineering Society (AES) paper, December 2014.

Moving Picture Experts Group (MPEG), http://mpeg.chiariglione.org/standards/mpeg-h.

N. Fonseca, "Immersive Sound Design Using Particle Systems," Convention e-Brief 193, presented at the 138th Audio Engineering Society (AES) Convention, 2015.

——, "3D Particle Systems for Audio Applications," 16th International Conference on Digital Audio Effects, 2013.

V. Pulkki, "Virtual Sound Source Positioning Using Vector Base Amplitude Panning," *Journal of the Audio Engineering Society*, June 1997

T. Sporer, J. Liebetrau, S. Werner, S. Kepplinger, T. Gabb, and T Sieder, "Localization Of Audio Objects In Multi-Channel Reproduction Systems," Audio Engineering Society (AES) paper, 57th AES International Conference, 2015.

Upmixing—Stereo to Surround

New Demands and Requirements for a Modern Upmix Process

As NUGEN Audio prepares to launch the NUGEN Audio Halo Upmix Plug-in, a new upmixer for the post-production market at IBC 2015, the company's Creative Director and Co-Founder Jon Schorah reflects on the changing landscape and increasingly complex demands of the modern audio production environment.

By
Jon Schorah
(United Kingdom)

The notion of upmixing audio from stereo to surround formats is not a new one. For many years, cinematic audio has been routinely delivered as a 7.1 audio mix. Where necessary, upmixing has been used to generate audio from original stereo material in a suitable format for the surround environment.

More recently, the demand for original TV programming in surround formats has led to a widespread use of upmixing in post-production. Even more recently, we've seen the emergence of object audio (e.g., Dolby Atmos), which makes use of 9.1 audio beds and promises to draw new attention to the subject of upmixing.

A movie theater is a relatively predicable environment, acoustically treated and relying on a playback system properly calibrated for 7.1 playback. However, the home listening environment has no such constraints; additionally, the delivered 5.1 surround mix may be automatically down-mixed into stereo by the consumer set-top box (STB). Object audio is also subject to these challenges. Although object audio mixes are largely destined for the movie theater, there is no reason that these same mixes cannot be re-interpreted for home listeners. Since object audio is theoretically infinitely scalable, we can join the movie theater and the home TV experience together via a single translatable mix.

Good Upmix Source

For an upmix process to be able to perform well, beyond the traditional 7.1 cinematic presentation, several new requirements come into play.

First and foremost, the process must be able to provide a good upmix from the original source material. Of course, "good" is somewhat subjective, but two seemingly conflicting criteria often present themselves depending upon the purpose of the upmixed audio. One is a requirement for the upmix to respect the original source material, providing a naturally extended panorama that "unfolds" the original into a surround context. This is important when the original source is a broadcast mix that is simply being translated for broadcast in surround. In this situation, creative decisions have been made and these preferences need to be respected and represented in the upmix produced. The other situation involves upmixing of original stereo material as part of a more complex arrangement being produced in surround. In this case, getting the upmix to fit and contribute to the overall sound of the surround mix is as important, if not more, than maintaining a direct relationship with the original source audio.

Flexible Process

The upmix process also needs to be flexible. Traditionally, the majority of cinematic releases have used a 7.1 configuration, with a 5.1 configuration for TV. For film audio engineers, a 7.1 upmix solution is perfectly adequate. However, many freelance engineers routinely offer services to both the film and television industries and, therefore, need a certain level of flexibility. For those with an eye on the future, the advent of object audio offers an enticing 9.1 bed track, including additional information for overhead loudspeaker arrays. According to one manufacturer's technical white paper—in theory, an object-based mix, including 9.1 beds, can be automatically translated to 7.1, 5.1, and stereo formats allowing the audio professional to "author once, optimize everywhere." This has the potential of introducing a universally translatable mix, but it remains to be seen how and if object audio will be established in practice.

Dialog Isolation

In the realm of restoration and re-imagining of old recordings and upmixing for television broadcast, there is an additional concern: dialog isolation. Most delivery specifications require a high degree of dialog isolation into the center channel for aesthetic purposes and delivery consistency. Recent increases in computing power have led to great improvements in processes that facilitate this isolation. Techniques that go far beyond simple mid-side and frequency isolation are now possible, with minimal over-isolation and transparent transmission of directional content passed to the sides.

Downmix Compatible

In addition, an upmix must, in most cases, be highly downmix-compatible. This means that a downmixed version of the upmixed audio (the upmix collapsed back down to stereo) must still sound very close to the original. This is important if the upmixed audio is ever to be repurposed in a different format (e.g., a feature film repurposed for 5.1 and/or stereo TV in which downmixes are commonly used to produce the broadcast audio). There are many upmix processes that can be used to generate a full-sounding upmix. However, some of these can cause serious downmix compatibility issues. For original TV production, the 5.1 downmix needs to correlate closely to the original stereo source as it is highly likely that the downmix will be what the consumer eventually hears. Even in the context of a new full-surround mix, where the downmix will not be directly related to a single upmixed audio source, it is still an important requirement to deliver a down-mixed track free from unnatural sounding delay and phase artefacts.

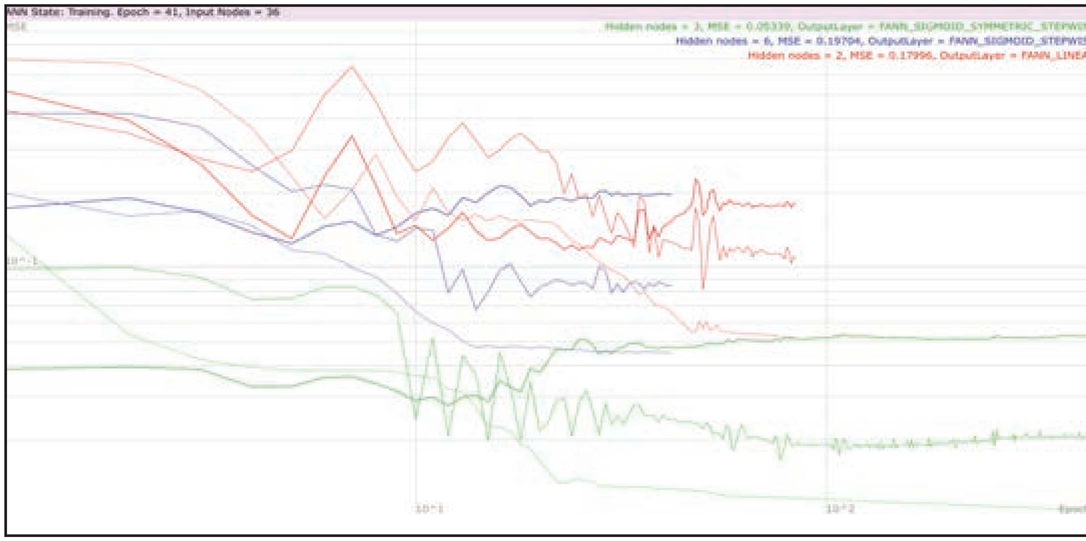
Loudness

Finally, there is the consideration of loudness. International standards now regulate the loudness of audio delivery for television across the world, and cinema sound is moving toward similar recommended practices. This raises a potential

About the Author

Jon Schorah is the creative director and co-founder of NUGEN Audio, one of the world's leading manufacturers of loudness products. Jon has a background in mastering and engineering and has considerable experience in wider aspects of the industry. A 1992 Leeds University (UK) graduate, in recent years Jon has focused on product design with a particular interest in the usability and workflow aspects of audio software.

NUGEN Audio is preparing to release its newest plug-in the NUGEN Audio Halo Upmix plug-in.



Halo Upmix uses a neural network artificial intelligence algorithm to extract the dialog portion from a piece of audio to allow isolation in the center channel, or in left-center-right (LCR). The neural network used was produced by running a genetic algorithm on trained neural networks. The graph shows some competing neural networks' test scores as they are trained using backward propagation.

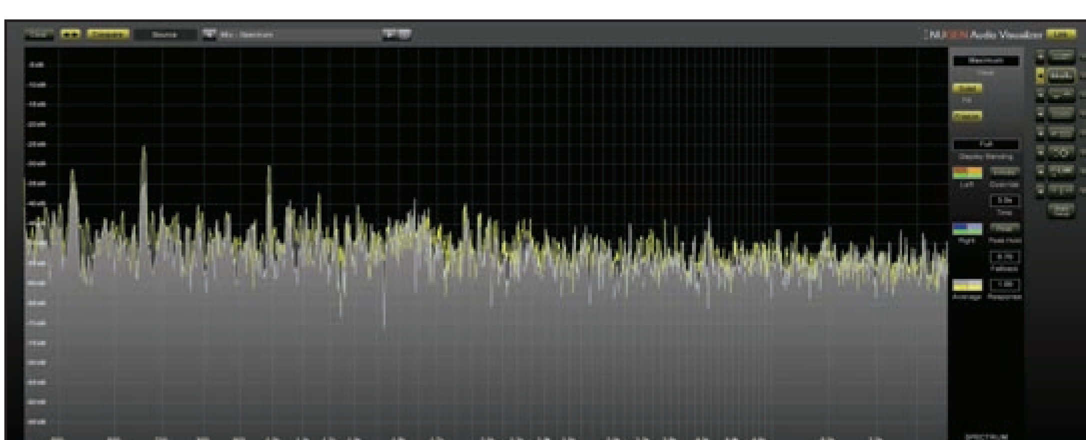
problem, since a 7.1 original mix, a 5.1 TV reversion, and a stereo TV downmix of the same audio may not all have the same loudness. If each mix could be discretely delivered this would not be a major issue because each mix could be normalized to loudness compliance with minor global offsets. However, this is not standard practice.

For the majority of home listeners, as we've noted, the 5.1 mix is delivered to the consumer's STB where it is automatically downmixed into stereo. To make matters worse, the difference in loudness between the surround and downmix versions is not a constant value from one program to another. Nor is the difference in loudness always in the same direction, it can be louder or quieter for a number of different reasons. Plus, the high degree of center channel dialog isolation often demanded for television broadcast can further exacerbate this loudness disparity.

With a possible difference of several loudness units (LU) in either direction, in some situations we might see a re-introduction of loudness jumps from one program to the next simply due to differing loudness in the consumer downmix. Gaining an understanding of this difference in loudness during the upmix process is of critical importance if delivery specifications are to be met, yet little research or recommended practice is currently available on the matter.

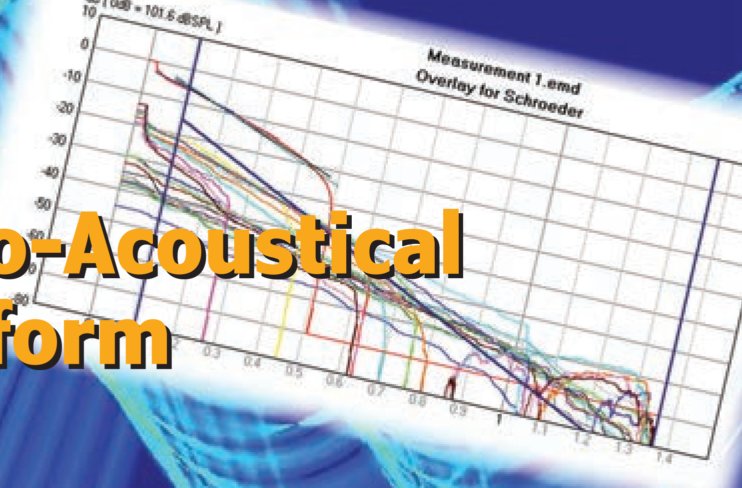
Upmix Seen in Context

In summary, as much as any other aspect of audio production, upmix requirements now need to be seen in the context of the "digital content everywhere" consumer revolution. Gone are the days of isolated production for a single playout scenario. Cinematic releases are routinely repurposed for television and premium television content broadcast in 5.1 is routinely downmixed into stereo, then further repurposed for mobile listening. An understanding of these different processes and how the original mix will translate is an important part of the upmix decision if creative intent and consumer satisfaction are to be preserved throughout the distribution chain.



Here a stereo downmix is compared with the original stereo source.


Build your dream system
With this how-to loudspeaker book, you will be able to crank up the volume on a first-rate system that you designed and built yourself.
cc-webshop.com
Speaker Building 201
A Comprehensive Course in Speaker Design
by Ray Alden
With 11 completely designed speaker systems including a 5.1 home theater system
Your guide to understanding the mysteries of speakers
A perfect mix of theory and practical application
Build Your Own FIRST-RATE System
For Beginners and Serious Enthusiasts
A
B
C
D
E
F
G
H



The EASERA Electro-Acoustical Measurement Platform

The Electronic and Acoustic System Evaluation and Response Analysis (EASERA) measurement platform was developed by Software Design Ahnert (SDA) of Germany and is distributed by Ahnert Feistel Media Group, AFMG Technologies GmbH, which many readers will recognize as the developers of the EASE computerized electro-acoustic modeling software. The development team, led by Dr. Wolfgang Ahnert and Dr. Stefan Feistel, released EASERA in 2006.

By
Richard Honeycutt
(United States)

EASERA's Sections

The EASERA measurement platform consists of four functional sections:

- A signal generator
- The measurement engine
- A real-time analyzer (RTA)
- The post-processor

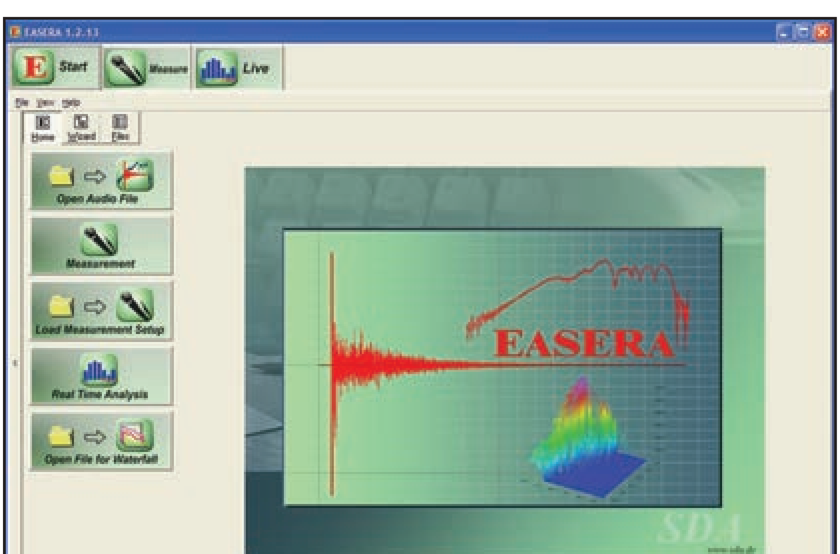


Photo 1: The user begins the measurement process from this opening screen.

The signal generator provides a variety of stimuli signals, including sine waves, sine sweeps, Maximum Length Sequences (MLS, available in the Pro version or as an optional module), TDS sweeps (with the optional TDS module), and/or noise signals. The sine sweeps, MLS, and noise signals are available in white, pink, or weighted versions. User-defined signals can also be generated.

The measurement engine collects and provides data for the RTA, as well as Impulse Response (IR) data for post-processing. In addition to functioning as a stimulus-response analyzer using the included signal generator, the measurement engine is also capable of collecting data using external impulse or wave signals. Sampling rates from 8 to 192 kHz are supported, with 32-bit resolution (hardware-dependent). Single- or dual-channel Fast Fourier Transforms (FFTs) can be performed, and automated measurements can be implemented. Calibration of external hardware is supported.

The RTA provides multiple ways to perform a fast onsite analysis of the acoustical or electroacoustical environment. The post-processing engine calculates all acoustic measurements from the IR, according to the International Organization for Standardization (ISO) Standard 3382 and higher.

EASERA can be run on Microsoft Windows XP or higher and in conjunction with nearly every modern measurement interface, although the use of Audio Stream Input/Output (ASIO) drivers is recommended.

Start Up

When EASERA is opened for the first time, the user is led through the licensing process and can choose either a purchased license or a 30-day demo license using the companion license management software EASERAGuard. When the licensing process is complete, the opening screen shown in **Photo 1** appears. From here, an audio file can be loaded, a preconfigured measurement initiated, or a new IR or RTA live measurement can be performed. After any of the large function buttons has been pressed, the new screen always contains START in the upper left-hand corner. Clicking the START button returns the user to the opening screen.

Clicking on either the Measure button at the top or the Measurement button at left brings up the Measurement Setup screen shown in **Photo 2a**. **Photo 2b** shows all four sections of the Measurement tab at the same time. The CONFIG buttons are used to inform EASERA of the sensitivity, gain, connection points, and input and output voltages of hardware devices. The CALIBRATE button sets up the software to calibrate the hardware.

The Select Setup takes the user to the Select Measurement Setup screen shown in **Photo 3**. The screen enables the user to select audio input and output devices in the computer or external hardware. Choices of overall measurement setups are listed in the left-hand panel and illustrated in the right-hand panel. Once the desired setup is highlighted, pressing the Select button at the bottom brings the user back to the Measurement Setup screen.

The Next button at the bottom right of the Measurement Setup Screen brings up the Choose Stimulus Parameters screen shown in **Photo 4**. After selecting the desired waveform, time, and frequency weighting, the user can press Next again to bring up the Adjust Levels screen shown in **Photo 5**. This is where the input and output levels and gain compensations are controlled.

Measurements

Once these parameters have been set, pressing Next opens the Start Measurement screen (see **Photo 6**), from which the user can select additional processing to be applied to measurements on the fly, enter labels for the measurement, as well as

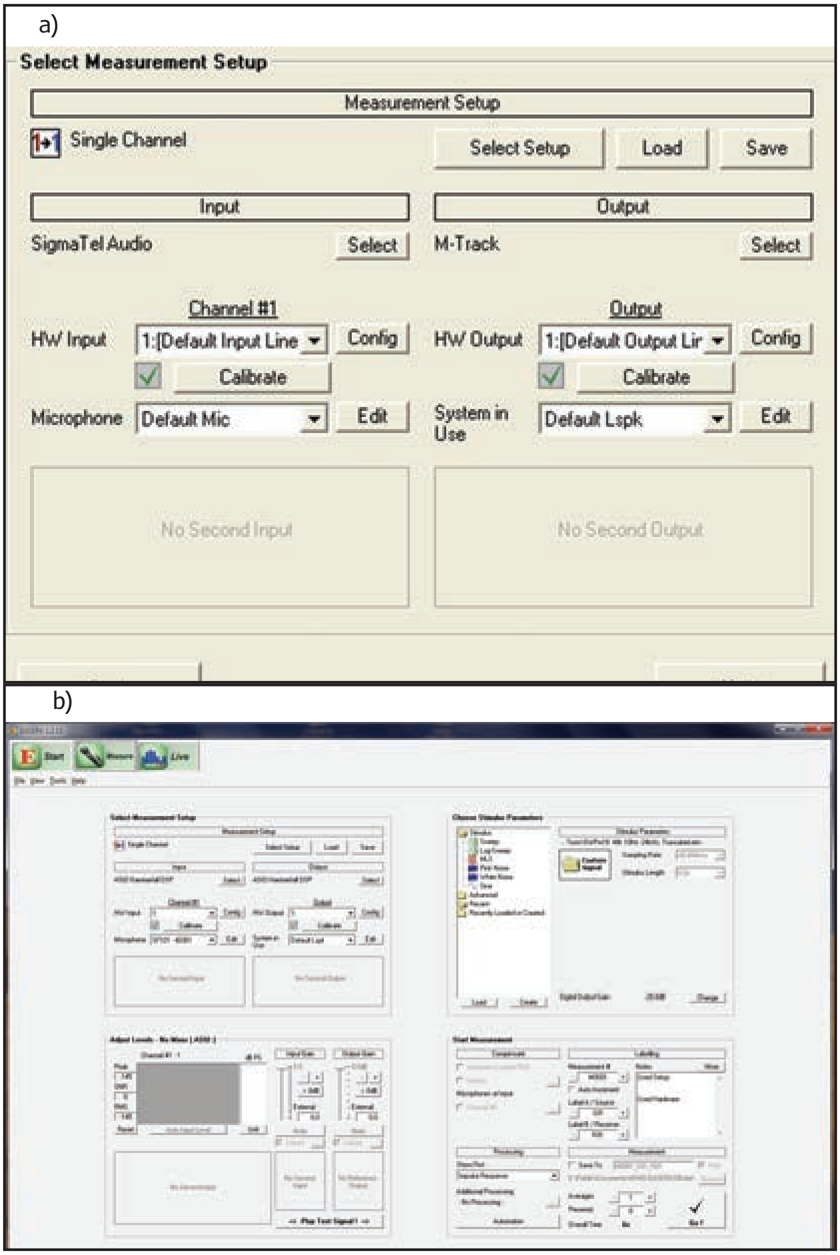


Photo 2a: From this screen, the user selects hardware devices. b: It is possible to configure EASERA so that all four of the sections in the Measure tab are displayed at the same time.

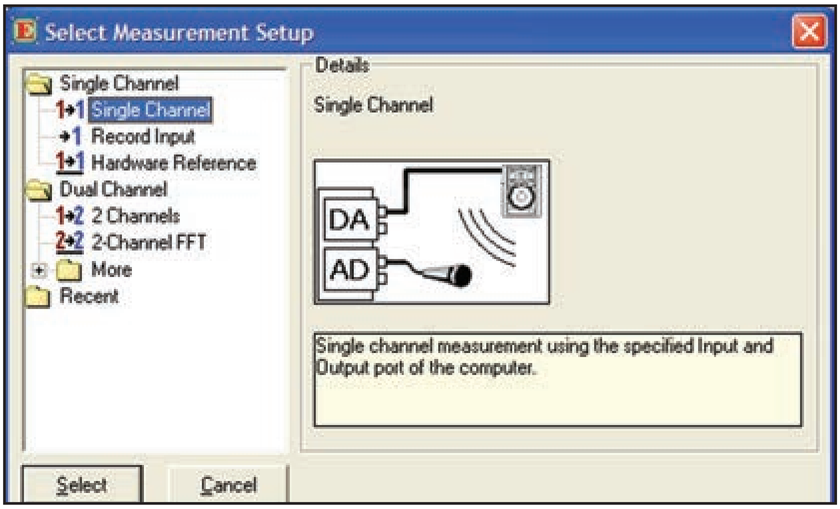


Photo 3: The user selects the overall setup of the measurement from this screen.

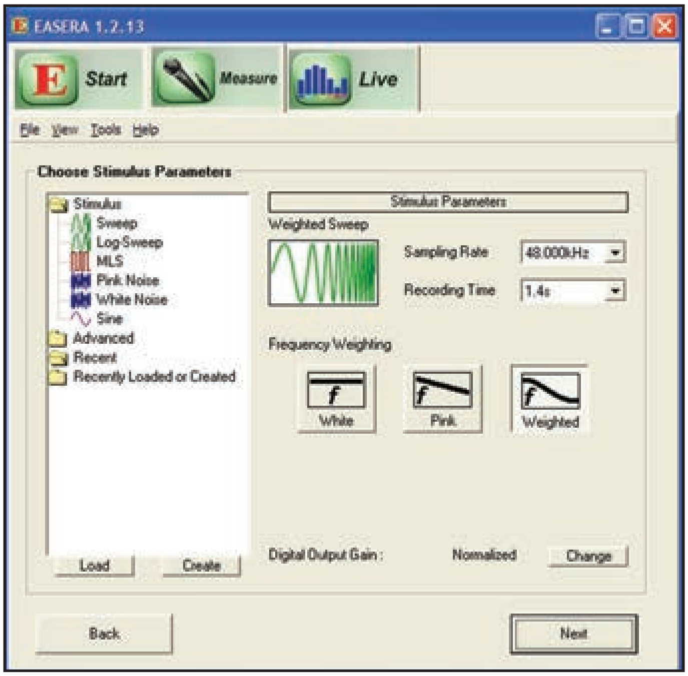


Photo 4: This screen provides control of EASERA's signal generator.

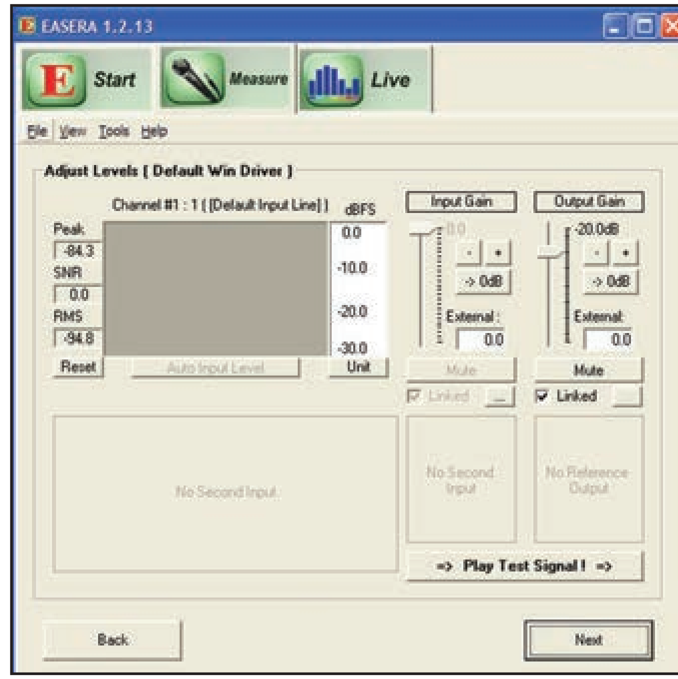


Photo 5: Input and output levels and gains are adjusted from this screen.

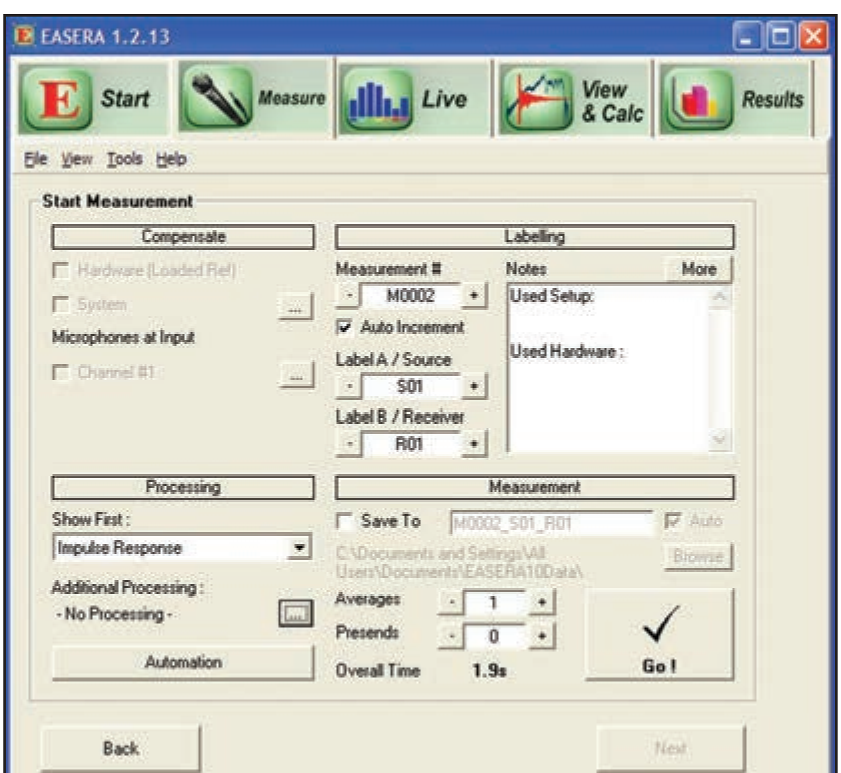


Photo 6: After reviewing the measurement setup, the user can execute the measurement from this screen.

additional notes to document things. Pressing Go executes the measurement. The duration of the measurement depends upon the time and number of averages chosen in the measurement setup. Once the measurement is complete, the results are immediately displayed in the View and Calc tab, as shown in **Photo 7**. If "Arrival, C50, D/R, SNR" is selected in the left-hand column, numerical data are shown in the left-hand column, with the impulse shown graphically in the right-hand column (see **Photo 7a**). The information displayed in the left-hand column exceeds the screen height of the computer I used, so the additional data (accessed by scrolling) is shown in **Photo 7b**. The first item in the left-hand column is the arrival time of the sound at the microphone, corresponding to the delay between the signal's creation and its reaching the microphone.

Clicking on the Graphs button at the left of the screen enables the display of measurement results. The data column at the left of the screen is replaced with a listing of specific measurements that can be displayed. A maximized version of this listing appears in **Photo 8**.

Photo 9 shows the decay lines that result from clicking Schroeder to display the Schroeder reverse integration of the decay curve. In the left-hand box, calculated overall EDT, T10, T20, and T30 values

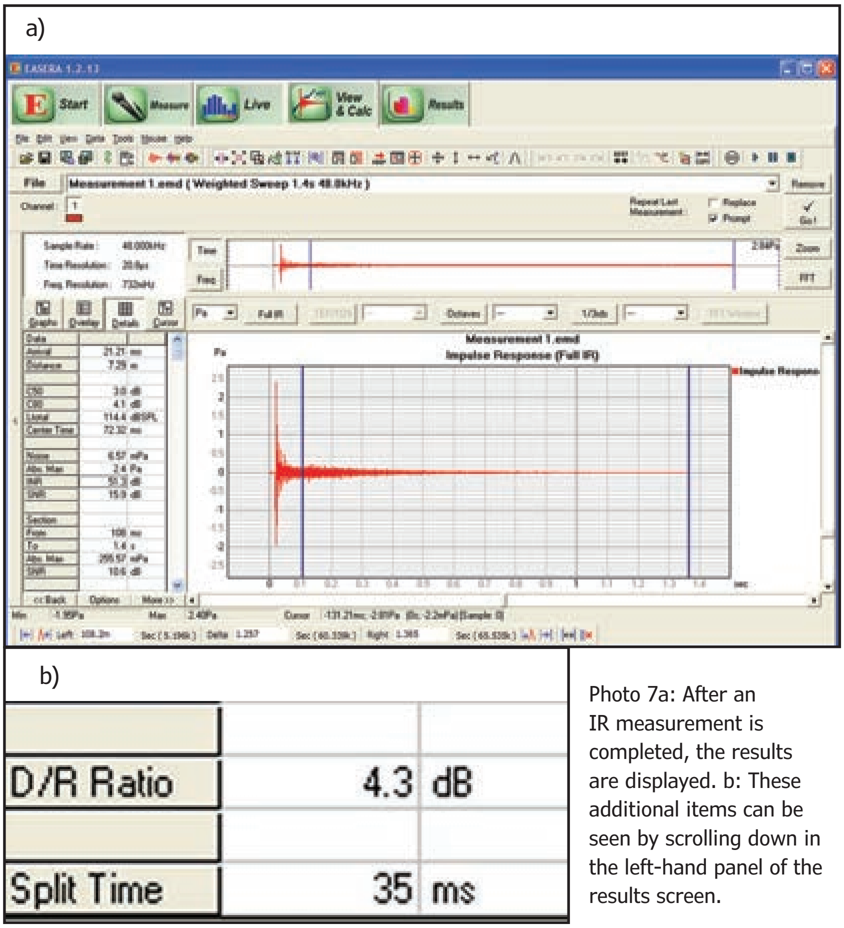


Photo 7a: After an IR measurement is completed, the results are displayed. b: These additional items can be seen by scrolling down in the left-hand panel of the results screen.

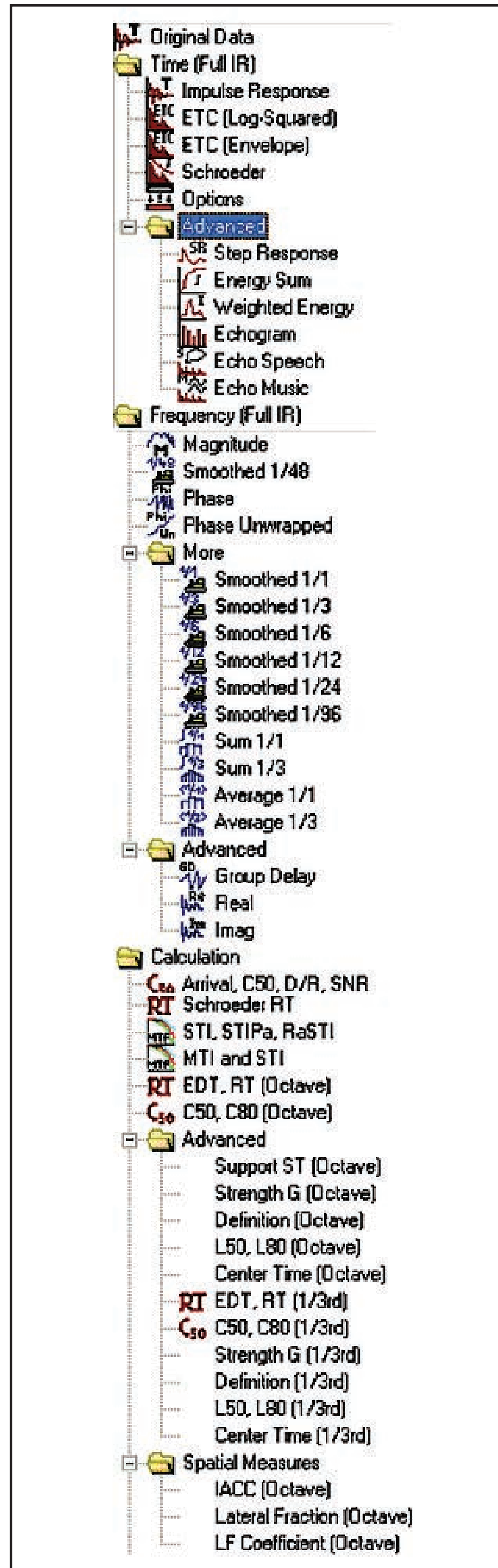


Photo 8: These calculated results are available for display.

are given numerically. More detailed frequency dependence of the RT can be seen by clicking on EDT, RT (Octave), upon which a graphical display of RT in octave bands is shown (see **Photo 10**). The overall average values of the RT parameters from 500 Hz to 4 kHz are also shown on this plot. On each graph, the span of the vertical and horizontal axes can be user-controlled by means of vertical and horizontal scroll bars, as well as by direct entry to edit the view limits.

As shown in **Photo 11**, the frequency dependence of the RT can be seen in higher resolution by clicking on the "1/3^{rds}" button on the Schroeder window. Overall numerical results appear in the left-hand column, and decay curves for each 1/3-octave band are shown. Alternatively, if the EDT, RT (1/3rd) button in the left-hand column is pressed, the EDT, T10, T20, and T30 values can be plotted versus frequency, with numerical results available in the left-hand column by pressing the Details button at the top of that column.

Speech intelligibility is most often quantified by the Speech Transmission Index (STI), which is calculated as the weighted sum of seven Modulation Transfer Indices (MTI), one for each octave frequency band from 125 Hz through 8 kHz, where each MTI value is derived from Modulation Transfer Function (MTF) values involving 14 different modulation frequencies, as described in the International Electrotechnical Commission standard IEC 60268–16.

The basic concept of this method of measuring speech intelligibility is that the source-to-receiver path may be characterized as a communication channel; and the speech signal, as a sinusoidally intensity-modulated wave having a certain modulation depth or percentage. At the listener position, room reflections, background noise, and perhaps electronic distortion artifacts tend to reduce the modulation depth. The MTF quantifies the extent to which the modulation is transferred from source to receiver, as a function of the modulation frequency, which ranges from 0.63 to 12.5 Hz.

By clicking on STI, STIPa, or RaSTI in the

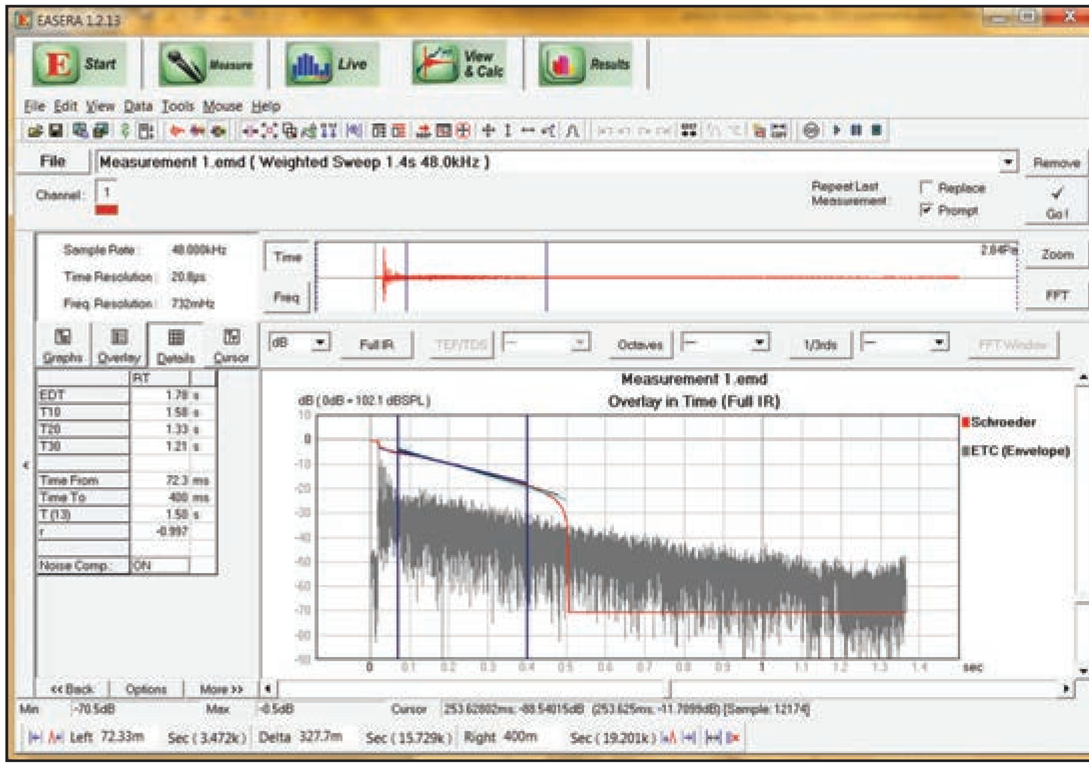


Photo 9: The Schroeder RT can be overlayed with the ETC Envelope.

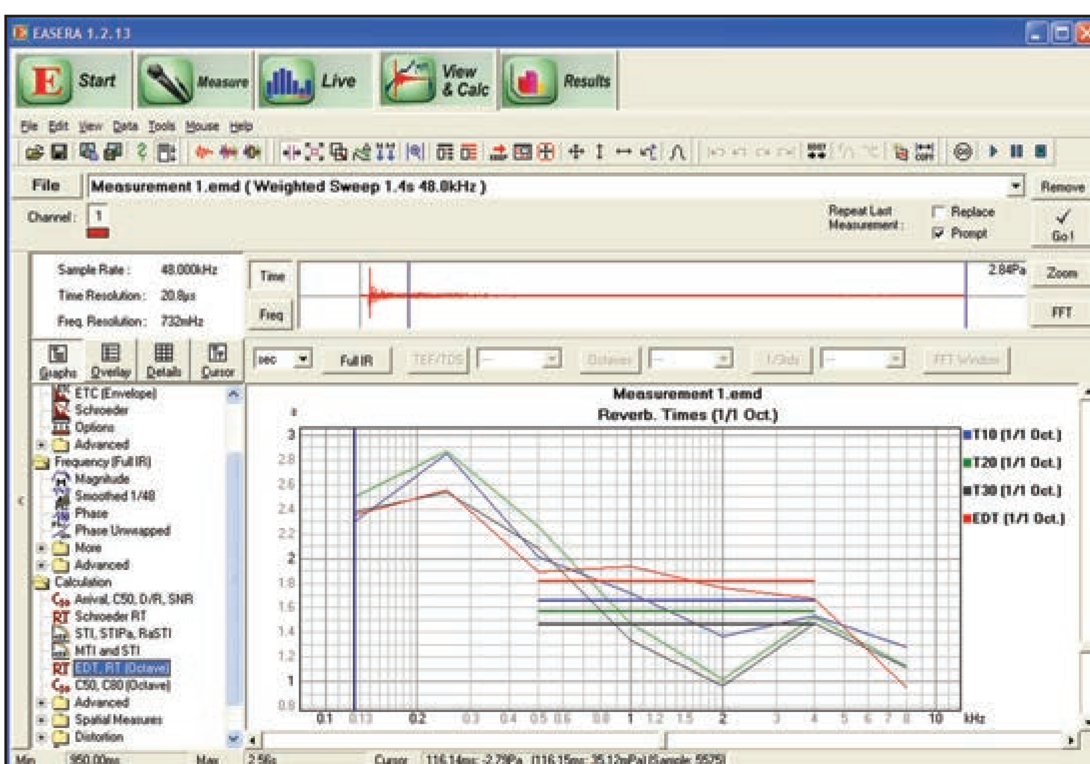


Photo 10: Octave-band RT can also be displayed.

left-hand column, the user can see the display shown in **Photo 12**. STI, STI (male), STI (female), RaSTI, and %ALcons (a different speech intelligibility metric) values are shown in the left-hand column, and MTF values versus modulating frequency for each of the seven octave bands are plotted. Clicking the More button brings up a complete table listing all the MTF and MTI values obtained in the calculation.

As an example of the many other architectural acoustics metrics that are available from EASERA, **Photo 13** shows a display of clarity measures, plotted by octave band. As with the other displays, numerical values can be shown by clicking in the Details button at the top of the left-hand column.

EASERA is one of the few measurement platforms capable of binaurally measuring Interaural Cross Correlation (IACC), which is involved in the perception of acoustical width and envelopment. With this inclusion, it certainly qualifies as one of the most complete computerized measurement platforms for architectural acoustics.

EASERA includes data-editing capabilities that are helpful for the advanced user. This review has not discussed the additional RTA and FFT capabilities that are useful in evaluating live and cinema sound systems.

EASERA Versions

EASERA is available in both a standard and a pro version. These differ in the size of the feature set. They are supported by an extensive downloadable tutorial, as well as a built-in Help function.

EASERA is also capable of live noise measurement overlaid on NC, PNC, and NR background curves. More complete noise analysis is available in the companion software SysTune, another AFMG product.

Also available from AFMG is the AUBION X.8, eight-channel data acquisition system. This unit has four integrated microphone preamplifiers and eight Mic/Line input ports and two Line/SPDIF output ports. It connects to the computer via a conventional gigabit Ethernet network switch.

More information about EASERA, as well as a downloadable fully functional, 30-day trial version can be found at http://easera.afmg.eu.

Resource

International Organization for Standardization (ISO), "Acoustics—Measurements of Room Acoustics," ISO standard 3382, www.iso.org/iso/catalogue_detail.htm?csnumber=36201.

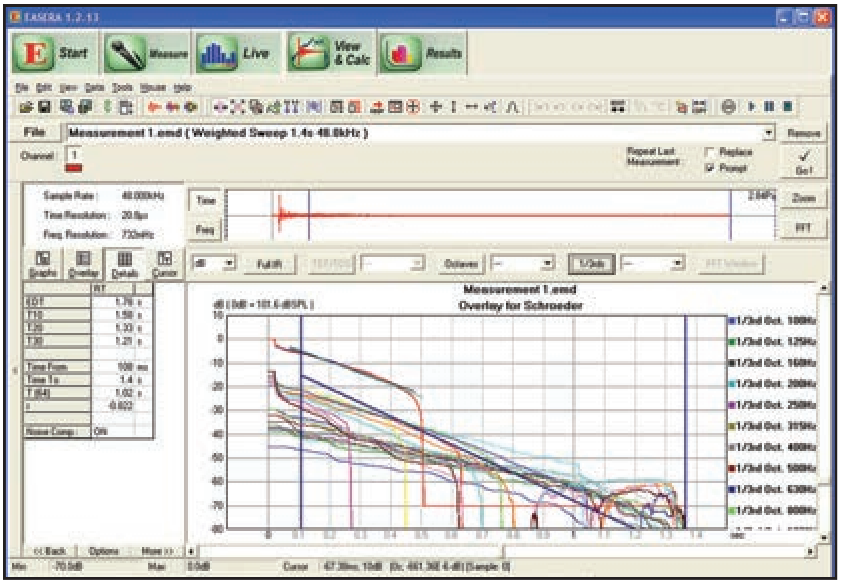


Photo 11: One-third-octave RT can be viewed numerically or graphically.

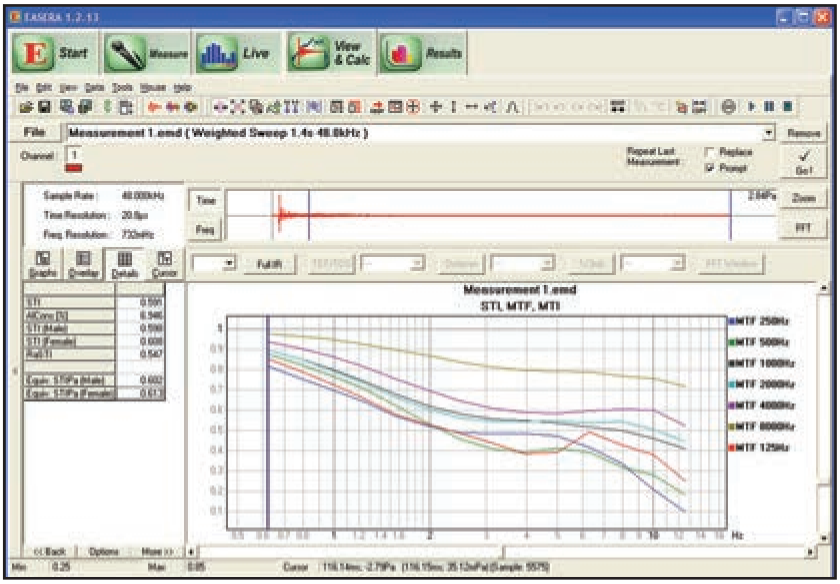


Photo 12: A full display of speech-intelligibility calculation results can be examined.

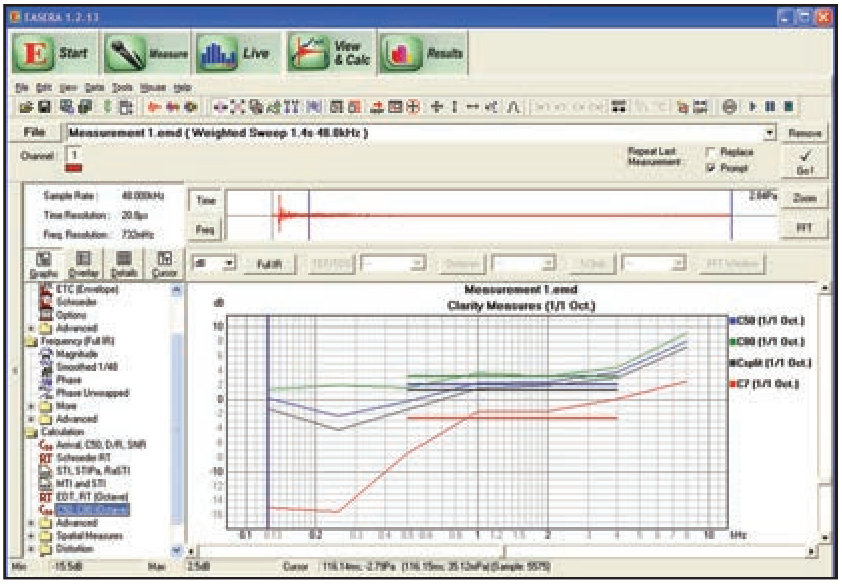


Photo 13: Clarity measures are an example of other acoustical values that can be shown by EASERA.

DIYer Takes Project to the Professional Level

An Interview with Kvart & Bølge Founder Bjørn Johannesen

By
Shannon Becker
(United States)

Bjørn Johannesen turned his amateur audio project into a burgeoning business.

SHANNON BECKER: Tell us a little about your background and how you became interested in audio electronics.

BJØRN JOHANNESEN: Born in 1950, I grew up when "stereophonic high-fidelity sound" was the new hot trend and I experienced the change in consumer audio. Radio broadcasts changed from AM to FM—and later on, to FM stereo. I still remember the first time I saw the stereo indicator light turn on our radio—and the first time I heard an LP in stereo. That was at a hi-fi exhibition. I was overwhelmed by this new and fantastic world of sound, which years later took a wrong turn due to the MP3 and poor loudspeakers, but that is another story.

Electronics and audio have always been among my favorite hobbies. I was not even a teenager when I started to assemble electronic kits. I soon learned how to make my own printed circuit boards for my amplifiers.

SHANNON: When and why did you start building loudspeakers?

BJØRN: When I was young, good commercial loudspeakers were out of reach, so I studied magazines and books about speaker building. In the late 1970s and the 1980s, there was a huge interest in building your own loudspeakers in Denmark. Denmark has a tradition for making loudspeakers and I think that had something to with this massive DIY movement.

I built my first loudspeaker when I was 14 years old, and I have built several loudspeakers over the years that were designed by others. This changed, however, when I learned about Martin J. King´s simulation software. Unfortunately, Martin no longer distributes his software, but you can still access articles on his website http://quarter-wave.com.

SHANNON: Tell us about your TABAQ project that was published in *audioXpress* in March 2007.

BJØRN: I have always built single-driver loudspeakers based on quarter wave in some form. Over the year, the loudspeakers got bigger and bigger, which seemed to be the wrong direction. Having understood Martin J. King's models, and thereby the nature of quarter wave design, I wanted to design a small loudspeaker using a 3" driver.

The goal was to design a loudspeaker that did not dominate the room, simply to build one with an acceptable bass suited for music listening.

I spent quite some time designing and simulating the speaker. The final result was close to the simulated prediction, and I was very pleased with it. The design is very simple: A straight line with the last part—which is formed like a port—narrowed down to half the area of the driver Sd. This is called mass loading, lowering the tuning of the cabinet.

I contacted *audioXpress* and the editors found my TABAQ (Tang Band Quarter Wave) design of interest. After the TABAQ project was published, I immediately got a lot of response. I then started a thread on diyaudio.com that has since reached more than 240,000 views.

Originally designed for a specific Tang Band 3" driver, I have discovered that several other drivers can be used in the TABAQ design. The only adjustments I would make to the original design is that I found out you can mount the driver 5 cm higher up, if you like. I have also built a folded version of the TABAQ design and made a modified cabinet to fit the Tang Band 2" driver.

This is a photo of the completed TABAQ project, which was published in *audioXpress,* March 2007.

This was the prototype for my adjusted TABAQ speakers that became Kvart & Bølge.

Here is the casting being poured for the loudspeaker's base.

SHANNON: Do you have plans to write additional articles for *audioXpress*?

BJØRN: I have a few projects that might be of general interest. My original TABAQ speakers are still playing in my living room, mounted with 4" Tang Band quarter waves. The cabinets could use a makeover after all these years, and I am considering using black glass for the front and top. If I manage this, I think the project might be of general interest.

For many, quarter wave—or transmission line (TL) designs—are still a mystery. I would also like to update my old article on the subject (www.coolcat.dk/bjoern/QWforDummies.pdf).

SHANNON: Why do you think it became so popular in the DIY circles?

BJØRN: My TABAQ design is very simple to build and I think my instructions make it straightforward. Furthermore, a suite of drivers can be used, both 3" and 4" drivers. The 3" drivers should have a Fs of max 105 Hz and a Qts of 0.6 or higher. For the 4" drivers the Fs should be below 70 Hz and the Qts, higher than 0.35.

SHANNON: Was this the first DIY audio project that you had published?

BJØRN: Actually my first-ever article was in *audioXpress,* July 2006, "The Invisible Hide Away Sub," which was followed up with an article I wrote on a redesign of this project in *audioXpress,* October 2007. This subwoofer is still in use, with the Scan-Speak 26W/4558T00 driver and a Hypex 1.2 Class-D amplifier module. The result is even deeper more controlled bass.

SHANNON: Tell us about your decision to turn this design into a business, Kvart & Bølge, Inc. (www.kvart-bolge.com).

BJØRN: The idea came from my friend Arved Deecke, who is a professional businessman and entrepreneur. My role was to be an audio consultant and Arved Deecke would take care of the production and marketing. I liked the idea of sharing my design and launching a commercial product. This was, of course, something that had been in my mind for some time, but I could not do it alone. Arved presented his vision at the right time. There are only a few commercial quarter wave loudspeakers available and they have a high price tag. We wanted to market a high-quality quarter wave speaker with at an affordable price. I think we managed that.

SHANNON: What were some of the challenges involved with starting your own business venture?

BJØRN: We did not want to market an "Assembled TABAQ for sale." Therefore, the name was important. We came up with Kvart & Bølge as this means Quarter & Wave in Norwegian and Danish. I was born in Norway, but have been living my adult life in Denmark.

Of course, there had to be made an agreement between the parties involved to take care of the commercial terms and intellectual rights to the design. We managed this with no problems. With Martin J. King, we entered a commercial license for his simulation software. So the paperwork was the easy part. Arved was responsible for the physical design, based on my acoustic design, and he spent quite some time finding the right manufacturing method as well as the right supplier.

SHANNON: What modifications were needed to commercialize the design?

BJØRN: First of all, we had to find a small driver with the right properties. The Tang Band driver I used in my TABAQ project is no longer available. We

Folded aluminum profiles are used for the speaker cabinet.

The aluminum profiles are being cut to size during the production process.

Here is the finished speaker with veneer.

Bjørn Johannesen is shown with his Kvart & Bølge speakers.

spent months testing several candidates. Eventually, we ended up with a high-excursion neodymium Tymphany full-range driver—designed in Denmark, of course.

We wanted to minimize the size of the cabinet without sacrificing the bass. We performed several simulations and made prototypes to ensure the bass was intact. Compared to the TABAQ, the volume, port, and tuning frequency had to be adjusted to get the best out of this driver. We also adjusted the stuffing after several test runs, but it does not differ very much from the simulation. We kept the front baffle of the cabinet narrow and the cabinet itself has a parabolic shape.

For manufacturing reasons, we wanted a casted foot with an integrated port. The cabinet can be finished with veneer, paint, or the customer´s own design. We selected aluminum for the cabinet to meet the flexible design goals.

SHANNON: Please explain your company's tag line Sound Sommeliers.

BJØRN: We borrowed the term "Sommeliers" from the vineyard world: A trained and knowledgeable wine professional knowing all aspects of wine service. Kvart & Bølge is doing the same thing, only with sound—targeting an audience who loves music as opposed to listening to cheap PC and Bluetooth speakers. Having said this, Kvart & Bølge products make excellent PC and TV speakers—and they are not expensive.

SHANNON: What is next for Kvart & Bølge?

BJØRN: There is a demand for bookshelf loudspeakers and soundbars. We get requests in these areas all the time and are, of course, considering our options.

SHANNON: Do you have any advice for *audioXpress* readers who are considering their next project?

BJØRN: For a first timer, I suggest choosing a well-proven design. There are a lot of them in the DIY community and everybody has their own favorite. A full-range (singe-driver) quarter wave loudspeaker is a relatively inexpensive solution and will often give you a good result. Combine it with a subwoofer if you need more/deeper bass.

I recommend a visit to Martin J. King's website, which I mentioned earlier. You can also find some general audio information on our company's website, www.kvart-bolge.com.

Editor's Note: You can read the original TABAQ project article online, as published by audioXpress *in March 2007 at: http://audioxpress.com/article/TABAQ-Tang-Band-Quarter-Wave.html*

The Kvart & Bølge speaker is shown here in various color versions.

The Right Filter (Part 3)

Phase Corrector 2 Brings Us Closer to the Ideal Crossover

As you may recall, I wanted to build a tunable near-linear phase analog crossover, using analog components. In the first two articles, I explained the method I used to get phase linearity over audio band, the hardware and briefly justified my design choices. In this final article, I will describe a second phase corrector for this design.

Photo 1: After a few attempts, I was able to perfect my tunable near-linear phase analog crossover, using analog components.

By
Vincent Thiernesse
(France)

Photo 2: This project is designed for a 2.5-kHz crossover.

After some trial-and-error, I believe I found the “right filter” for my tunable near-linear phase analog crossover (see **Photo 1**). Whereas the first phase corrector, which I described in my July *audioXpress* article, focused on engineering with its crossover frequency tuning capability, this second phase corrector focuses on quality. It is designed to treat signals that are not limited in the audio band, so it will exhibit a better transient response. The drawback is that the second phase corrector is dedicated to only one frequency cut-off while the first phase corrector could work on a wide crossover-frequency range. The one I describe here is designed for a 2.5-kHz crossover (see **Photo 2**).

Phase corrector 1, which has a fixed frequency, provides better results in terms of voltage swing and is a little better in terms of harmonic distortion, noise being equal. But you might want a better transient response if the source is not 20-kHz limited, as it happens to be with most digital audio sources (e.g., a CD player, MP3 player, etc.), which is why I tried to synthesize a second phase corrector that would provide an abrupt attenuation after 20 kHz.

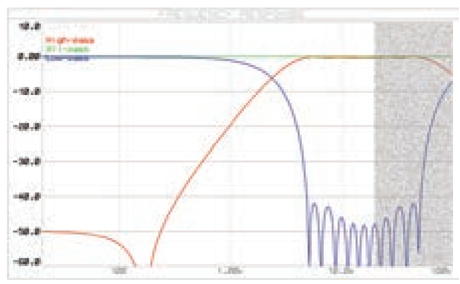
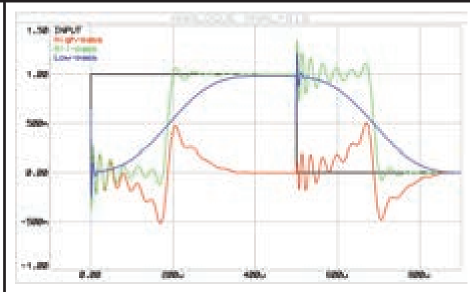
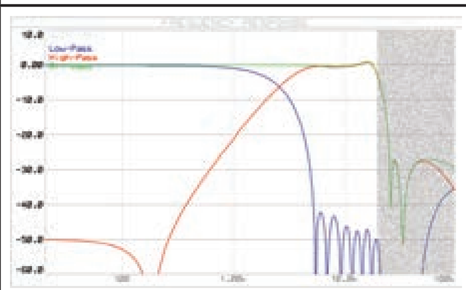
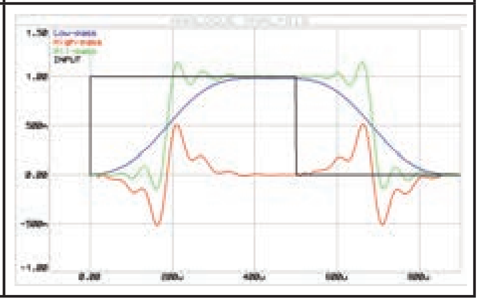
PROTEUS SIMULATIONS	Frequency response	Step response
Corrector 1 (tunable over 1 octave)		
Corrector 2 2.5-kHz fixed crossover frequency (Additional first-order poles omitted)		

Table 1: The frequency response and step response of phase corrector 1 and phase corrector 2 are compared.

But, I didn't want this phase corrector to increase complexity. So, I replaced the first phase corrector with a low-pass filter with phase shift that applies ad-hoc phase correction to the filter. The results are as good as expected. The only remaining problem is a few Gibbs oscillations due to abrupt filtering. However, it is necessary to attenuate before the following phase distortion or you will get a steep roll-off. You get phase distortion at pulses for which:

$$\frac{\partial^2\varphi}{\partial\omega^2} \neq 0$$

and:

$$\left|\omega^2 \times \frac{\partial^2\varphi}{\partial\omega^2}\right|$$

could be a reliable quantitative indicator of it. However, the ripples you get on output when you have phase distortion are smaller when the phase distortion is abrupt.

Let's consider a unity gain phasing system. Its step response is made of oscillations before the output steps. We can observe that:

- There may be N ripples before/after a delayed step for a N × 2π phasing system. You only get them before if:

$$\frac{\partial^2\varphi}{\partial\omega^2}$$

is monotonous.

- These oscillations are a kind of burst.
- Frequencies of this burst cover the band where:

$$\frac{\partial^2\varphi}{\partial\omega^2}$$

is not null (i.e., where there is phase distortion).

Phase Corrector 2 Parts List

Quantity	Parts References	Value
	Capacitors	
4	C1–C4	1 nf/5%
2	C5, C6	100 nf
2	C7, C8	47 µF/5%
4	C10–C13	10 pF
	Integrated Circuits (ICs)	
1	U1	NE5531
2	U2, U3	TL082
	Resistors	
12	R1, R2, R5–R10, R12, R19, R20	100 kΩ
1	R3	2.43 kΩ
1	R4	2.32 kΩ
1	R11	2.7 kΩ
1	R14	1.8 kΩ
1	R15	16.13 kΩ
1	R16	3.48 kΩ
1	R17	7.98 kΩ
1	R18	1.3 kΩ

Table 2: The parts used for the second phase corrector are listed, along with their values.

- Amplitude of these ripples is also in relation to:

$$\frac{\partial^2\varphi}{\partial\omega^2}$$

- Ripples integrate $t_{DELAY}(\omega)$ domain.
- Delayed step occurs at:

$$T_{delay} = \left.\frac{\partial\varphi}{\partial\omega}\right|_{\omega=0}.$$

This predictive formula is an attempt to summarize qualitative observances:

$$u(t) \approx \frac{E}{g(\omega_i)} \times \cos\left(\int_{t=0}^{T_{delay}} \omega_i(t)\,dt + \frac{\pi}{2} \times \left(1 - g(\omega_i)\right) \right)$$

where:

$$\begin{cases} \omega_i(t) = f^{-1}\left(t = \dfrac{\partial\varphi}{\partial\omega}\right) \quad \textit{with} \quad \dfrac{\partial\varphi}{\partial\omega} = f(\omega) \quad \textit{and} \quad T_{delay} = \left.\dfrac{\partial\varphi}{\partial\omega}\right|_{\omega=0} \\ g(\omega_i) = \sqrt{1 - 2 \times \omega_i^2(t) \times \left.\dfrac{\partial^2\varphi}{\partial\omega^2}\right|_{\omega=\omega_i(t)}} \end{cases}$$

f^{-1} is non-injective in case $\frac{\partial t}{\partial\omega} = \frac{\partial^2\varphi}{\partial\omega^2}$ changes of sign over $\omega \in [0;+\infty[$ (i.e., if it is not monotonous).

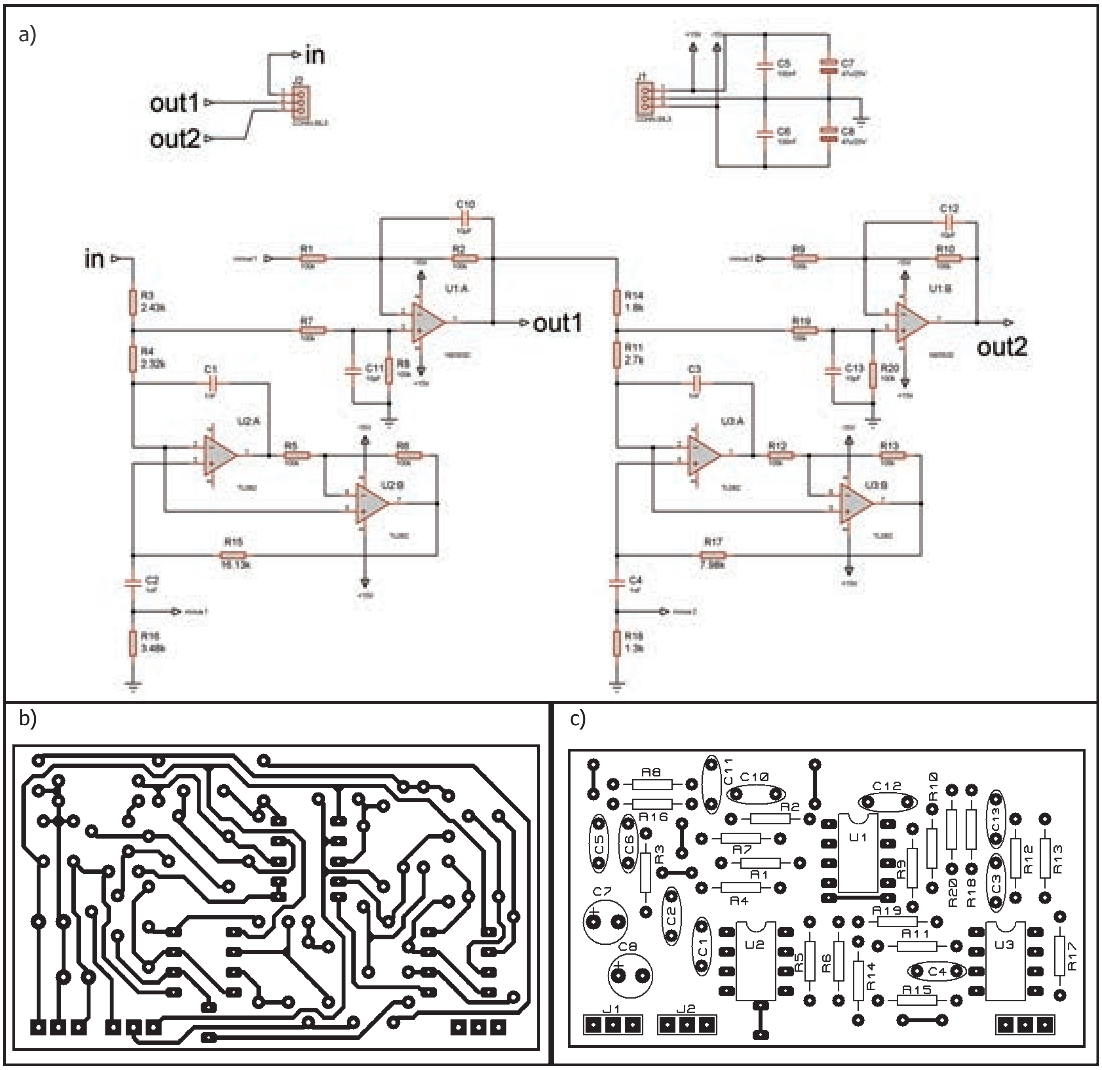


Figure 1: This schematic (a) and the PCB's front (b) and back (c) for the second phase corrector is shown. The dimensions are 2.625" × 1.5".

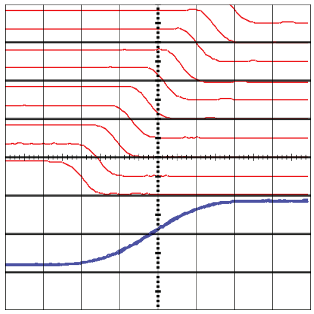
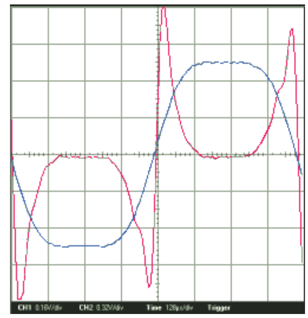
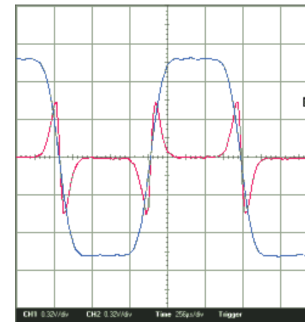
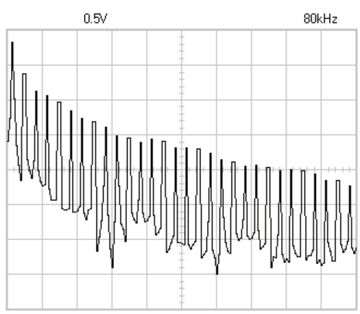
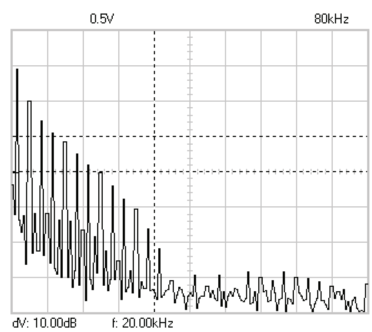
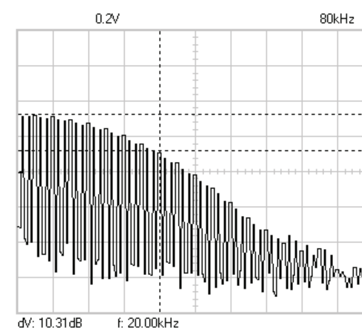
Various Testing Waves Measurements			
Test procedure elements			
	Test points 1…10 Input is a square LP output in blue	First overshoot is 18% higher than the second one (high-pass output): => R1-R10,R12,R13 must be ~18/3 = 6% smaller => ~ output symmetry.	
Test signals	Square	Convoluted square	Gaussian impulse
			

Table 3: Various test wave measurements are shown. For the test points 1–10, the input is a square low-pass output shown in blue. The first overshoot is 18% higher than the high-pass output => R1-R10, R12, and R13 must be ~18/3 = 6% smaller => ~ output symmetry.

Other variations of this formula are being explored, but this formula is sufficiently reliable for my purpose that concerns pre-step ripples.

As I said in previous articles, there are at least three solutions:

- Additive phase-linear 20-kHz low-pass filter
- Increase phase corrector complexity as it:
  - Extends phase linearity band
  - Reduces wave amplitude because of more abrupt time-delay roll-off
- Mix of a phase corrector and low-pass filter

I chose the third solution because it doesn't increase the project's material complexity.

Phase Corrector 2 Simulation

The way the second phase corrector is designed is somewhat intuitive. I've been placing transmission zeros after 20 kHz in order to get an abrupt gain-attenuation. The first zero is associated with a low damped denominator so that the gain roll-off associated to first zero is narrow. The second zero is wider. The more you place a zero above 20 kHz, the more wide it can be. Wide zeros will manage to correct low-order Taylor coefficients of phase distortion whereas narrow zeros manage to correct higher-order coefficients.

Two zeros are enough to ensure satisfying rejection (greater than 25 dB at 25 kHz). An additive pole is required to avoid gain overshoot. It should be placed after the pseudo-digital filter, rather than before, so the phase corrector slightly pre-amplifies the treble and you get a little better signal-noise ratio (SNR). Once you accept this method and apply it, you just have to adjust your phase corrector, giving more importance to phase linearity than to a 0.3 dB gain overshoot at 18 kHz.

The phase distortion has an exponential-like shape until 15 kHz. Consequently, this second phase corrector does conduct low damped phasor cells just as the first phase corrector did. The additional pole adds the phase you miss because of the non

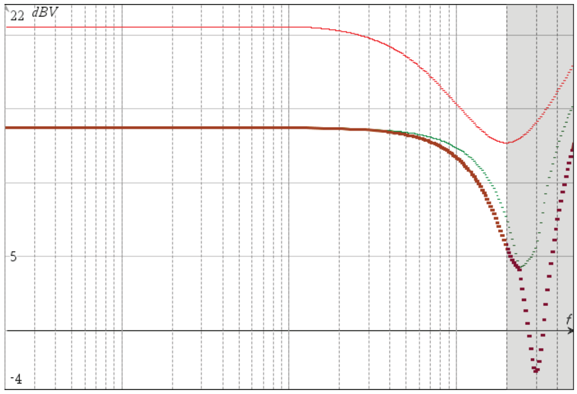
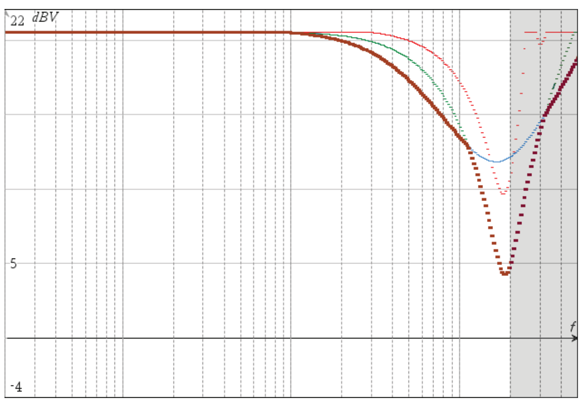
Output Swing	
Phase Corrector 1 (3 kHz fixed frequency board)	
Phase Corrector 2	

Table 4: The voltage-swing limitations are shown for the phasor cells (blue), the corrector's phasor cell 1 (green), the corrector's phasor cell 2 (red), and the overall limitation (brown).

exponential-like shape beyond 15 kHz.

Finally phase distortion remains greater than 10° until 20 kHz and is about 25° at first transmission zero (total phase 360 *8). Thus, phase linearity is ensured in almost any condition (see the symmetric wave forms in **Table 1**). Here, the frequency response of phase corrector 2 is close to being the "perfect filter." We expect time delay ripples to be lower than 2 µs.

Concretely, this phase corrector is made of five poles and two double zeros (see **Figure 1**). Thus, its topology cannot be the same as phase corrector 1. It uses R,L,C series filters with an active inductor—R//C),L was used for the first phase corrector. I wish it was tunable but it is not yet because I also have to change the frequency tuning board (C2 and C4 should be on this new board).

It may be used with a 2.5-kHz fixed frequency board, but the resistors are not calculated the same. If you want another crossover-frequency, both boards have to be changed. Initially, I wanted a 3-kHz crossover-frequency (see **Table 2**). The simulations were good, I tried to be more ambitious with a 2.5-kHz frequency and succeeded, but it was more difficult.

The new values for the fixed frequency board would require 10 resistors—R1—R10 at 14.7 kΩ and 10 resistors R21—R30 at 22 kΩ. A method to adjust these resistors is shown in **Table 3**. If you

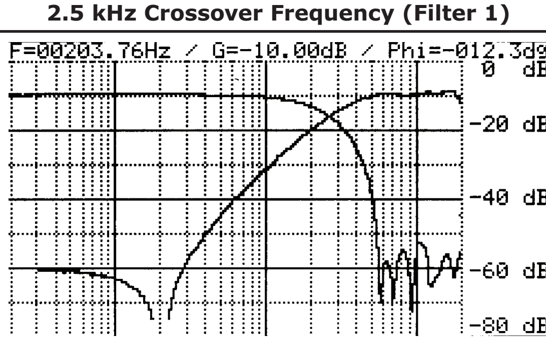
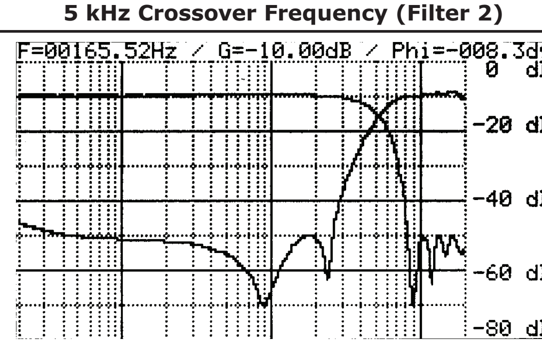
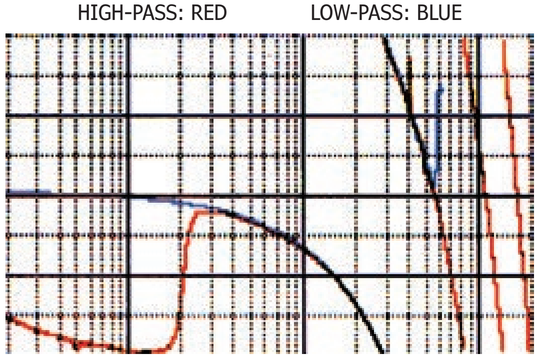
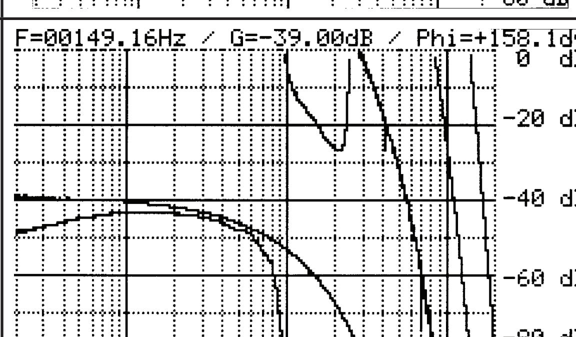
	2.5 kHz Crossover Frequency (Filter 1)	5 kHz Crossover Frequency (Filter 2)
Frequency Response		
Phase Response: High Pass + Low Pass	HIGH-PASS: RED LOW-PASS: BLUE 	

Table 5: For the frequency domain measurements, the low-pass and the high-pass share the same phase across the band. With the first filter, the response is 200 Hz to 6 kHz. With the second filter, the response is 1.5 kHz to 8 kHz. The low-pass phase acquisition stopped at the first zero for graph clarity. (Images courtesy of a home-designed Bode plotter)

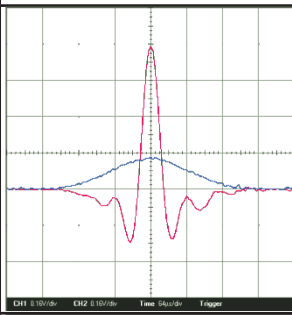
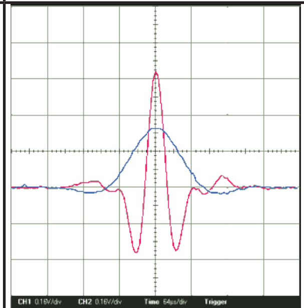
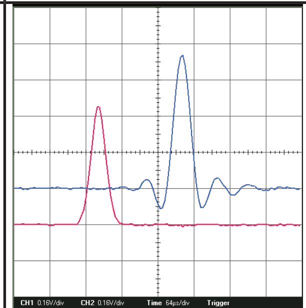
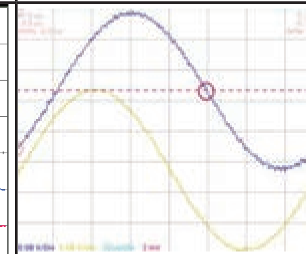
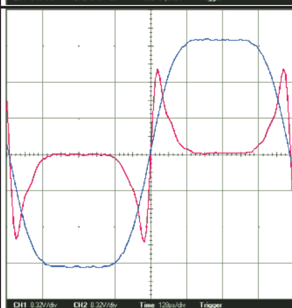
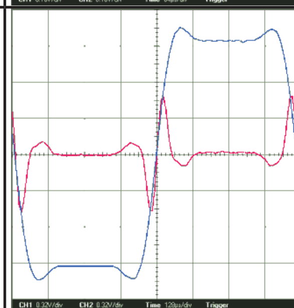
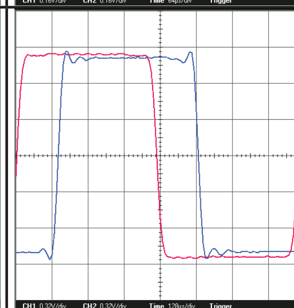
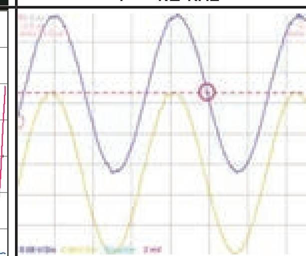
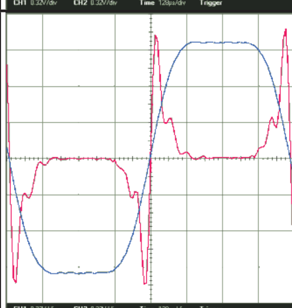
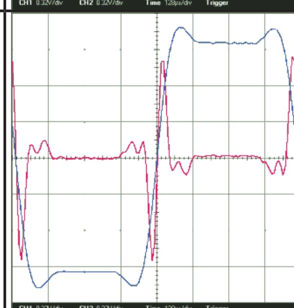
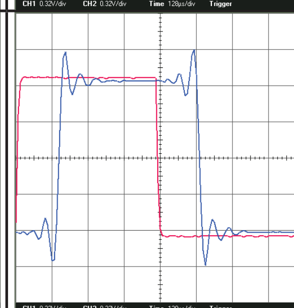
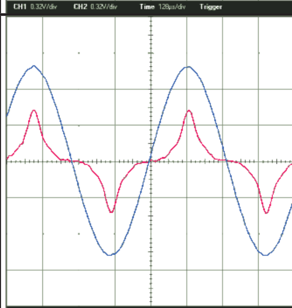
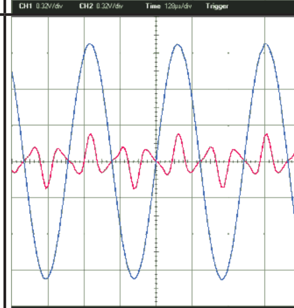
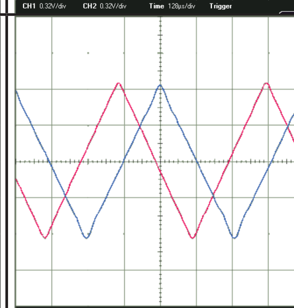
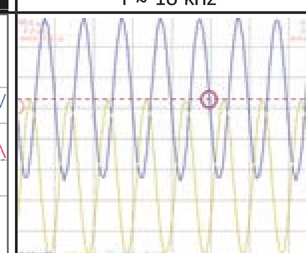
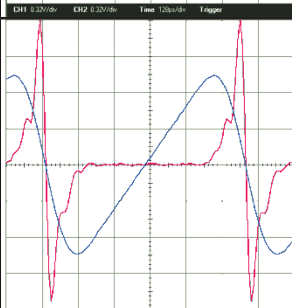
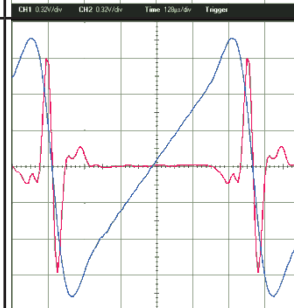
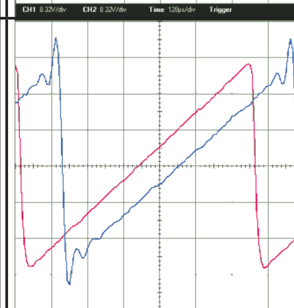
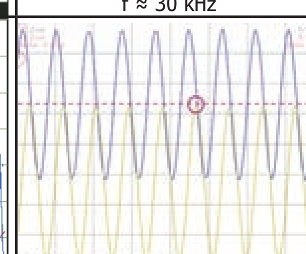
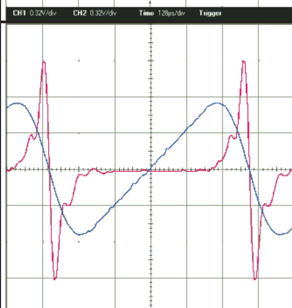
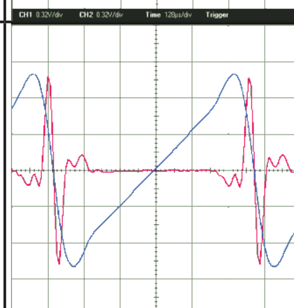
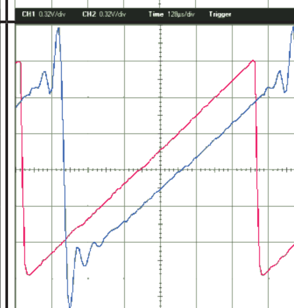
20.5 kHz Crossover Frequency	Filter 1	Filter 2	Low Pass + High Pass (common for both filters)	Time Delay Flatness
Impulse response				 f ≈ 4.2 kHz
Square response (square convoluted with Gaussian)				 f ≈ 10 kHz
Square response				 f ≈ 18 kHz
Triangle response				 f ≈ 30 kHz
Saw response (square convoluted with Gaussian)				 f ≈ 36 kHz
Saw response				The time delay variation of 3-kHz fixed crossover frequency with phase corrector 1 is less than 2 µs until more than 30 kHz with a 3-kHz crossover frequency. The result is the same for phase corrector 2 until 2 kHz.

Table 6: The time domain responses, including the low pass + high pass and the time delay flatness, are shown for filter 1 and filter 2. The red trace shows the input and the blue trace shows the output.

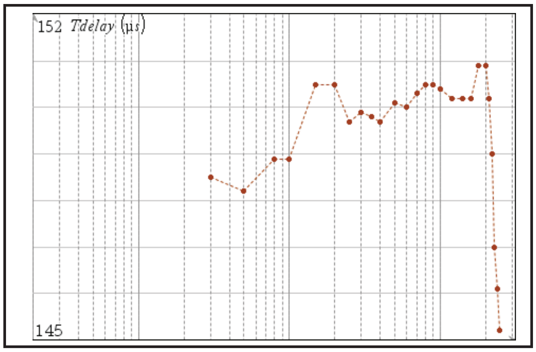


Figure 2: The time delay flatness measurement is shown using phase corrector 2.

need a 2-µs time-delay ripple instead of 4, this is an example of what you could get at first power-on. Also, the capacitor on the main board has to be changed: C19 and C119 = 1.2 nF instead of 470 pF.

Output Voltage Swing Issue

Voltage swing gets limited because of the high-quality phasor cells, especially those of phase correctors (see **Table 4**). Resonance will happen at the output of U2:B and U3:B. Saturation of these

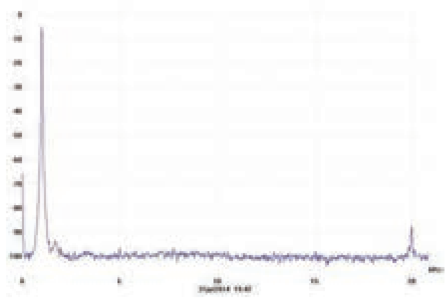
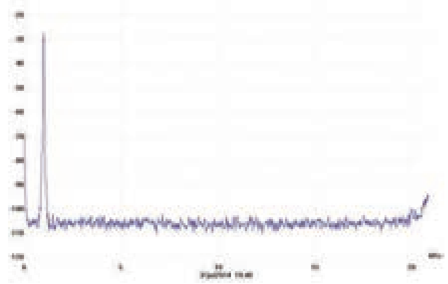
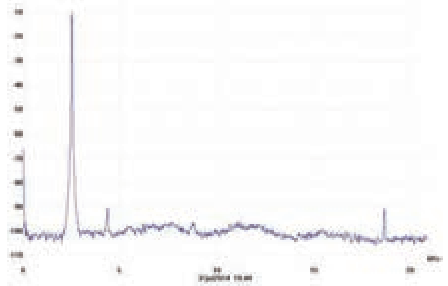
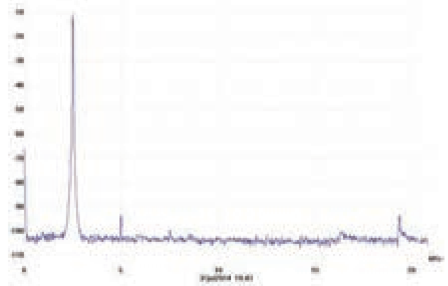
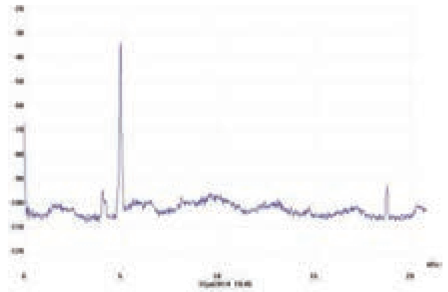
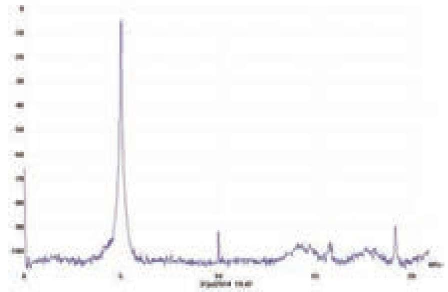
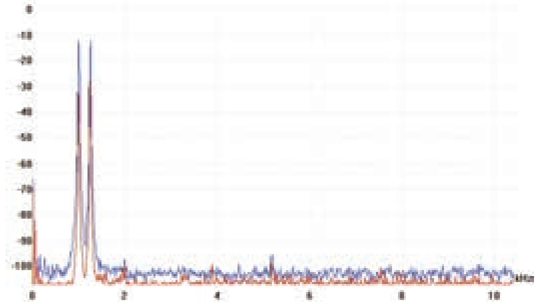
	Low-Pass Output	High-Pass Output
F = 1 kHz		
F = 2.5 kHz		
F = 5 kHz		
Intermodulation		
Spectrum acquisitions: PicoScope ADC-216. Generator: Motorola Photon Q.		

Table 7: The distortion and intermodulation measurements are shown for the low-pass and high-pass outputs at 5 kHz, 2.5 kHz, and 1 kHz.

op-amps causes distortion. Nevertheless, this is a smooth kind of distortion. Phase corrector 2 is less affected by this limitation than phase corrector 1, which has higher Q factors because of low-pass attenuation. Two cells can be sorted in two ways, so the ones of phase corrector 2 are sorted to give the best results. This voltage-swing limitation criterion imposes no special phasor cells sorting for phase corrector 1, because of their unity gain.

Phase corrector 1 provides more restrictive limitation and is even –15 dB worse with a tunable frequency board. So the limitation could be as low as 125 mV RMS (approximately –18 dBV) in the worse case and at the worse frequency, which is about 10 times the crossover frequency. Then, as soon as crossover frequency exceeds 2 kHz, the voltage-swing limitation becomes significantly less restrictive.

The last point is that phase corrector 2 requires an additional pole. This pole may be placed before or after the filter. Placing it before the filter will provide better voltage swing than the presented curves, approximately a 9 dBV (2.8 V RMS) limitation above 19 kHz, and placing it after the filter will give better noise level…but not better SNR because the voltage-swing limitation is worse.

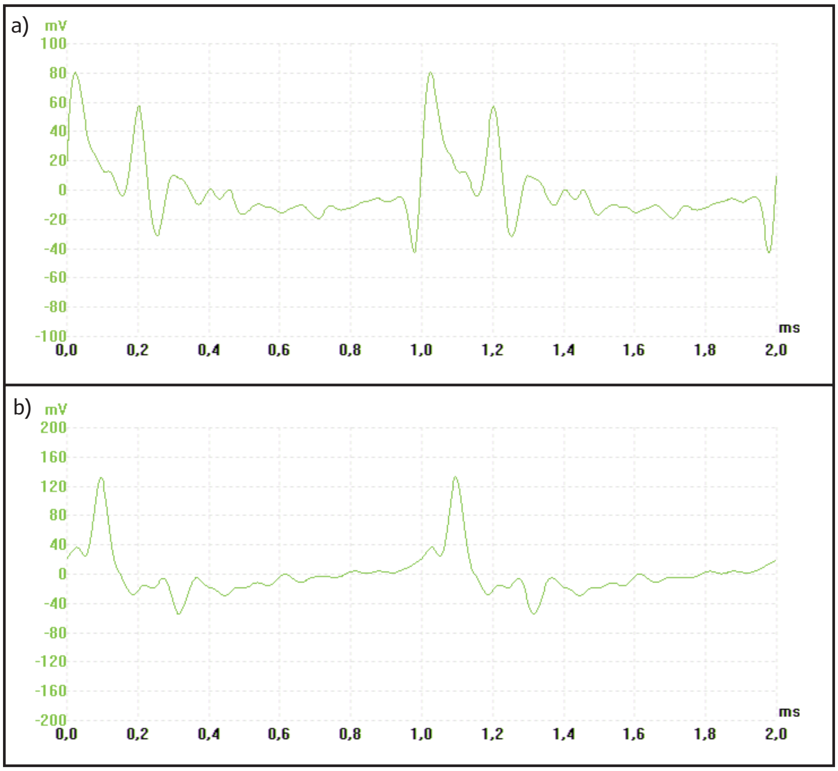


Figure 3: The impulse response was measured using a conventional filter (a) and the "right filter" (b).

Frequency Domain Measurements

You don't need to worry about the curves shown in **Table 5**. Although the first prototype used two 5% weighing resistors (R10-20 and R31-39), it matched the specifications. You may tune RV1 and RV2 to get precise rejection or keep them in middle position. Note that both phases are the same in transition band as expected.

Time Domain Measurements

The time domain measurements are very close to the simulations shown in **Table 6**. They may require small adjustments to get perfect symmetry.

The time delay ripples for the second phase corrector are shown in **Figure 2**. And, it is not surprising that the time delay flatness measurement is not as good as it was the phase corrector 1.

Noise and Distortion

The distortion and intermodulation measurements (IM) shown in **Table 7** used a total harmonic distortion (THD) at less than 0.001% at 1 kHz. At higher frequencies, the spectrum exhibited a little bit of harmonic distortion. Nevertheless the THD remained lower than 0.01%. Though my measurements are not precise enough to determine the exact THD, I thought I should also measure the intermodulation ratio, just to make sure that the THD didn't hide any surprises. It appears the intermodulation is lower than 0.003%.

You may also note some small external disturbances at approximately 4.2 kHz and 18 kHz, which are not harmonics. They correspond to a close-by 60 kHz interference from a nearby switching power supply that I could not suppress during my measurements.

For noise and output voltage swing we want a (S/N)dB greater than 100 dB. To achieve that, you can use higher quality op-amps than NE5532 and TL082. But even with these modest op-amps we have been able to get good results.

You probably won't need to adjust the inputs/outputs for most of common applications. They may

Project Files

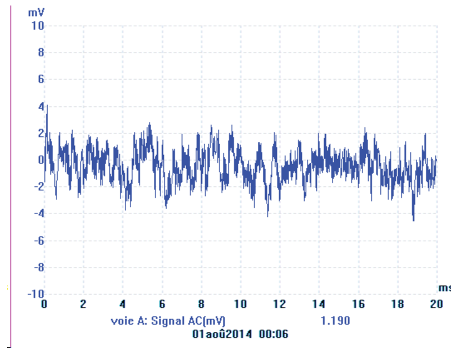
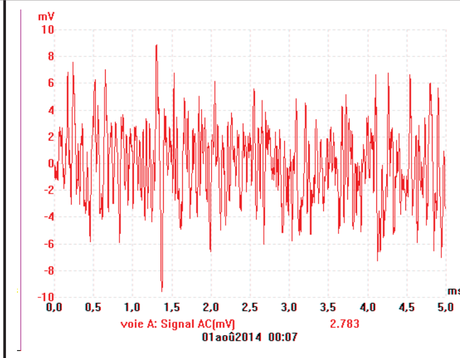
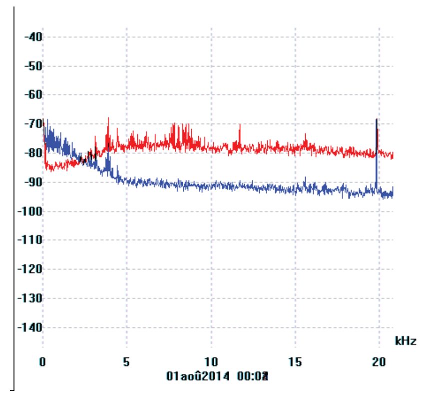
To download additional material and files, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html.

Resource

V. Thiernesse, "The Right Filter," *audioXpress*, June 2015.

——, "The Right Filter (Part 2): Crossover Frequency Tuning," *audioXpress,* July 2015.

Table 8: The noise measurements are shown for the low-pass and the high-pass outputs.

Noise (×100)	Low-Pass Output	High-Pass Output
Time domain		
Frequency domain		Total noise: ~25 µV RMS If you add both contributions, it is close to a white noise.
Spectrum acquisitions: PicoScope ADC-216. Measured via a ×100 amplifier.		

even be directly coupled to amplifiers. To complete our measurements, I tested the impulse response of an IQ13 with a conventional filter and with the right filter to show how useful this design can be for speaker designers (see **Figure 3**). Looking at those measurements, you can see that the impulse response's phase linearity is significantly enhanced.

Overall Impressions

Small modifications of the phase correctors (different component values for phase corrector 2 and structural modifications for phase corrector 1) will provide improved worse-case voltage-swing limitation of approximately 15 dB. The same +15 dB enhancement is expected with the SNR.

As for how much aspect-oriented programming the signal will go through: considering that the important signal for high-pass is the fifth phasor output, the answer is a minimum of seven, which is somewhat reasonable when you think about the phase linearity constraint.

Thus, noise and distortion are not critical. And this is an argument for the advantages of phase linearity. In any case, if the worst element of a hi-fi system are the transducers, distortion won't affect the sound like it would with a nonlinear phase crossover because of the absence of prior signal distortion.

When designing this type of project try to use as few op-amps as possible and use optimal overall filtering to avoid transient artifacts.

Author's Note: The "right filter" that I have described in this article series is an analog approximation of a digital solution. Janan Zaytoon's hybrid systems philosophy helped inspired my design. I am currently working on another version of phase corrector 1 that would provide a 12 dB better voltage swing. I should have it ready within the next three months. The design will be available at www.muselec.fr.

About the Author

Vincent Thiernesse was born in France in 1973. He is a teacher in applied physics. He has been an electronics hobbyist since the age of 13. Vincent has written seven articles for the French magazine *Electronique Pratique* about measuring, audio filtering, and audio amplification. Vincent has a website www.muselec.fr, where you can find some of his audio creations and music compositions.

Troubleshooting Hollow-State Equipment

There is a close relationship among interests in hollow-state design, current hollow-state technology, and troubleshooting of classic hollow-state equipment. In the old days, many electronic hobbyists got their start working in radio or TV repair shops. Today, many electronic devices are designed not to be field-repairable.

By
Richard Honeycutt
(United States)

In some cases, field repairs would be possible, but replacement parts are not available. For classic hollow-state equipment, there was a period during which sourcing repair parts became challenging, but several retailers have stepped in to fill the gap, and parts availability is now pretty good.

Determining the Problem

The first step in troubleshooting any electronic device is deduction. Determine the problem.

Pilot lights and tube filaments don't light up.

This can be caused by a blown fuse or open circuit breaker. Fuses don't always look blown when they are. The filament can appear to be good, but have no electrical continuity. Or in ceramic- or cardboard-encased fuses, you can't even see the filament. Check fuses with an ohmmeter: they should measure an ohm or less. A simple continuity check is not enough. A fuse that is operated close to maximum current, with occasional spikes, for a long time can suffer from metal evaporating from the filament, resulting in increased resistance. A high-resistance fuse will not properly operate. Several decades ago, I wasted a lot of time troubleshooting a solid-state integrated audio amplifier that turned out to have a bad fuse. The power-supply voltage measured normal when the unit was tested with no signal, but dropped when a few watts of output were drawn. Turned out that the fuse filament measured over a kilohm!

Other possibilities for "dead in the water" equipment include defective power switches and defective power cords. (Of course, the power receptacle into which the equipment is plugged must be "live"!)

Everything lights up, but there is no output.

This can result from a power-supply problem such as an open rectifier or a dead tube somewhere in the lineup. An open output transformer is another possibility.

Less obvious, but still common, are faulty potentiometers (volume controls, especially) or switches. Both can suffer from contamination of contact surfaces resulting in breaks in the signal path. Operating the controls and switches a few times may temporarily restore operation, but even if this is the case, the control or switch should be cleaned using a spray contact cleaner. **Photo 1** shows the nozzle of a can of contact cleaner being applied to the small opening above the terminals of a potentiometer. Avoid the mistake of trying to clean controls and switches by spraying cleaner into the bearing or bushing that surrounds the shaft. The cleaner cannot reach the contact surfaces in this way. Some potentiometers have sealed cases and are difficult to clean. Although the potentiometer can be removed from the equipment and the back cover removed, usually the contacts inside do not need cleaning; the sealed case has prevented contamination. Such potentiometers, if noisy or intermittent, should be replaced. In any case, when contact cleaner is used on any control, the control should be operated vigorously from one extreme of motion to the other to add scrubbing action to the cleaner's chemical effect.

Don't overlook input and output connectors. RCA connectors have a bad habit of developing funky connections with age. Other connectors, while more reliable in general, can also suffer over the years.

Photo 1: Spray cleaner into the opening of a potentiometer to relieve noisy operation.

There are pops, thumps, hums, buzzes, or other noises in the output.

Often pops and scratchy noises are symptoms

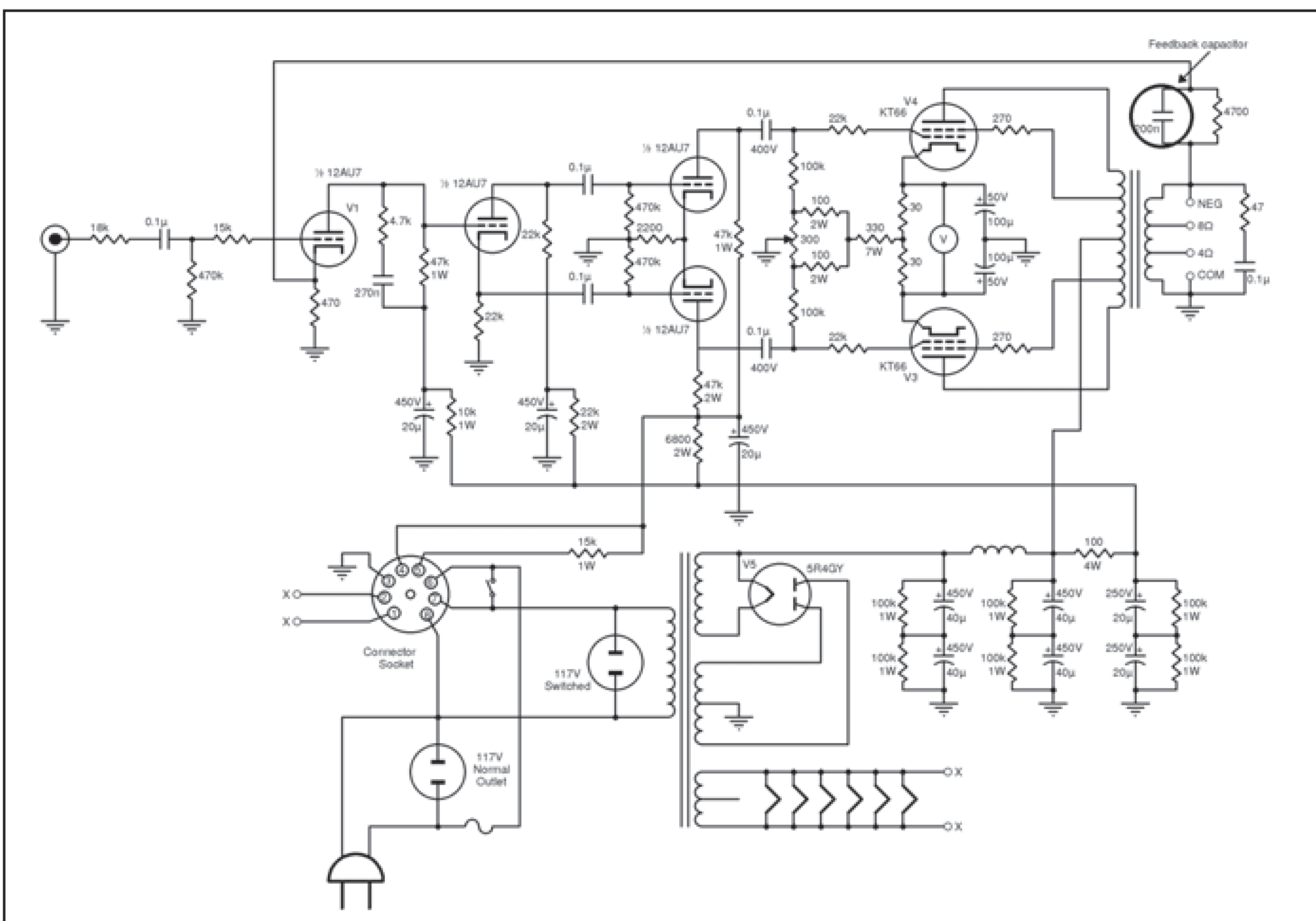


Figure 1: The overall feedback capacitor of this Williamson circuit is circled and identified at the top right of the schematic.

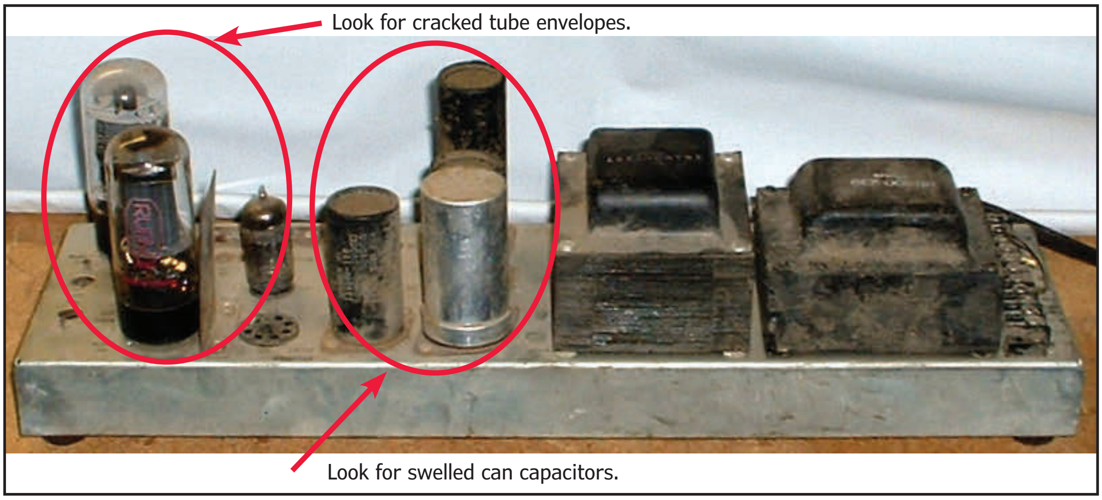


Photo 2: Externally examine the equipment.

of poor connections in controls, tube sockets, or I/O connectors, but they may also indicate gassy power or rectifier tubes. (Gassy tubes are much less common now than they were in the 1950s and before, but classic equipment may still contain old tubes, which may have developed gas leaks.)

With the unit powered on, if you wiggle a tube in its socket and you hear popping sounds from the speakers, the tube socket may need cleaning. This can be done by pulling the tube, spraying the socket contacts with contact cleaner, and repeatedly plugging/unplugging the tube. If you power down before doing this, you won't have to listen to the noise it will generate.

Thumps can be created by microphonic preamp tubes. Power tubes can become microphonic, but usually only as a result of previous overheating. You can test for a microphonic tube by using one hand to hold it firmly in its socket while thumping it with the other. If a thumping sound is generated in the speaker, the tube is microphonic and should be replaced. (First-stage preamp tubes operated at high gain may be slightly microphonic but still operate satisfactorily.) Microphonic power tubes may spark internally when thumped, and in this case, should definitely be replaced.

Hums (pure low-frequency tones), often originate in the power supply, and can be symptoms of filter capacitors that have opened or developed a high Equivalent Series Resistance (ESR). In the 1930s and earlier, wet electrolytics were used for filter capacitors. These consisted of an aluminum can that acted as one plate (usually the negative plate, which was grounded), and a screen wire folded into an extended zigzag shape that served as the other plate. An electrolyte (water solution of mineral salts) filled the can and was kept in place by a rubber seal at the base—at least until the rubber dried out.

When a voltage was first applied to the capacitor, electrolytic action caused the positive plate to develop a thin layer of aluminum oxide that acted as the capacitor's dielectric. The thinness of this oxide

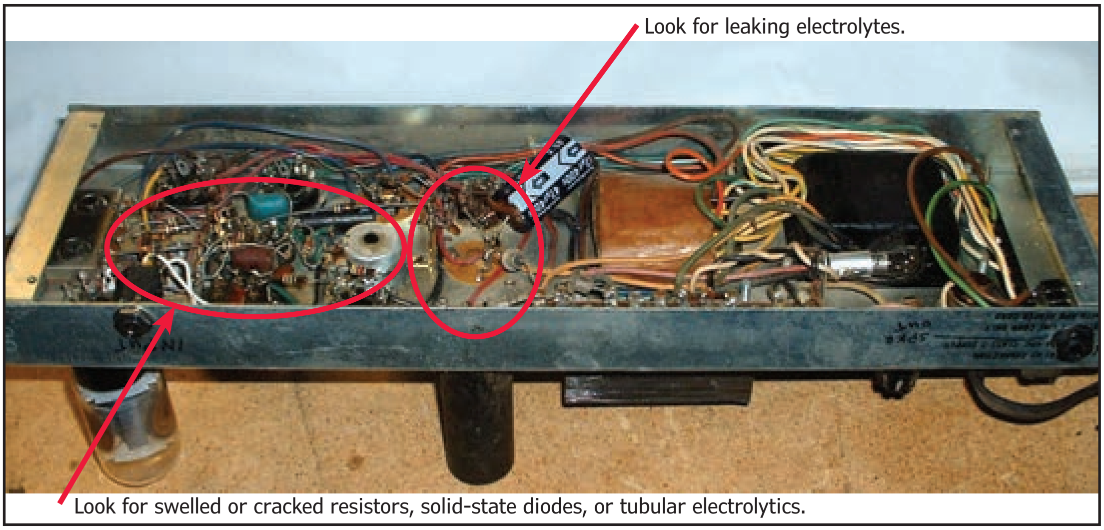


Photo 3: Then, internally examine the equipment.

layer causes the capacitor to have a high capacitance per cubic centimeter. This process is called "forming" the capacitor. After an aluminum electrolytic capacitor has been unused for a longtime, the oxide layer can break down. Then when the equipment is first powered on again, the capacitor must re-form, which can take a while.

During this time, the equipment will have a low power-supply voltage, and will not properly operate. Sometimes the electrolyte leaks out or evaporates over a period of years, and the 10 µF filter capacitor becomes a 0.01 µF unit. The lack of filtering causes hum. Older repair manuals suggested replacing the electrolyte with salt water to restore operation, but given the unknown concentration of salt and the difficulty in re-sealing the can, it is better to replace dried-up "wet" electrolytics.

The other cause of hum is output tubes drawing excessive current, usually as a result of a bias problem [insufficient negative control-grid bias (in fixed-biased circuits) or a shorted cathode-bypass capacitor (in cathode-biased circuits)]. In a fixed-bias circuit, bias issues can be caused by leaky coupling capacitors at the grid of the output tubes, leaky or shorted bias-supply filter capacitors, or open bias diodes.

Buzzes sound like hums played through a fuzz box. They are made up of the power-line sine wave with harmonics that make them sound more like the word "buzz" than the word "hum". In equipment that previously operated correctly, buzzes are often caused by open

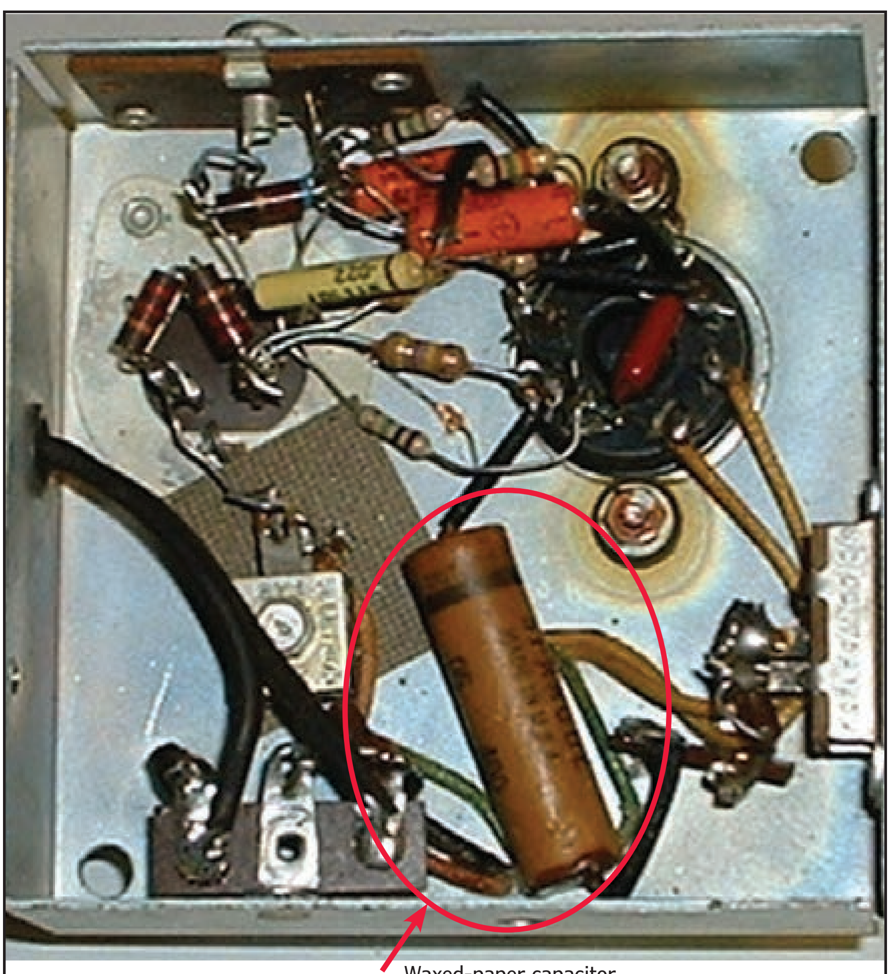


Photo 4: This waxed-paper capacitor has not been overheated.

circuits in a high-impedance portion of the signal path. A preamp tube with an internal filament-to-cathode leak is another possibility.

The output contains squeals or strange-sounding distortion.

Either of these symptoms can result from parasitic oscillation. Several causes are possible. Failure of a ground connection is one, although that more commonly results in low-frequency oscillation called "motorboating." An open feedback capacitor is another suspect. Often these are ceramic capacitors that are connected between the plate and control grid of an amplifier stage, or from the output back to the input, as in the Williamson circuit (see **Figure 1**). The third candidate for causing parasitic oscillation is a missing tube shield, or a telescoping tube shield that is not fully extended. Back in the 1970s, a respected guitar amplifier contained a pentode stage in which a telescoping tube shield was used, and if it became-"un-telescoped," the amplifier would oscillate at a high frequency, causing squeals and/or distortion.

Photo 5: This old tube tester can save a lot of time.

Visual Inspection

After applying "elementary deduction" to localize the problem to one section of the circuit, the next step is visual inspection. With the power off (see **Photo 2**), cracked tube envelopes or cap connectors (used on tubes having top-mounted grid or plate connections) can be seen from the top side of the chassis, as can swelled can-type electrolytic capacitors. With the power on, tubes having burned-out filaments can be identified by the lack of a warm red-orange glow. Tubes whose envelopes allowed atmospheric gas to seep in will have a blue glow between the internal elements. (This should be distinguished from the innocuous blue fluorescent glow at the glass envelope sometimes seen in beam power tubes, caused by electrons hitting the glass.) Gassy tubes will cause popping noises in the output and should be replaced. Rectifier and power amplifier tubes whose plates glow a dim red may not be defective, but are probably drawing too much plate current, signifying a problem elsewhere in the circuit. For a power amplifier tube, it's usually a bias problem.

Underneath the chassis (see **Photo 3**), swelled or cracked components likely need replacing. These can include carbon-composition resistors or tubular electrolytic capacitors. Can-type electrolytics have small rubber pressure-release seals in the base that can rupture if the capacitor overheats, as can result from an overvoltage having been applied. In this case, the ruptured seal can indicate a defective capacitor.

A common event that has caused the replacement of many good capacitors by inexperienced technicians is the tendency of wax-paper capacitors to slowly develop wax drips. These capacitors predated today's polyester and polystyrene capacitors. They used waxed paper as the dielectric and were sealed with wax. After exposure to the heat in the bottom of the chassis, these capacitors often became physically deformed in such a way as to look suspicious.

Photo 4 shows a normal wax-paper capacitor. One that had been overheated would show bulging or dripping wax, but might still operate properly. In fact, wax-paper capacitors were relatively long-lived, for three reasons: (1) there was no real pressure tending to force the wax from between the plates; (2) the wax itself only sealed the paper, which was the actual dielectric; and (3) the wax paper dielectric was self-healing. If a voltage surge caused breakdown of the dielectric by sparking through the paper, the resulting heat would liquefy the wax, which would then flow in to repair the rupture.

Actually, ceramic capacitors failed more commonly than wax-paper ones. The failure mechanism is moisture seepage and condensation through the holes where the leads pass through the ceramic. The plates and the lead-to-plate solder joint can then corrode.

Test the Tubes

If troubleshooting by symptom analysis and visual inspection does not result in a cure, the next step is to test the tubes. Unlike the old days when tubes often burned out or became gassy or developed weakness in the internal structures, causing internal shorts and/or leakage, failure of modern tubes is statistically no more likely to be the cause of a problem than is failure of any other component. This is true not only because of the increased reliability of modern tubes, but also because of the decrease in the life span of passive components due to heat exposure. But tubes do fail, and testing them is easy.

If you have a tube tester (see **Photo 5**), just set the switches, plug in the tube; wait for it to warm up, and read the meter. Tube filament failures are almost a thing of the past. Fortunately, the old "drugstore" tube testers that only tested the filament have also gone the way of the horse and buggy. Any tube tester that you are likely to have is probably a "dynamic conductance" or "mutual conductance" tester that detects shorts and leakage, and measures the ability of rectifier tubes to pass current, or measures the transconductance of amplifier tubes.

Modern preamp tubes' usual failure mechanisms are weakened supporting structures, causing microphonics, noise, and perhaps reduced gain. Power tube failure mechanisms are mainly confined to reduced power output due to long use slowly deteriorating the filament's emissivity, and internal damage caused by problems in other parts of the circuit.

If you don't have a tube tester, it's only marginally more trouble to test the tube by substituting known good ones, one at a time. The downside, of course, is the need to have a complete set of replacement tubes for the device you're troubleshooting. Fortunately, few of us have to repair old 23-tube TV sets these days!

If these steps do not lead you to a solution, it's time to resort to the multimeter and oscilloscope.

Industry Calendar

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

August 14–16, 2015 — **California Audio Show**
Westin SFO, San Francisco, CA, US
www.caaudioshow.com

The 2015 California Audio Show (CAS6) is now in its sixth year in the beautiful San Francisco Bay Area. The event will feature the most recent consumer audio offerings presented by more than 120 brands in 50-plus rooms at the Westin SFO hotel. The event will include the HeadMasters, an entire section dedicated to headphones for high-end audio and related products, and the VivaVinyl! section, with more than 10,000 records from CD to vinyl, and the latest high-resolution audio (HRA) offerings.

August 23–27, 2015 — **SET Expo 2015**
Convention: August 23–27/Exhibition: August 25–27
Expo Norte, São Paulo, Brazil
www.setexpo.com.br

The Brazilian Society of Television Engineering (SET) is one of the most prestigious associations in Brazil and Latin America with direct connections to the National Association of Broadcasters (NAB) and the International Broadcasting Convention (IBC) industry associations and their respective event counterparts. Since 2014, the SET, in cooperation with the NAB, started to directly promote its annual convention and exhibition in a different, modern, and more spacious facility. Even though this is a broadcast-oriented event, SET Expo is undoubtedly the largest and most dynamic business opportunity for the audio and video industries in the Latin American region. The SET Expo benefits from a strong direct presence of manufacturers, with more than 21,000 m^2 for exhibits and over 300 confirmed exhibitors.

September 4–9, 2015 — **IFA Berlin**
Berlin Expo Center, Berlin, Germany
www.ifa-berlin.com

In terms of business impact, the International Funkausstellung (or The International Consumer Electronics Show) in Berlin is probably the most important annual event for the consumer electronics industry, introducing new products in time for the sales season. If the Consumer Electronics Show (CES) in Las Vegas, NV, is the place where future trends and innovation are highlighted, the IFA Berlin show is where many of those products and brands make it or break it. IFA is also the most important event for European manufacturers and many worldwide brands to gain some attention for many deserving products. Audio at IFA is mainly focused on personal/mobile devices and the more accessible and lifestyle oriented products. A wide range of products, solutions, and presentations make the visit to IFA 2015 worthwhile for any company and professional focused on consumer trends and markets.

AMP IT UP

Add high-quality wireless connectivity to your speakers

Dayton Audio MCWA60 Full-range Class D Plate Amplifier with Wi-Fi and Bluetooth 4.0 (with aptX®)

- Create or retrofit self-powered multimedia and monitor speakers
- Wi-Fi and Bluetooth 4.0 connectivity plus optical, RCA, and 3.5 mm inputs
- Fully featured wireless remote for simple, convenient operation
- 2 x 30 watts of Class D power per channel and only 4.5" wide x 6.25" high


SPA500 and SPA1000 Subwoofer Plate Amplifiers


SPA250 Watt Subwoofer Plate Amplifier


SA Series Compact Subwoofer Plate Amplifiers

Visit **parts-express.com/axm** to connect with all of Dayton Audio's strong, versatile plate amplifiers

DAYTONAUDIO

Distributed By:

PARTS EXPRESS
YOUR ELECTRONICS CONNECTION

Tel:800-338-0531
725 Pleasant Valley Dr.
Springboro, OH 45066