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Why Are We Still Debating Speaker Edge Diffraction?

The question in the title is one of many I could use to illustrate topics of debate between subjective preferences, measured facts, and science in audio and how they contribute to the way the audio industry functions. Speaker design is particularly prone to such debates, sometimes even in well-studied areas, such as the diffraction effect.

As Vance Dickason wrote in his *The Loudspeaker Design Cookbook*, "I have frequently commented when asked about the importance of diffraction that the diffraction caused by cabinet edges and baffle protrusions is probably at least as hearable as the diffraction caused by the vase your wife or girlfriend put on top of your speaker, which is to say, not at all." While this may not be far from true, the benefit from damping a front baffle is still a very real and important tool for increasing the quality of the subjective listening experience…"

"Subjective" is the keyword in Dickason's description, while for those who design speakers that are meant to be professional tools—for example, reference studio monitors from Finish brand Genelec—the problem cannot be ignored: "Diffraction is caused by discontinuities on the front baffle and sharp cabinet edges act as secondary sound sources."

As Genelec describes when documenting its Minimum Diffraction Enclosure (MDE) approach, "eliminating edge diffractions is needed in order to improve the flatness of the frequency response and the power response," as well as providing "precise imaging and outstanding sound quality on- and off-axis which leads to clarity and definition of the inner details of the music."

So why do we still have so many "squared" boxes in the market, when the benefit of a well-designed speaker cabinet with round edges is well documented?

As John Atkinson, the editor or *Stereophile* magazine, stated in his paper "Loudspeakers: What Measurements Can Tell Us—and What They Can't Tell Us" (presented at the 103rd Audio Engineering Society Convention, 1997), "while measurements can tell you how a loudspeaker sounds, they can't tell you how good it is."

There's also an excellent *audioXpress* article from James Moriyasu (February 2003, available online, www.audioxpress.com), which offers a study examining speaker cabinet edge diffraction and how it can be reduced. In that article, Moriyasu measures a driver's sound pressure level (SPL) and frequency response in different cabinet shapes, taking into account previous experiments and conclusions published in books from Joseph D'Appolito and Dickason. He concluded that a speaker's image quality is compromised by diffraction because it turns the entire perimeter of the cabinet's front edge into a secondary sound source. Moriyasu states, "This might be another reason why mini-monitors or narrow-faced loudspeakers can often image better than larger systems."

Harry Olson's article titled "Direct Radiator Loudspeaker Enclosures" (*Journal of the Audio Engineering Society*, 1951) illustrated the effect of enclosure shape on cabinet diffraction, measuring 12 speaker shapes. He also established that an enclosure in the shape of a sphere gives the least amount of "ripple" in the SPL response.

As Dickason subsequently describes in *The Loudspeaker Design Cookbook*, "while a sphere may be the best performer in terms of minimal SPL variation, egg-shaped and cylinder-shaped enclosures are also quite good in this respect," adding "It remains true that some of the best reviewed and successfully marketed loudspeakers used simple rectangular shapes."

More recently, in their 2013 Audio Engineering Society (AES) paper "Quantifying Diffraction in Time Domain with Finite Element Method," Juha Holm and Aki Makivirta (Genelec Oy, Finland) write: "Results agree with theory of diffraction. Inversion of polarity in the diffracted wave front is demonstrated. Rounded corners of the enclosure can reduce the amplitude of the diffraction. Using a waveguide can increase the source directivity and further reduce diffraction. Rounded corners and a waveguide can alter the diffracted spectrum."

Their work confirms earlier findings on diffraction by John Vanderkooy and by Soren Rasmussen and Karsten Bo Rasmussen, (*Journal of the Audio Engineering Society*, 1994). The importance of time domain was also confirmed by J. R. Wright (*Journal of the Audio Engineering Society*, 1997), who states in his article "Fundamentals of Diffraction" that "time domain is much more instructive than the frequency domain in the study of diffraction."

It seems to me the audio industry should agree more when it benefits from being science-focused.

João Martins
Editor-in-Chief

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COLUMNISTS

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

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

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

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

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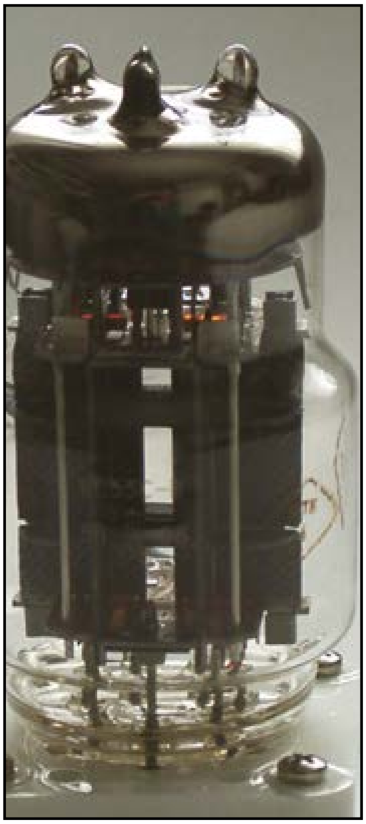

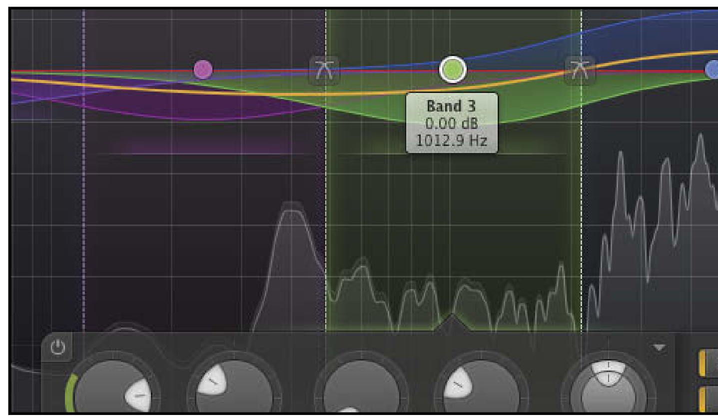

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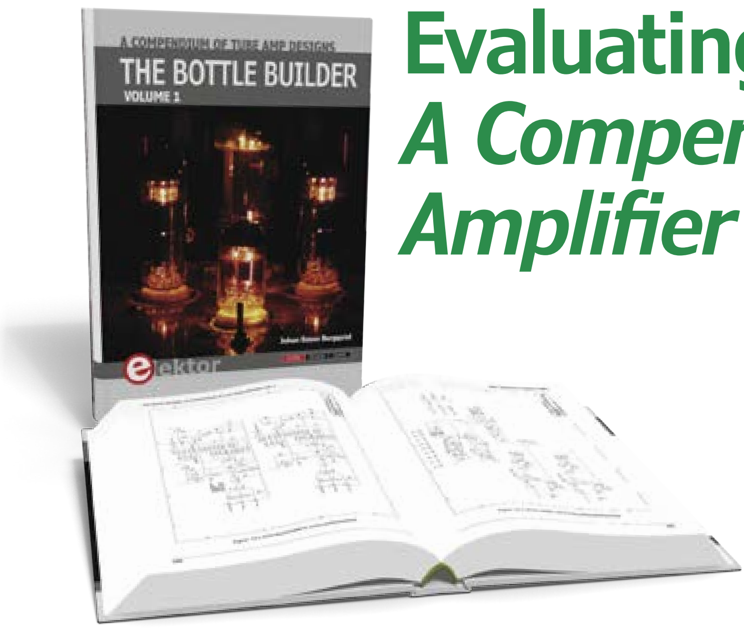

Evaluating *The Bottle Builder: A Compendium of Tube Amplifier Designs*

This review of *The Bottle Builder, Volume 1: A Compendium of Tube Amplifier Designs* discusses Johan Basse Bergqvist's hands-on collection of his amplifier designs. A book illustrated with more than 400 figures and photographs to help readers understand audio—no engineering degree necessary.

Review By
Peter Delos
(United States)

This month we explore a great tube amplifier design reference book by a unique individual whose life experiences in audio electronics are written down in a way that will help you get to a better project faster. The recent release by Johan Basse Bergqvist, *The Bottle Builder, Volume 1*, provides an incredible set of reference designs for anyone at any level who wants to build audio tube amplifiers. The approach is written as a project book, not a theory book. There is a time for theory. Then, there is a time to put the theory to use. *The Bottle Builder* is about getting down to work and realizing practical results that sound great.

The author, Johan Basse Bergqvist, is an engineer, a musician, and an audiophile with a knack for building projects that produce the desired results. The combination of these skills leads to a uniquely valuable perspective on audio design that is routinely reflected in the book and passed on to the readers.

Several design projects are provided, 40 in total. The designs are explained, and the unique features or methods he uses are described in further detail. Each design includes detailed schematics and a complete parts list. Many of the projects also include layout documentation in the form of CAD photos of the PCB layouts. The range of projects is very diverse and includes something that will appeal to everyone. Stereo amplifiers, guitar and bass amplifiers, preamplifiers for phono, and microphones are all covered. Several variants for each type are included, and the power amplifier designs range from a few watts to several hundred watts, which meet almost any power level you might tackle.

The compilation of design documentation on each project is impressive. The book's subtitle, *A Compendium of Tube Amp Designs,* is completely appropriate and an accurate description of the book. Rarely is such a detailed set of design documentation provided in a published format.

This level of detail is usually only available at a working level inside a company when developing a product for manufacturing. As an example, a thorough parts list is detailed on every project, including part numbers, manufacturers, and physical dimensions. This provides not just a list for that project, but a valuable starting point for parts to consider or part families to consider in your own designs. Similarly, clear schematics are provided on every project and documented in a way you can use as-is, or copy elements of a design for your own project.

Quality is emphasized throughout the book,

The Bottle Builder, Volume 1: A Compendium of Tube Amp Designs
By Johan Basse Bergqvist
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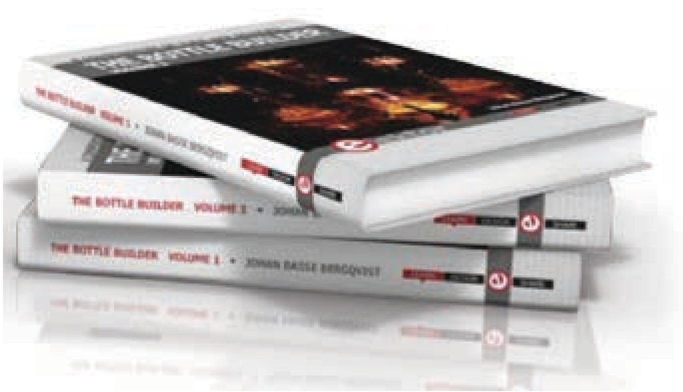
Format: Hard cover
Pages: 440
ISBN: 978-1-907920-38-7
Elektor, www.elektor.com

particularly in the parts selection. As audio builders, we notice and appreciate quality. Oddly enough, I also noticed the quality of the book is very good. There is a hardy hardcover and well-bound pages, with built-in bookmark strings to save your place—well done.

The book provides value to everyone at any level. If you are new to building audio projects, you can build exactly what is documented with confidence that you will have working results. For those beginning to branch out on their own, the great tips and known designs provide invaluable reference points. For experienced builders, the compendium of design information in one spot will save time, lead to better results, and trigger new ideas you can bring to the next level.



The book is worth having on your bookshelf. It will be relevant to tube amplifier audio design for a long time. I predict that years from now, it will be considered a somewhat legendary reference source, providing a unique contribution to the tube amp audio world.

About the Reviewer

Peter Delos is a Radio Frequency (RF) Circuit Design Engineer at a major electronics company in the greater Philadelphia, PA, area. He is also a guitar player. Peter began modifying and designing guitar amplifiers for gigs and recordings when he couldn't buy what he wanted. He does guitar amplifier repairs, modifications, and custom designs for local musicians and friends.

Sound Cards for Data Acquisition in Audio Measurements (Part 2)

Examining the Necessary Requirements

In this article series, we are examining ways to create a low-cost system (emphasis on "low-cost") for lab-grade audio electronics measurements. After reading last month's introduction, I hope I convinced those of you who haven't started using computer-based measurement systems to give it a try. It will greatly extend your capabilities in design, construction, evaluation, and troubleshooting.

By
Stuart Yaniger
(United States)

A very usable system can be implemented for less than $200. You'll need three things to get the measurement capabilities that amateurs could only dream of last century—a sound card (hardware), measurement software, and a means of interfacing the device under test (DUT) to the sound card. This month, we'll start by examining sound card requirements.

Since most sound cards are intended for gamers, home theater, and musicians, we need to decide what features to look for in a sound card targeted for audio measurement.

Inputs

Musicians like lots of inputs with lots of volume controls. But, you don't generally need them, and it is one more way to get levels wrong. A bare-bones two-input sound card is ideal. The mike preamps built into some of these units could potentially be useful for low-level measurement, but in the few I've tried, the circuitry was mystery meat and greatly limited the distortion and noise performance. You want line-level inputs (i.e., 10-to-20-kΩ input impedance and 1 to 4 VRMS, full scale).

Balanced inputs and outputs are a major plus for measurement. They give you the most flexibility in the sorts of things you can easily measure and provide more options for grounding to achieve the lowest measurement noise. Balanced inputs will reject common-mode noise, and can also be run single ended, if appropriate for a measurement. The most commonly used input and output connectors are XLRs (pretty universal in pro recording) and 0.25" phone jacks (which are popular in semi-pro and amateur music production). Some units even have combo jacks that can accept both.

RCA phono jacks for analog I/O are a sure sign that the inputs and outputs are single ended. Likewise mini-phone plugs. You can still work with those, but it's unnecessarily limiting and noise can easily contaminate the measurements.

Sample Rates

This will determine the high-frequency limits of your measurements. The Nyquist theorem shows the maximum frequency that can be accurately sampled, known as the "Nyquist limit," it is half of the sampling frequency. (Historically, this work is attributed to both Harry Nyquist and Claude Shannon, but for whatever reason, Nyquist's name seems to dominate. My non-mention of Shannon is for convenience and consistency, and is not meant to slight this brilliant mathematician, who was also a singular pioneer of modern information theory.)

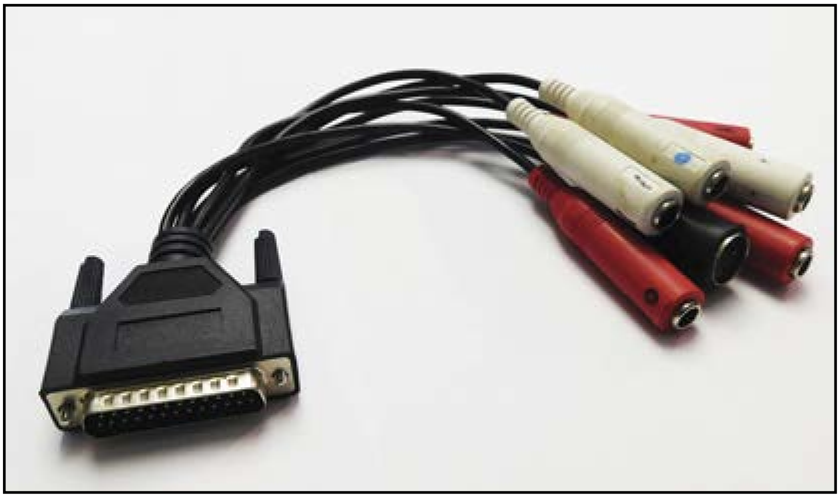


Photo 1: A breakout cable for PCI sound card, though not essential, is a nice feature giving you flexibility in connector choice.

Generally, there are filters in A/D converters to limit the input frequency to below the Nyquist limit, but these filters are not perfect brick walls, passing all data below the Nyquist limit and blocking all data above the Nyquist limit. So as a practical matter, the highest measurable frequency will be somewhat lower than the Nyquist limit. As a rule of thumb, I assume that I can't trust anything above 90% of the Nyquist limit (e.g., for a 192-kHz sample rate, the Nyquist limit is 96 kHz, so I can use the measurement up to about 87 to 88 kHz).

All things being equal, higher sample rates are better. The bare minimum is 96 kHz for any serious measurement—192 kHz is sufficient for testing most equipment within the audio band. There seems to be very little push for sample rates higher than 192 kHz in the music recording market, so we're probably limited to that for the foreseeable future. And even that sample rate is the exception, since 96 kHz seems to have become a de facto "hi-res" standard, with 192-kHz sound cards quickly vanishing from dealer stock.

The two sound cards I use in my day-to-day work are an internal (PCI) card, the M-Audio Delta Audiophile 192, with two-channel balanced I/O and 24-bit/192-kHz capability; and an external card, the Echo Audiofire 2, with two-channel balanced I/O and 24-bit/96-kHz capability, communicating to my laptop via Firewire (IEEE-1394 interface standard). Both use 0.25" stereo phone plugs for each channel.

In terms of input and output levels, 0 dBFS (full-scale level) = 4 dBU (1.25 VRMS) for the Echo, and 0 dBFS = 14 dBu (3.9 VRMS) for the M-Audio. If you can afford the price, the Lynx L22 has amazing performance, with a 200-kHz sample rate, distortion, and noise in the single-digit parts-per-million range, and is available in the low-profile PCIe format.

For an external card, the Focusrite Scarlett 2i2 is one I've used a few times with good results. As with most other USB interfaces, its maximum sample rate is 96 kHz, but the distortion and the noise performance of the unit I tried was excellent.

Current availability of these specific models is spotty, but there are several similar units filling their niches, and I'll be reviewing some good ones over the next few months.

One nice-to-have feature is I/O via a breakout cable rather than the jacks built into the unit (see **Photo 1**). This reduces issues of strain relief with long cables and as a bonus, enables you to change the type of connector by clipping off the old jacks and soldering on new ones. That's somewhat tougher to do with jacks directly soldered to a PCB. It can be helpful to use a marker of a contrasting color to label the jacks, especially for those of us with old eyes or the inevitable poor lighting behind the computer.

Limitations

With all of this wonderful measurement power (literally) at our fingertips, there is a great temptation to think that there's no more need for "conventional" test equipment. Sadly, this is not the case because there are limitations with sound-card-based measurements.

First and foremost, there is the bandwidth. As previously mentioned, the high-frequency limit of a computerized test system is determined by the Nyquist frequency. So if a circuit oscillates at 20 MHz (which is not an uncommon occurrence), it will be invisible to you. Likewise, if you want to measure the residual AC on the output of a switching supply or a Class-D amp, the harmonic content of a jagged

Resources

"Gibbs Phenomenon," *Wikipedia*, http://en.wikipedia.org/wiki/Gibbs_phenomenon

"Shannon-Nyquist Sampling Theorem," *Wikipedia*, http://en.wikipedia.org/wiki/Nyquist–Shannon_sampling_theorem

Sources

Audiofire 2
Echo Digital Audio Corp. | www.echoaudio.com

Scarlett 2i2 audio interface
Focusrite, plc | http://us.focusrite.com/usb-audio-interfaces/scarlett-2i2

Lynx L22 PCI sound card
Lynx Studio Technology, Inc. | www.lynxstudio.com

Delta Audiophile 192 audio card
M-Audio | www.m-audio.com

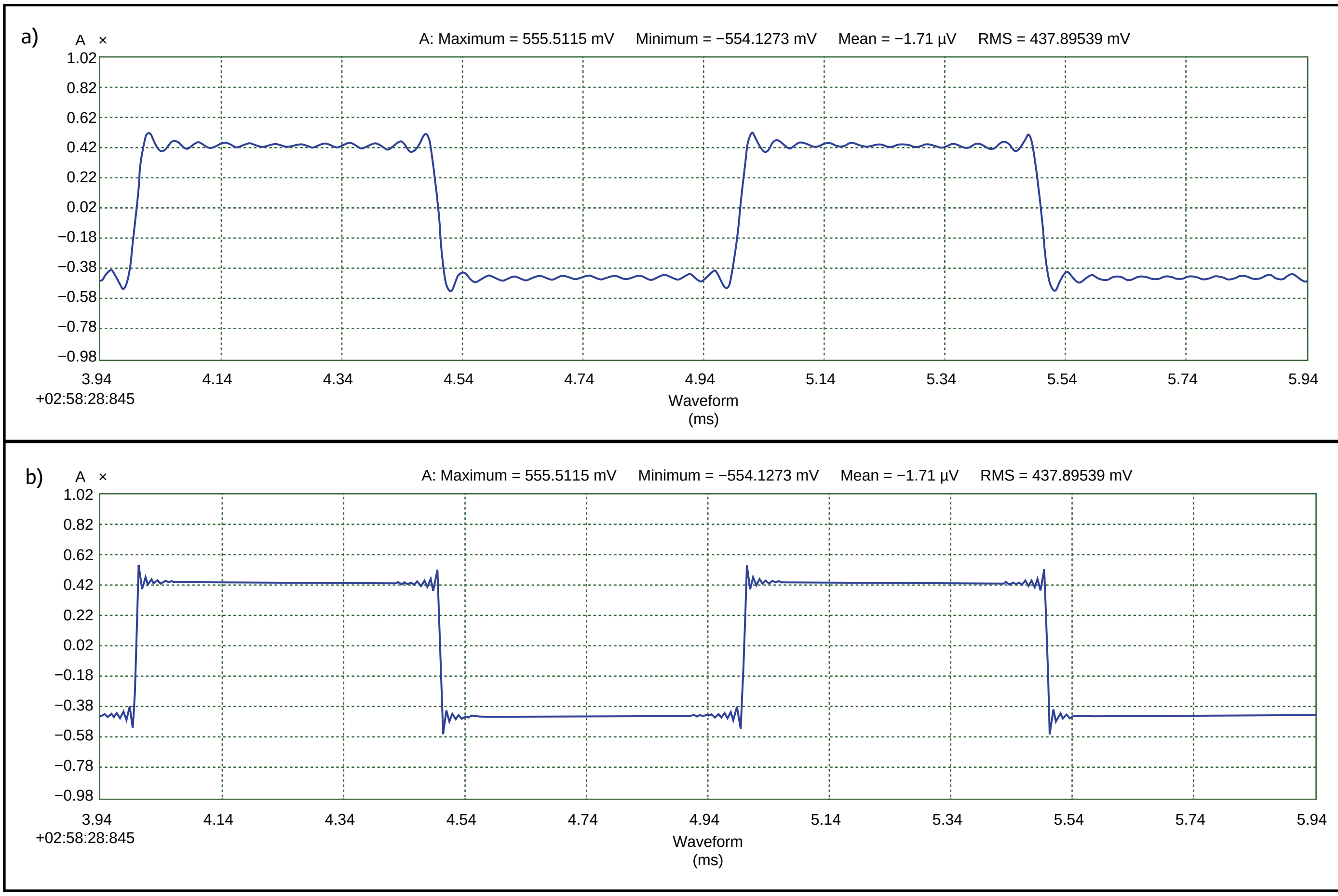


Figure 1a: A 1-kHz square wave generated with a 96-kHz sample rate. b: a 1-kHz square wave generated with a 192-kHz sample rate.

30-kHz waveform, or the quality of an SPDIF signal, you're out of luck. If you want to look at the rise of distortion in an amplifier with increasing frequency (which is common due to falling open loop gain), the harmonics above the second order for a 20-kHz signal will not be detectable with a 96-kHz sound card, nor will harmonics above the fourth order be detectable with a 192-kHz sound card.

The other consequence of limited high-frequency bandwidth is that clean square waves can't be generated at midrange and treble frequencies. Square waves have an infinite number of odd-order harmonics. Truncation past some upper frequency limit results in a waveform with ringing (the infamous Gibbs Phenomenon). The bandwidth-limiting at one-half the sampling frequency necessary to satisfy the Nyquist criterion will slow the rise time and cause ringing (see **Figure 1**). Besides the ringing due to bandwidth limiting, these

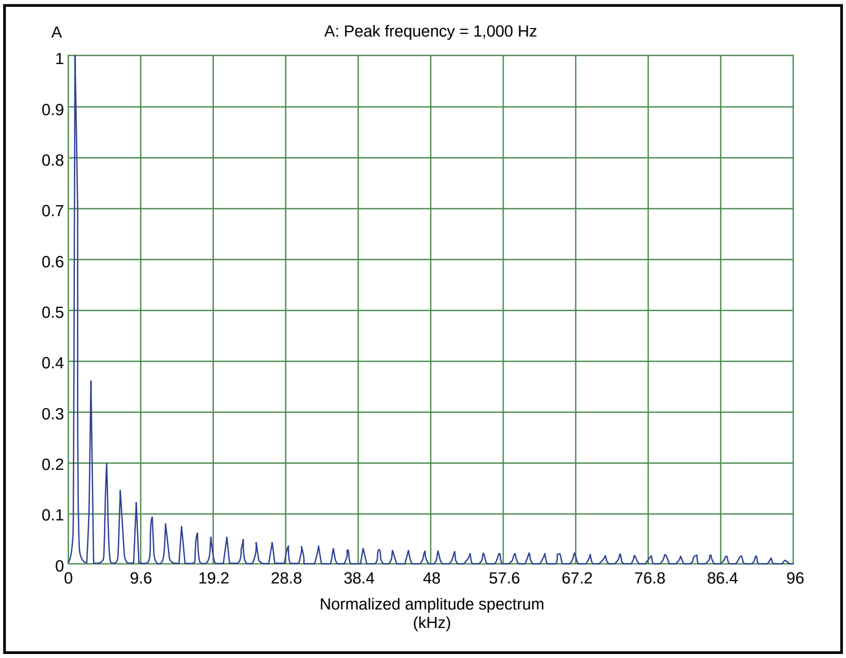


Figure 2: The frequency domain spectrum of a 1-kHz square wave generated with a 192-kHz sample rate. Note that all harmonic components below the Nyquist frequency are present.

diagrams also show the effect of sample rate: higher sample rates have a sharper overshoot but of lower magnitude. The frequency spectrum of a sampled square wave is shown in **Figure 2**. All the harmonics are present up to the Nyquist cutoff. We're just limited by the basic mathematics, which demand infinite bandwidth and that's the unfixable cause of the ringing.

So using square waves to check amplifier stability will best be left to conventional instruments (e.g., square wave generators and oscilloscopes), despite the temptations of the digital oscilloscopes and square wave generators built into the measurement software.

At the other end of the frequency range, the inputs and outputs of sound cards are AC coupled, so DC and very low frequency measurements (e.g., measurement of bias drift over time and dielectric absorption of capacitors) can't be done without building some elaborate chopper-type interface. A good sound card can go well below subwoofer roll-off frequencies and should even be able to measure phono cartridge-tone arm resonance frequencies (7 to 12 Hz, optimally).

Second, there is impedance. A sound card's input impedance is likely to be somewhere around 10 kΩ. That's no problem if you're measuring things such as preamps with low output impedances or power supply outputs, but if you want to measure a high-impedance source (e.g., the signal at the plate of a tube or drain of a FET), the sound card will load it down. However, this issue can be easily remedied.

Third, audio signals range from microvolts to hundreds of volts, whereas sound cards have their "sweet spot" for top performance in the 1-to-2-V range. If, for example, the input signal for a distortion measurement is too low, it will be difficult to pick up the harmonics in the noise. If a signal is too large, measured distortion will be high because of the sound card's input stage overloading. If the signal is large enough, it can even damage the sound card inputs. As with impedance, this is also easily taken care of, as we will see later in this article series.

Fourth, you need to be able to check your work. There are so many moving parts that odd-looking measurements need to be verified. You may find that you'll want to do this as a matter of course. Seeing a lot of high-level high harmonics in a sine wave measurement or a funky frequency response? They may be there, but it could also be that something is clipping or there's a wrong setting or a multitude of other reasons. An analog oscilloscope will provide a useful check. Likewise, unusual noise—is it a grounding issue or a pickup from some radiating device?

Fifth, the distortion spectrum vs. level may be suggestive, but using an oscilloscope to examine the output of the distortion residual from a conventional distortion analyzer will be faster and more certain. Of course, you can multiply your measuring power by running the output of the distortion analyzer into the sound card measurement system, allowing it to be monitored in the time domain (i.e., virtual oscilloscope) or the frequency domain (i.e., spectrum analyzer).

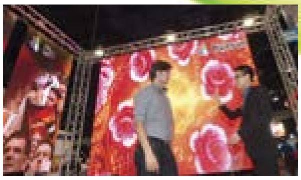
Comparing the distortion waveform with the input waveform can provide clues as to the distortion source by noting the point(s) on the waveform where the distortion occurs. For example, if there's a spike in the distortion waveform that lines up with the input signal's zero crossing, you're probably dealing with crossover distortion.

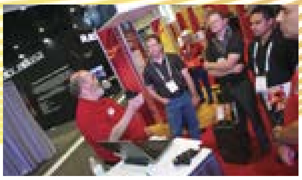
Bottom Line

The virtual Swiss Army knife of a good sound-card-based system will take care of the vast majority of your needs, but every once in a while you have to reach into your metaphorical toolbox and pull out a tape measure or a monkey wrench. The big tools will always have their uses. And the naked sound card will need to be clothed to take advantage of its versatility and performance. Fortunately, that part is easy.

Next month, we'll look at computers and some measurement software options.


WOW




Enhanced Stereo (Part 1)

The Basics

As I mentioned in my "Signal Dynamic and Loudness" article (*audioXpress*, May 2015), it's difficult to know how to group the various approaches to realistic sound reproduction. I wrote an outline, arranged and re-arranged the methods, and finally decided on this grouping: first, all the techniques that use just two loudspeakers and then, those that have three or more. The first group includes Enhanced Stereo, Binaural from Loudspeakers, and others. So that's where I'll start this month's article.

By
Ron Tipton
(United States)

The goal of stereo enhancement is to artificially create sound-stage widening by "processing" the input. This is primarily done in a digital audio workstation (DAW) with a plug-in, where a VST plug-in is the most common in Windows. Some of the plug-ins operate on either a mono or stereo input and some are stereo input only. Although limited widening can be heard starting with a mono file, I have discovered the effect is more realistic if the mono file is first converted to stereo. This can be done with a mono-to-stereo converter such as Musereo, which can also add some user-selected enhancement, so the resulting stereo file isn't just two combined mono files.

A conversion can also be done in a DAW or audio editor by high-pass filtering the mono file and then importing it and the original into an audio editor, such as the free Audacity. Then, select both files and export as stereo. An eight-pole Butterworth high-pass (–48 dB/octave) with a cut-off frequency of 100 to 200 Hz works well in most cases because it alters both the frequency and phase content between the two channels. I have included two sample mono-to-stereo file conversions in the Supplementary Material: lady-stereo.wav and lady-musereo.wav, which was enhanced using the default settings in Options. It shows a significant listening improvement over the original mono file.

Does the processing make the music more realistic? It does make it more interesting to listen to, so for me, the answer is yes. I'll describe a few of the many plug-ins that are available, starting with those that are free. Then I'll introduce you to a hardware widener and a few plug-ins that are not free but are very good at widening the stereo sound stage. There is another type of enhancement plug-in that adds reverberation using the impulse response of "real" rooms so I will include some of these because it's an effective means of increasing the recording's "liveliness." In general, these are not psychoacoustical, that is, the enhancement can be saved as a file and replayed.

BetabugsAudio's WideBug

This free plug-in is very simple with only one knob that smoothly controls the stereo width between 1 (with no expansion) and 6, which can be overkill (see **Photo 1**). With no input or output volume controls, you need to closely monitor the levels and reduce the input level with the plug-in host if distortion is detected. Using Sony Audio Studio as the VST host, I tested Widebug and the other plug-ins with two input files: lady-stereo.wav created from the Artie Shaw 1937 mono recording of "That's Why the Lady is a Tramp," and lady-musereo.wav.

For Widebug, I chose a times two and times three expansion and both made for interesting listening. Both versions increased the output amplitude so the input level had to be reduced before expansion. Also, the widening seems more pronounced using a pair of monitor loudspeakers spaced 2′ to 3′ apart than from column loudspeakers spaced 8′ to 10′ apart. A stereo clip, lady-widebug.wav, produced at a times three expansion of lady-stereo.wav, is included in the Supplementary Material.

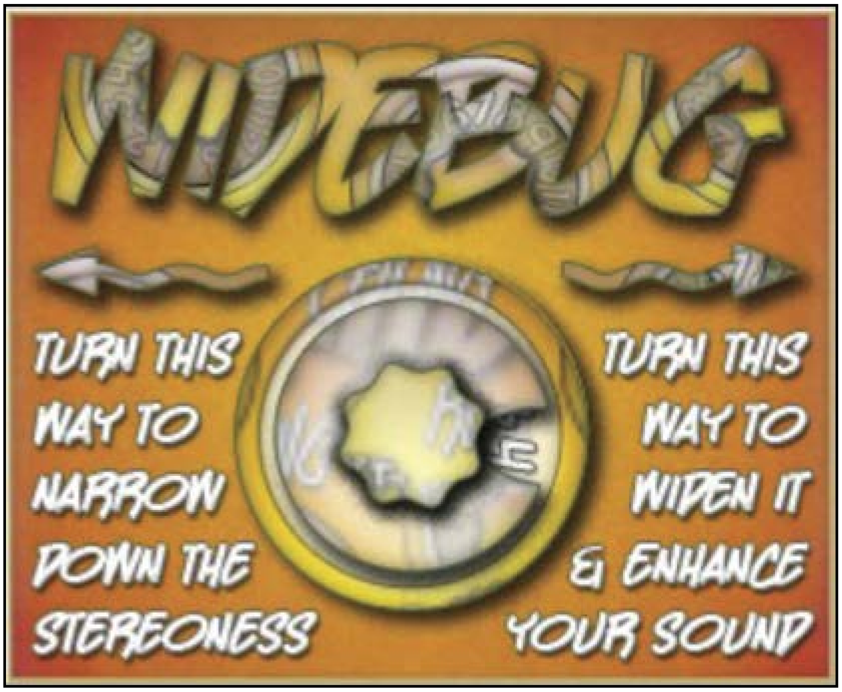


Photo 1: The rather amusing Widebug interface is easy to use with its single control but the input and output levels must be set in the plug-in host.

Photo 2: As with the Widebug, the M-ST input and output levels must be set in the host. Although it accepts a mono input, this does not produce a stereo output file and the expansion is limited to just 150%.

WOK's M-ST

Another free one-knob plug-in accepts either a mono or a stereo input, but it only goes to a 1.5 expansion, labeled as 150% on the dial (see **Photo 2**). But even with this limitation, it sounded good with both of my test files. I did test it with a mono input, lady.wav, but as I mentioned earlier, the expansion sounds more realistic starting with a stereo input. Also, "running" the plug-in with a mono input does not produce a stereo output file—the output is still mono. The Supplementary Material includes an example clip, lady-mst.wav, with a 1.5 expansion of lady-stereo.wav.

Voxengo's Stereo-Touch

Voxengo has many plug-ins but this one is free and it performs well. The lack of an input volume control is inconvenient but it's hard to complain about free software. The control interface has more than one knob, but several presets are included (see **Photo 3**).

For testing variety, I tried the "Wide" preset with the "ShhhPeacefull" track from a Dizzy Gillespic HD Tracks download and was impressed with the expansion. The clip shhh-stereo-touch.wav is also in the Supplementary Material. The "Narrow Space" preset was more subdued but was still very pleasant. The User Manual, included in the Supplementary Material, explains what the controls do, which helps shorten the learning curve. I've included a clip from lady-stereo.wav as lady-stereo-touch.wav using the "Surround Delay" preset. This preset seems less wide than "Wide" and wider than "Stage." I've also included two MP3 clips from Voxengo's website, the preset name is in the file name.

Brainworx's bx_solo

bx_solo is a free VST plug-in for stereo input files from the Plugin Alliance. Although it's free, activation is required but that's easily done online after creating a free account. The user interface screen has only one control knob, Stereo-Width, settable from 0% (mono) to 400%, where 100% is no enhancement. There are also push-buttons for Left, Right, Mid, and Side channel only. For example, pushing the Mid button extracts the Mid (mono) channel from the input stereo.

I also used the "ShhhPeacefull" track for testing.

Photo 3: The Stereo Touch interface looks complex but several presets are included and you can easily make changes to them to observe the effect on the expansion.

Project Files

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

Resources

B. Butterworth, "What You Might Not Know—But Really Need to Know—About Bass," November 2014, http://audiophilereview.com/room-acoustics/.

HDtracks, www.hdtracks.com.

Plugin Alliance, www.plugin-alliance.com.

Sources

A1 Stereo Control expansion plug-in
A1Audio.de | http://a1audio.de/index.php/a1stereocontrol

Audacity cross platform software
Audacity | http://web.audacityteam.org

Widebug stereo enhancer plug-in
BetabugsAudio | www.betabugsaudio.com

bx_solo VST plug-in
Brainworx Music & Media GmbH | www.brainworx-music.de/en/plugins/bx_solo

Musereo Mono to Stereo Converter
Musereo | www.musereo.com

Sound Forge Audio Studio editing software
Sony | http://www.sonycreativesoftware.com/soundforgesoftware

Stereo Touch plug-in
Voxengo | www.voxengo.com

M-ST plug-in
Wok Wave | www.kvraudio.com/product/m-st-by-wok (www.wokwave.com no longer provides download)

Photo 4: Alex Hilton did an excellent job designing this free plug-in. His SafeBass option correctly puts the low frequencies in the soundstage center. There are several presets to get you started.

The track's peak amplitude was about 2.5 dB below clipping, but when I ran bx_solo in Preview mode at 200% width, it increased the amplitude into the clipping range. I had to reduce the track's volume by 3 dB to avoid this problem. The width increase is very noticeable and I've included an output clip, bxsolo.wav, along with a clip from the original track named shhh-44-2.wav in the Supplementary Material.

A1 Stereo Control

This free stereo input only plug-in from Alex Hilton is a marvel for several reasons: it's free and it has a graphical display of the expansion and panning (see **Photo 4**). Perhaps even more importantly, it correctly centers in the soundstage the bass frequencies below 200 to 250 Hz because these frequencies do not contribute to the stereo image. Why this is true, is explained in a short article by Brent Butterworth and a copy is included in the Supplementary Material as About-Bass.pdf.

Alex calls this "SafeBass" and it can be turned on or off and the cut-off frequency varied using the controls on the right side of the graphical display. The plug-in documentation is brief so experiment to find out what the other controls do. The input and output level knobs need no explanation nor does the Mono switch, which converts a stereo input to mono output. There are several presets that can be selected with the arrow buttons at the screen top.

I tested it with the preset named "Stereo 200% + SafeBass" with the frequency control set at 100 Hz. For the lady-musereo.wav input file, I set the Input attenuator to −4 dB and the output attenuator at 2 dB. This clip is named lady-a1.wav in the Supplementary Material.

Other Plug-Ins

I also tested two other free plug-ins—Sjoe.dll from www.terrywest.nl and Stereo-enhancer.dll from www.tonebytes.com—but I could only hear very limited or no expansion from them. Of course, maybe my listening environment or my hearing played a part in the results. Perhaps you will hear some expansion that I didn't. Next month, we will continue our quest for realistic sound with the second part of the enhanced stereo article. 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 Review

Exploring AXPONA 2015

Love for Music and the Best Sound

Photo 1: Vinnie Rossi' LIO is designed as a chassis-with-power for a line of modular, function-specific modules.

Celebrating its third straight year in Chicago, IL, and its second year at The Westin O'Hare, Audio Expo North America (AXPONA) surpassed 2014's turnout with record-breaking numbers for attendance, exhibitors and displays, while the Ear Gear Expo doubled in size. Oliver Masciarotte reports on three jam-packed days of the latest in high-end listening equipment and the hottest in music reproduction technology.

By
Oliver A. Masciarotte
(United States)

It was with a bit of trepidation that I decided to pack up the car, swing by my friend Mike Newman's place to drop him into the copilot seat, and make the almost-manageable road trip to Chicago for our first visit to AXPONA 2015. Both being veterans of Rocky Mountain Audio Fest (RMAF), we were pleasantly surprised to find a much nicer venue with high-quality vendors and exhibits. Lest I get sidetracked waxing poetic about Chicago's yummy Polish food, let's dive right into those deluxe audio demo rooms.

Induced noise in truly high-fidelity gear has been an eternal shortcoming of AC mains power. In the past Vinnie Rossi (http://vinnierossi.com), of Red Wine Audio fame, has resorted to battery power to keep his noise floor "black." With his newest product, the LIO, he's leveraged dual banks of ultracapacitors to act as a power reservoir in place of batteries. To make the product future proof, LIO is designed as a chassis-with-power for a line of modular, function-specific modules (see **Photo 1**). The current modules include an integrated amp, a tube preamplifier, a phono preamplifier, a DXD/DSD2 DAC, resistive or autotransformer volume controls, a Class-A hollow-state gain stage for color, and other functional modules like an input selector and headphone amp. LIO starts at $2,500 and can cost as much as $6,500 when fully loaded.

Headphone Offerings

In the headphone area, ZMF Headphones (www.zmfheadphones.com) was showing its new, closed-back ZMF Blackwood ($699). With comfortable, nicely sealing leather ear pads, a wideband 12 Hz to 27 kHz ±3-dB frequency response and a 50 Ω impedance, the handcrafted Blackwood is a lower distortion reference model that still retains ZMF's unique, user-adjustable low-frequency tuning along with improved transient response and imaging (see **Photo 2**). A semi-open-back model is in the works.

Along with the rest of its line, OPPO Digital (www.oppodigital.com) had its new stylish and lightweight PM-3 foldable cans ($399) on display. At 320 grams, the PM-3 is currently the world's lightest closed-back planar magnetic design (see **Photo 3**).

Another planar contender was an offering from Audeze (www.audeze.com). Its new entry-level EL-8s ($700) possess the same house sound as its more expensive siblings, in a 30 Ω design that uses mostly metal rather than wood to bring down the weight and the cost, while improving robustness. A closed-back EL-8 is also available for monitoring on set and on location. As with other Audeze products, I found the voicing to be noteworthy in its neutrality.

Photo 2: ZMF Headphone's wild man Zach Mehrbach and his new Blackwood closed-back headphones are shown.

Headphone-Related Electronics

Some interesting new headphone-related electronics were also on display. Brannan Mason, co-owner of noted IEM manufacturer Noble Audio (http://nobleaudio.com/en/shop/wireless), had something new at the show that wasn't an earphone. The Noble Bluetooth Solution (BTS) is a novel add-on approach to wireless phone connectivity with an emphasis on retaining the benefits of your favorite IEMs (see **Photo 4**). At just shy of 0.5 oz, the minuscule BTS transceiver ($100) is designed to provide at least 7 hours of continuous music playback or talk time, and recharges in 2 hours. It features the latest Bluetooth 4.0 with aptX and Apple lossless support, as well as an omni microphone with push-button controls for making and taking calls without messing with your cans or your phone.

Wyred 4 Sound (https://wyred4sound.com) was showing a prototype of its upcoming Íntimo DXD/DSD2 DAC/headphone amplifier. With differential Class-A guts, both balanced and unbalanced outputs, and a low/medium/high gain switch, the $1,000 unit rolled off the line in June.

JDS Labs (www.jdslabs.com), better known for its DIY kits, parts, and subassemblies debuted its new O2+ODAC, a 96 kHz/24 bit transportable DAC/

Photo 3: OPPO Digital's swanky new PM-3 closed-back planar headphones are lightweight foldable cans.

Photo 4: Brannan Mason's bitsy Noble Bluetooth Solution (BTS) is a novel add-on approach to wireless phone connectivity.

Photo 5: Woo Audio's spiffy WA8 is an all-tube transportable DAC/headphone amplifier.

Photo 6: Questyle's QP1r will soon to hit the shelves.

Photo 7: Astell&Kern's new AK Jr is an entry-level portable high-resolution audio player.

headphone amplifier that's assembled and ready to rock ($280).

As a follow up to the HERUS+ dongle–style DAC, Resonessence Labs (www.resonessencelabs.com) was showing a prototype of what its been messing with in the lab: The self–powered HERUS Mini ($600). With a 1/8" headphone jack, the Mini is meant to be mated to a smartphone for hi-fi conversion.

One of my favorite, do-no-wrong manufacturers is Ayre Acoustics (www.ayre.com) in Boulder, CO. Ayre was showing a new $1,800 headphone amp/DAC/preamp called The Codex, which will support pulse code modulation (PCM) up to 192 kHz/24 bit, plus DSD2. The company's man-in-the-trenches, Alex Brinkman, was really excited about it for several reasons. "First and foremost, it is the most multi-functional product we have ever been able to bring to market. As a high-performance two-channel brand, most of our products up until this point have been able to do only one thing: function as a DAC, a power amp or a preamp. The Codex breaks that limitation and allows for the same high levels of performance to which people have grown accustomed with our lineup. It can be used in the following configurations:

1 - As a headphone amp/DAC connected to a digital source (Win/Mac, Apple TV, XBOX One, etc.) and a set of high performance headphones.

2 - As a DAC/preamp: The volume control allows the user to connect the analog (unbalanced RCA and balanced XLR) outputs to a power amp in a traditional two-channel rig. This makes for a simple, streamlined system for someone just getting into a larger bookshelf or floor-standing speaker configuration.

3 - As a stand–alone, entry level DAC for a high-end system. This would allow the user to employ a USB/TOSLINK DAC at half the price of our award winning QB-9 DSD."

Woo Audio (www.wooaudio.com) had several prototypes of a new WA8 all-tube transportable DAC/headphone amplifier (see **Photo 5**). Though the elongated VHS-sized casework wasn't final, the design was reminiscent of Dieter Rams' work for Braun. A nice touch was the relatively large window through which the military tubes were visible. Pricing isn't set but it should be around $1,500. Its projected battery life is currently 3 hours of continuous use.

Etymotic (www.etymotic.com) showed its new ER•20XS, the next generation of the industry-leading ETY•Plugs "High-Fidelity" hearing protection. With triple flange tips, the $20 ER•20XS earplugs can also be used with memory foam eartips with essentially no change in response. Etymotic also demonstrated

Photo 8: The Focus XD line is an active wireless-ready range from Dynaudio.

Photo 9: The proof is in the listening—wood blocks vs. IsoAcoustics' new Aperta isolation mounts.

its mk5 earphones ($60), the newest addition to Etymotic's hi-fi Isolator line. The mk5 delivers outstanding house sound with high isolation at an entry-level price.

Though I brought my own, Pono Players were in evidence around the room—being used as sources—and several were equipped with cable sets that supported the product's balanced output. Another source of interest for me was a yet-to-be-released portable from current mode acolytes Questyle (en.questyleaudio.com). Though the price has yet to be determined, the QP1r, shown in see **Photo 6**, should enter the market at around $900.

Market leaders Astell&Kern (www.astellnkern.com) also showed an exciting new portable. The AK Jr features high output, a low 2 Ω output impedance and a slim, 6.9 mm design (see **Photo 7**). It retains the company's entry-level gear's ability to transcode DSD on the fly to PCM, 192k max, and sports a nice big 3" display. At $500, the AK Jr is Astell&Kern's new low-priced, gateway drug.

On to Loudspeakers

Though announced late last year, I had not seen Dynaudio's new Focus XD line (www.dynaudio.com). One of its rooms showcased the top-of-the-line Focus 600 XD, a three-way, four driver active floorstander ($13,500). The unit has built-in DSP driving a 150 W amplifier per driver, for 600 W total per speaker (see

Photo 10: Sanders Sound's MODEL 10d ESL is Sanders Sound Systems' flagship design.

Photo 11: Muraudio's novel Domain Omni PX1 ESL provide 360° of horizontal coverage.

Photo 12: Mark Waldrep's Millennium Falcon OPPO made for excellent sound and picture.

Photo 13: The supertweeter-equipped Soundfield Audio Dipole 1 has a semi-active, open-baffle design.

Photo 8). The Focus 600 XD accepts digital inputs via an RCA, up to 192k, as well as an analog RCA in. Alternatively, the speaker can be connected to the Dynaudio Hub. The Hub has inputs for both analog and digital and wirelessly transmits a 48k signal to the Focus XD via Wi-Fi.

Although it was preaching to the choir, IsoAcoustics (www.isoacoustics.com) was showing its new Aperta Series isolation mounts with a simple and compelling configuration to substantiate the company's claim of improved sound. Set up were two pair of identical Paradigm 15B speakers and Sanus SF floor stands. One pair was equipped with Apertas between the speakers and the Sanus stands, while the other had an equal height of wood blocks as a control (see **Photo 9**). Switching between the two produced the proverbial night and day difference. The wood blocks and the Sanus stand-only combination sounded congested and muffled, while the IsoAcoustics/Sanus combo sounded like, well... like I expect Paradigms to normally sound—clear and snappy.

Having heard Sanders Sound Systems' hybrids at last year's Rocky Mountain Audio Fest, I dropped in on its room and was rewarded with the refined yet powerful sound one would expect from a well-engineered, lower-cost electrostatic loudspeaker (ESL) system. The demo consisted of a $5,500 Magtech Stereo 500 W into 8 Ω amplifier driving the new MODEL 10d—a $15,000 electrostatic hybrid design with a 10" aluminum cone driver mated to a transmission line enclosure for the low end (see **Photo 10**).

Muraudio (muraudio.com) offered another strategy for implementing an ESL, in its active $63,000 Domain Omni PX1 (see **Photo 11**). With its barrel-shaped electrostatic mid/tweeter, and array of three 25 cm acoustically suspended aluminum cone woofers radially spaced at 120°, the speaker produces an apparent point source at any angle horizontally and within ±8° vertically. Powered by $96,000 worth of Simaudio (www.simaudio.com) electronics and just shy of $58,000 of Nordost (www.nordost.com) interconnects and power distribution, the result was impressive. A significant difference between the Muraudio and the Sanders Sound ESLs are that, while the MODEL 10d uses an inexpensive digital crossover requiring digitizing and subsequent re-conversion back to analog, the Domain Omni PX1 employs a fourth-order Linkwitz-Riley analog crossover.

I also found what, to me, is a new line of loudspeakers. AudioKinesis (www.audiokinesis.com) was showing its unique Zephir 46, a $4,900 Z-shaped bipole with two 6" mid/bass drivers on the front flanking a horn-loaded, Radian Audio 1" beryllium compression driver. In the back are two additional 6.5″ mid/bass drivers and four 1″ domes. The rear-facing driver complement is aimed at the ceiling à la the new Atmos home-theater speakers. AudioKinesis emphasizes the need for controlled reverberation to support a believable rendition, hence the upward firing drivers and bipolar design. As with several other rooms at the show, one of the sources was the new $3,000 Exogal Comet Plus DAC.

Speaking of reverb and Atmos, there were a few multichannel rooms, two of which were quite compelling. My buddy George Klissarov at exaSound (www.exasound.com) occasionally had a 5.0 Magnepan rig playing, with his e28 DAC as source and Papa Nelson's amps for gain.

On a grander scale, my buddy Mark Waldrep, an intrepid explorer of heavenly HRA, had a monster setup in the lowest level, proof that home theater can also be hi-fi. His signal chain started with a "…OPPO Digital BDP-103 (factory, $510) modified to output three S/PDIF streams to allow for digital connection to three Benchmark ($2,000) DAC2 HGC converters (into) five Benchmark AHB2 power amplifiers ($3,000) set in bridge mono mode, which are capable of 130 dB plus of dynamic range." DH Labs interconnects and speaker cables fed five Revel Salon 2 speakers ($9,000) while a JVC DLA-RS-67U up-converting UHD projector ($6,700) and a Stewart Filmscreen provided visual accompaniment (see **Photo 12**). The audio content was recordings Waldrep produced for AIX Records, and losslessly encoded with Dolby TrueHD.

Always drawn to supertweeters, the Soundfield Audio Dipole 1 ($12,000) is a semi-active, open-baffle design featuring a 8.5″ × 11″ planar magnetic horn-loaded mid-tweeter covering 700 to 14 kHz, and a 1″ horn supertweeter taking care of 14 to 28 kHz (see **Photo 13**). An open baffle 12″ Eminence Kappalite 3012LF handles 60 to 700 Hz, while a 9.5″ × 6.5″ planar magnetic horn-loaded mid-tweeter (700 to 14 kHz) fires rearward and out of phase. The active aspect addresses the shortcomings of the usually anemic very low-frequency quality of open-baffle designs, with a 12″ Rythmik servo'd sub addressing the 20-to-60-Hz range. The Dipole 1 was powered through an external box with a 350 W servo–amp and a passive crossover for the remainder of the drive complement. The rig was driven by the Power Modules VT-01 v2 preamp ($5,495) into the Power Modules MB-60 monoblocks ($9,995/pair) all wired up with about $26,000 worth of Equilibrio Level 3 from Italy's Viero Cables (www.vierocables.com). A fully active version of the Dipole 1, featuring Hypex Class-D amplification, can be special ordered (www.soundfieldaudio.net).

Photo 14: Watch 'em float! Technic's SE-R1 Class-D amp supports the SU-R1 Network Audio Control Player.

Photo 15: TIDAL Audio's big rig featured the new Contriva G2 tall tower floorstanding speaker, plus the Presencio preamplifier, and 380-W Impulse monoblocs.

Photo 16: Living large—Cabasse's new La Sphère is a grander version of the L'Océan, and employs a generous 22″ honeycomb dome woofer behind a triaxial 1.1″ polyether tweeter, 4″ P2C annular mid, and an 8″ Duocell annular low-mid.


Photo 17: Sadurni Acoustics' visually striking Staccato features old-school, multiway horns.

Players and Components

As with generations before, GenXers and flush Millennials look back to a "simpler time" when vinyl and console stereos ruled the roost. The folks at April Music (www.aprilmusic.com) have taken that demographic to heart with the new version of its all-in-one "music center." The Aura note V2 ($2,750) includes a CD transport, FM tuner, and a 125 W into a 8 Ω ICEpower ASX250 stereo amp. The Aura note V2 also includes a version of April Music's respected DAC with 192k USB in, as well as a flash drive input, aux ins, and a headphone out.

Another retro trend is the increasing use of moving coil meters. The men of Matsushita have gone one better with their latest offering. Technics' $17,000 wideband (–3 dB from 1 to 90 kHz) SE-R1 stereo power amp sports the biggest, long throw meters I've see since my days at dbx (see **Photo 14**). More importantly, its Class-D architecture departs from most industry practices by using a very high 1.5 MHz switching rate for its modulator. The SE-R1 was part of an all-Technics system (www.technics.com), including a SU-R1 Network Audio Control Player ($9,000), which handles DXD and DSD2, as well as the SB-R1 towers ($13,500). Like the Vinnie Rossi LIO mentioned earlier, the SU-R1 uses off-line, charged capacitors to power itself, and the quad woofer and coaxial mid-tweeter SB-R1 is notable, like its companion amp, for also exhibiting wideband response, with output down 10 dB at 90 kHz.

By now, you should all know about the recently purchased TIDAL, the lossless streaming service provided by Aspiro AB of Sweden. Well, there's another TIDAL, all caps no less, and this one is from Germany. The 16 year old TIDAL Audio (www.tidal-audio.de) manufactures electronics and speakers. And, AXPONA 2015 was the event to premiere a bunch of new gear in the US.

To wit, the new Contriva G2 tall tower floorstander ($69,700), plus the Presencio preamplifier ($78,000), and 380 W Impulse Monoblocs ($65,000). Sourced from the highly respectable Bricasti M1 DAC ($9,000), the setup was woven together with Purist Audio Design's Luminist Revision cables, and supported by a StillPoints rack and isolation (see **Photo 15**).

One of the best sounding rooms at the show was also possibly the most costly set of gear as well. The Cabasse folks (www.cabasse.com) have been making speakers for more than 60 years in France, and they are know for their small form factor, distinctive globe-shaped coaxial and triaxial lines. For AXPONA, Cabasse teamed up with Esoteric, premiering the gargantuan $180,000 La Sphère concentric four ways hooked up to $188,000 worth of Esoteric electronics (see **Photo 16**). The active La Sphère model is a grander version of the L'Océan, and employs a generous 22" honeycomb dome woofer behind a triaxial 1.1" polyether tweeter, 4" P2C annular mid and an 8" Duocell annular low-mid. With AudioQuest cabling, Shunyata power distribution, and Auralex acoustic treatment, the audio was wonderful.

Speaking of an over the top assemblage of gear, Sadurni Acoustics (sadurniacoustics.com) teamed up with Blue Smoke (www.bluesmokesystems.com), Stillpoints (www.stillpoints.us), and Merrill Audio (www.merrillaudio.net) to create some of the best sounding playback using horns I've heard in a while. Sadurni's giant triple horn-loaded Staccato ($40,200), shown in **Photo 17**, was driven by Merrill's Veritas amps ($1,200). What's noteworthy is that the Veritas are based on Hypex's Ncore technology. For $12,000, you get a great sounding 1,200 W into 2 Ω monoblocks. Merrill Audio also has a stereo version, the Taranis ($2,500), which uses the same Ncore gain stages and seems to be quite a bargain.

Though I started out a skeptic, I'm now a believer in the AXPONA way of life, and will be back next year for a speaking engagement and more new gear to explore… Until then, thanks for reading! For more information about AXPONA, visit www.axpona.com.

About the Author

Oliver. A. Masciarotte is a graduate of the Lowell Institute of MIT. Masciarotte has been the principal of Seneschal, a consultancy to the professional audio and rich media industries, for more than 25 years. He has authored more than 100 trade articles in print and authored *To Serve & Groove*, a book about file-based music playback for the home.

Meet the FabFilter Pro-MB and the Pro-Q2

Beauty in the Sound

In audio software, usability is closely associated with the graphic user interface (GUI) and it is at least as important as the quality of the results. Unfortunately, we see potentially good software products harmed by poorly designed graphic interfaces. This article is about FabFilter's two most recent releases—the Pro-MB and the Pro Q2 plug-ins, which combine great GUIs with quality audio results.

By
Fernando Rodrigues
(Portugal)

FabFilter is a small software company established in the Netherlands, allegedly born from the fascination its founders felt for the old analog audio technology and the possibilities of modern technologies, intensified with some dissatisfaction regarding the quality of the software available to make music and the interaction with those tools.

History of FabFilter

FabFilter's first product was a synthesizer, the FabFilter One, a proof of concept that more could be achieved regarding the quality of filters available with virtual instruments—hence the name FabFilter.

FabFilter quickly gained recognition for the quality of its products and its GUI approach. The FabFilter Volcano and the FabFilter Twin, which followed the FabFilter One, firmly established the company as a name to be taken seriously. The GUIs were not common. They didn't follow any known paradigm, neither were they trying to recreate hardware from the past. Instead, they created tools adapted to work in a computer environment. These GUIs were, and have been ever since, well adapted to that purpose, with clear and well-designed graphics, allowing an almost physical and visual signal flow with built-in analysis tools that really help get the job done.

Other companies followed this paradigm, some of them more or less at the same time, some a little later, but none, in our opinion, did it as well as FabFilter. And the users paid homage to FabFilter, which is today recognized as one of the leaders in the music software industry. For those developers who think sound quality is everything, well... think again.

Introducing the Pro-MB

We all know what dynamic compression is, and what it does to audio. Multiband compression simply divides the audio into several bands and applies different compression settings and ratios to those bands. An iZotope tutorial mentions the classic example of a bass drum and a guitar. In this example, we have to deal with the attack transients, and with preserving the "punching" of the bass while not allowing the guitar sound to become "harsh."

FabFilter
Pro-Q 2: channel and bus EQ
Pro-MB: multiband compressor
Available in: VST, VST3, Audio Units, AAX Native, and AudioSuite formats (64 bit and 32 bit), as well as RTAS (32-bit)
www.fabfilter.com

This would be almost impossible using single-band compression. Also, when dealing with a complex signal (e.g., a full mix) intermodulation may appear (making all the audio volume suddenly go down when there are peaks being compressed). That's why, when we are in the presence of complex signals—which happens frequently in mastering—a multiband compressor may be handy.

Usually, the approach to multiband processing consists of splitting the incoming signal into bands using a set of crossover filters. The FabFilter concept differs, approaching the audio from the perspective of the musician and the audio engineer—who think of audio in terms of bands, not crossovers. Thus, instead of defining the crossovers, we define the audio region and the frequency range. With FabFilter's graphic approach, this couldn't be easier. The GUI clearly shows that the rest of the audio remains untouched. (We also have the usual analysis tools, which give us the visual feedback to support what we are hearing.)

Pro-MB supports up to six processing bands, freely placed anywhere in the spectrum through the GUI and the handles. It works in three ways, which FabFilter designates as unique dynamic phase, linear phase, and minimum phase. The last one, minimum phase, is the traditional way MB processors work. It can process dynamics in any way, from the highly transparent compression to limiting, expansion, and even pumping upward compression and punchy gating. (These last features, we have to confess, are not on our list of pleasant ones, but there are users who like them.)

Besides it's more obvious use as a master multiband compressor, the Pro-MB can also be used as a regular compressor, as an alternative to the Pro-C, for example, with different results.

Presenting the Pro-Q 2

The equalizer is one of the oldest and most used tools for audio. The Pro-Q2 is the second version of one of FabFilter's most celebrated audio processors. There are some other EQs that are often quoted when people think about this kind of tool, but having used Pro-Q since its launch, this was always one of our main tools because of its transparency—it can also add coloration if we want—and because it's so easy to use and get good quality results.

We still have premium features such as up to 24 bands of EQ, the large and highly praised GUI with GPU-powered graphics acceleration, a spectrum analyzer, an intelligent band solo mode, and several range modes for special purpose tasks (e.g., mastering, mixing, etc.).

There are several new features in this version including the natural phase mode, which according to FabFilter, not only perfectly matches the magnitude response of analog EQ'ing, but also closely matches the analog phase response. Also deserving special mention is the Spectrum Grab. We simply have to look at the spectrum analysis, rest the mouse pointer over the curve, and select a particular frequency we want to edit. Pro-Q instantly creates an EQ node there, and opens the EQ controls in the base. It's as easy as it can get.

Finally, we have to mention EQ Match, which enables users to automatically match the spectrum of another track input through the side-chain, an improved linear-phase mode, new EQ Tilt and Band-Pass shapes, several gain tools (i.e., auto-gain, gain scale, and gain-Q interaction), an EQ Match mode, and

The FabFilter Pro-MB and Pro-Q2 plug-ins are used in Cubase.

We tested Pro-MB as a "regular" compressor and it did a good job. We like when a new tool goes beyond the obvious and Pro-MB does that.

Pro-Q 2 was harder to evaluate. The predecessor was already very good. FabFilter says it changed the Linear Phase modes, using a different internal processing method, leading to better sonic results, especially in large-scale projects.

a piano keyboard display. (This may be considered a weird thing to include in an EQ, but musicians may find it easier to identify frequency ranges that way.)

At Work

The first tests were conducted in Reaper, using an old project containing a piano track, a sax track, a drum track, and another track with some string orchestra and other sounds. The channels were already filled with FabFilter processors, so it was a matter of replacing Pro-Q with Pro-Q 2 and using Pro-MB on the drums and string orchestra parts. We are aware that Pro-MB wasn't originally intended to be used on single tracks, but these two were different since the drums were already a sub-mix containing all the instruments of a regular drum kit and the orchestra is, by its own definition, also a sub-mix.

Using Pro-MB gave emphasis to those bands that needed it and enabled us to obtain a sound that was more coherent and detailed. Of course, we used Pro-MB on the master bus too, together with Pro-L, and this is where it really shined. We were also able to make some comparisons with the previous Pro-C (whose results weren't bad at all). In the tracks, we used the same settings for Pro-Q 2 that were used with Pro-Q, and we had the chance to make some A/B comparisons, and also use some of the new features, such as the Spectrum Grab, to make some surgical edits from there.

The A/B comparisons showed that the sound resembled what we already had with Pro-Q (same settings give the same results). So, it was not a matter of improving the sound. Pro-Q always sounded great. Of course, FabFilter claims that the new natural phase and the updated linear phase improve the sound, but frankly, we couldn't hear it.

We were already happy with Pro-Q, and the workflow improvements with Pro-Q 2 more than justify the upgrade (that Spectrum Grab is amazing, indeed). After questioned about this, FabFilter clarified that the sound improvement would be more noticeable in large-scale projects with many instances of the Pro-Q2 in action.

Another thing that amazed us was how light these are on CPU. The project was of modest proportions, but each channel had Pro-G, Pro-Q 2, and Pro-C or Pro-MB, and the Master Bus had Pro-MB and Pro-L. All this never went over 6% with the CPU loading. This happened on our iMac i7 2.8 GHz running the latest OS X 10.9.

More Testing

We moved to a more powerful system, and to Windows 7, which gave us the opportunity to test the plug-ins in VST3 format, inside Steinberg's Cubase. For testing purposes, we used files we had previously used to test the Solid State Logic (SSL) Duende Native plug-ins because we wanted to see how the

The Pro-MB is a truly versatile tool, able to be used as a regular compressor as well as a master multi-band compressor. The "light" CPU load and the Spectrum Grab in Pro-Q2 also convinced us.

About the Author

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

Resource

"Multiband Compression Basics: iZotope Mastering Tips," iZotope, www.izotope.com/en/community/blog/tips-tutorials/2014/06/multiband-compression-basics-izotope-mastering-tips.

FabFilter Pro Q2 and the Pro MB behaved when compared with a well-established sound paradigm and with an audio session that was already prepared. So, we settled for those plug-ins.

The first challenge was to somehow reproduce the SSL chain. Since the Gate was barely used, we opted out (we could have used Pro-G, of course). For the compression, we used the Pro-C, and for the EQ, the Pro-Q 2.

The Pro-MB was in charge of defying the all mighty SSL Bus Compressor (okay, they are not the same type, but again, it's just sound, and we were testing).

Pro-MB is more versatile than you might think. Since it is band oriented, not crossover oriented, we can use it in different ways. Using just one band, it actually makes a very good bus compressor. It doesn't have an Auto release, but the controls are very precise, so we chose a release of just 5%, which we found matched the previous compressor's results and adjusted the other parameters by ear.

In the end, the results were remarkably similar, which means that we can use the Pro-MB as a regular Bus Compressor, with the advantage of having a Linear Phase Mode. This tool is also very versatile, for instance its fantastic interactive multiband display.

System Requirements

Windows:
- 32-bit: Windows 8, 7, Vista or XP
- 64-bit: Windows 8, 7 or Vista (x64)
- VST 2/3 host or Pro Tools

Mac OS X:
- 32-bit: OS X 10.5 or higher
- 64-bit: OS X 10.6 or higher
- AU or VST 2/3 host or Pro Tools
- Intel processor

However, using the Pro-Q 2 to match an SSL EQ posed some challenges. When we open a new EQ instance, we do not have the traditional (and comfortable) panel with familiar knobs, where each knob has a function and, therefore, we intuitively start adjusting. Instead, we have that fantastic graphical display, which shows the spectrum in real time, and we have to "build" our EQ from scratch.

But it wasn't so difficult. Just a click and we had an EQ node, and immediately we got a window in the bottom of the display, with the control knobs. If we made a mistake, we have an undo option.

We even had the ability to copy the settings to a B panel, and switch to it to make further adjustments, and then jump safely back to the A panel. This, by the way, is common to all the FabFilter plug-ins and very helpful.

So, we just had to create a node for each of the bands and adjust the filters accordingly (low cut, bell, etc.), and put the values more or less where they should be. After that, it was a matter of listening and adjusting. We mainly used the Natural Phase here, since we found that this worked better for the character we were seeking to give to the sound.

In the end, we reached our goal and proof of concept—these tools are up to the task even when competing with the best, and the results are quite pleasing.

Conclusion

FabFilter plug-ins can be trusted. That's been our opinion since we first tried them, several years ago. The company has evolved, and launched new products, yet it has remained faithful to what may be seen as its principles—create good sounding products that are intuitive to use and well adapted to the computer environment.

The Birth of the World's First True Software Radio

Discover a radio transmitter, code named Pizzicato. This development platform allows complete freedom of modulation schemes, encoding, carrier frequencies, and more.

By
Monty Barlow
(Director of Wireless Technology, Cambridge Consultants)

Good radio spectrum is a scarce resource—only low frequencies (a gigahertz or lower) propagate well over distance or through walls so they are in great demand. For radio, greater efficiency requires the use of dynamic or "cognitive" techniques to sense the radio environment and switch parameters on the fly. This could give access to more of the estimated 90% of the allocated spectrum, which is not in use at any one time.

There is something almost magical about radio electronics. It takes a message (e.g., voice or data), launches it invisibly into the air, and reliably recovers it many miles away.

Over the last century, radio engineers have refined the analog circuitry that makes this possible—striving for higher performance, power efficiency, and miniaturization. But now Cambridge Consultants has demonstrated a world's first—a radio transmitter, code named Pizzicato, built solely from off-the-shelf digital components.

New mathematical tricks allow standard digital computing to directly generate high-frequency radio signals. A Pizzicato-based on-chip radio would directly benefit from Moore's Law—shrinking in cost, size, and power consumption with each new generation of silicon fabrication. It could be programmed to generate almost any combination of signals at any carrier frequencies, nimbly adapting its behavior in a way that is impossible with conventional radios.

The last 20 years have seen many mature analog technologies made obsolete by digital replacements. Cathode ray tube (CRT)-based TV sets and computer monitors have been replaced by LCDs in every home and office, and the suspense of waiting for holiday photos to be developed from 35-mm film has been replaced by instant review and social media sharing. Now, for the first time, the same is happening to radio technology. A team at the product design and development firm Cambridge Consultants has created and demonstrated the world's first software radio transmitter.

Isn't Radio Already Digital?

There is a common misconception that radio is also now a digital technology. This may stem from standards such as Digital Audio Broadcasting (DAB), which convey digital data over radio links, or the labeling of radio products with terms such as "digital" and "software-defined." Regardless of whether a radio link is carrying data of analog or digital origin, radios are still very much analog in 2015. For example, a 4G LTE cell phone (which applies more processing power to building and analyzing data packets than a 1980s supercomputer) still uses predominantly analog technology for handling multi-gigahertz (GHz) carrier frequencies.

Analog is Running Out of Steam

For all their downsides, CRT-based screens and 35-mm film are still scalable—only economics

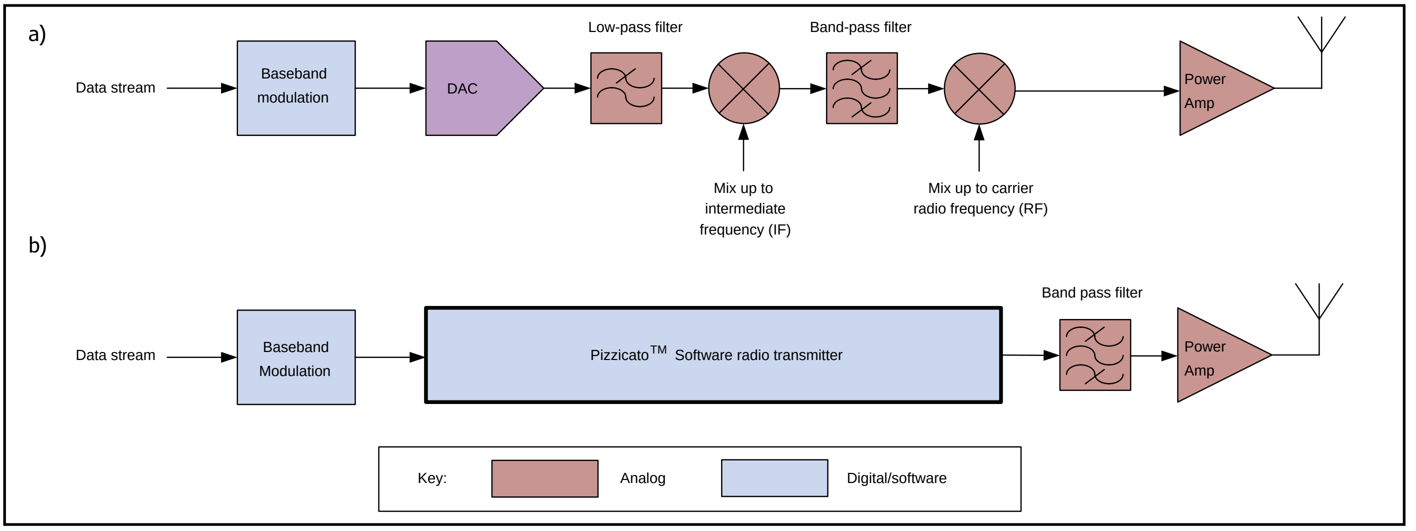


Figure 1: A standard software-defined radio transmitter (a) vs. a Pizzicato software transmitter (b).

prevented every person on the planet from taking photos or watching a television set. Radio is different. Only frequencies below a few gigahertz propagate well through walls or over distance so these desirable frequencies are in great demand. We have all experienced poor connectivity because of the dense concentrations of users in gridlocked traffic or at a major sports event. If the next billion people are to gain access to mobile broadband, or the "Internet of Things" is to scale to 100 billion devices, we need to change our approach. We cannot just manufacture more of today's products and ignore the resulting congestion of the airwaves.

Radio designers are exploring many avenues. Radios will become increasingly smart (cognitive) in how they scavenge for scraps of unused spectrum in real time. They will push up to higher—less desirable—carrier frequencies, using techniques such as beamforming from multiple antennas to concentrate precious transmitted power in only the directions needed.

As these changes are realized, the analog sections of radios will become an increasing bottleneck—crowding numerous high-frequency analog radios together on one chip and switching their operational parameters every few microseconds is a bad idea.

Waiting for Moore's Law to deliver higher analog performance is not an option either. Unlike the digital circuitry used in CPUs and RAM, analog circuitry does not typically improve over successive generations of silicon chips. For example, a capacitor's physical size is defined by a handful of physical constants and equations. Making chips with finer features or lower core voltages does not change these fundamental limits.

Is Truly Digital Radio Possible?

In 2014, a team at Cambridge Consultants embarked on a research program to completely eliminate all analog parts in a radio, initially focusing on the radio transmitter. The idea was to apply rapidly evolving technology driving the Internet revolution—the ability to cheaply transport many gigabits/second across circuit boards and down fibers—to the relatively stagnant field of radio design.

Figure 1 shows the concept. The radio transmitter was code named "Pizzicato," which is a word familiar to players of stringed musical instruments as the term for very short, plucked sounds. It replaces the classic digital-to-analog converter (DAC) and the one or two stages of up-conversion in a radio with purely digital processing. Apart from a final analog filter (to prevent disruption of users of adjacent spectrums), this is a purely software-based (i.e. truly digital) radio.

Delta-Sigma Conversion

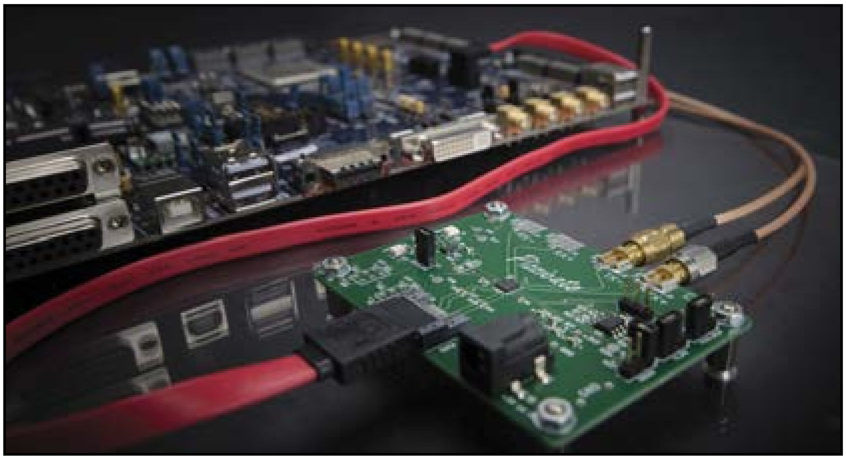
Audio enthusiasts will be familiar with the "1 bit" converters used in CD players from manufacturers such as Technics. Instead of using a 16-bit parallel DAC to convert digital audio samples to analog at 44.1 kilosamples/second, such a player would produce a digital bitstream at 64 times the audio sample rate (e.g., 2.8224 megabits/second). The output of this so-called delta-sigma converter was a digital signal containing a hi-fi signal down at audio frequencies, and unwanted noise at higher frequencies. A simple low-pass filter was used at the output to generate high-quality audio. Classic DAC problems such as nonlinearity were neatly avoided.

Pizzicato by contrast is a band-pass delta-sigma converter—similar in concept to low-pass audio converters but with the hi-fi region moved up to radio frequencies. The unwanted noise ends up both below and above the hi-fi region in frequency.

Figure 2 shows a notch in the noise in which we can place our wanted radio signal. Visualizing how such a converter works is tricky, unlike with low-pass designs where the wanted signal is just a smoothed

About the Author

Monty Barlow is a seasoned leader in the wireless industry. He joined Cambridge Consultants in 1994 and is currently the Director of Wireless Technology, where he oversees the Digital Signal Processing (DSP) groups in the company's Wireless Division. Monty has created novel development processes that enable creative thinking to be quickly captured and rolled into high-quality, cost-sensitive products. He has led several major programs in creating world firsts in areas including broadband communications, ASICs, and speech processing.



Cambridge Consultants' Pizzicato platform is a radio transmitter built from off-the-shelf digital components. It enables users to experiment with the potential of a software radio transmitter. The first demonstration allowed us to create 14 simultaneous cellular base station signals.

version of the bitstream. For those mathematically minded, the idea is that we can construct a signal whose sideband (normally unwanted in a low-pass design) happens to fall in our region of interest and contains our wanted signal.

Making It Work at Radio Frequencies

Band-pass delta-sigma converters are a proven technology, and have been successfully used in applications up to many tens of megahertz. The problem with addressing radio carrier frequencies of many gigahertz is that the digital bit rate must be many gigabits/second (Gbps)—at least twice the radio carrier frequency. Since a delta-sigma converter is iterative, each transmitted bit depends on the history of all previous bits. Previous "divide and conquer" techniques attempted to share out the computational load between multiple processing blocks introducing significant distortion.

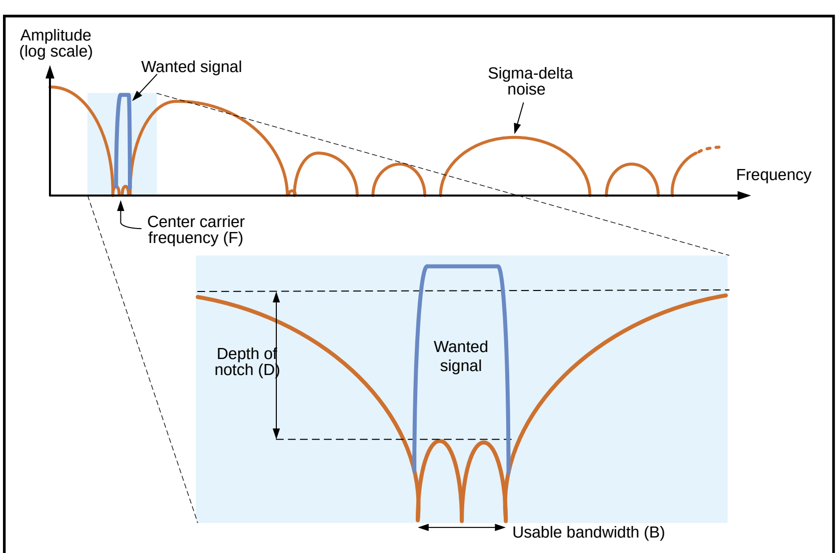


Figure 2: The transmitted signal (blue) is placed in a notch in the delta-sigma noise (orange). Increasing the bit rate enables us to deepen the notch (D), widen the usable bandwidth (B), and increase the radio carrier frequency (F).

The breakthrough made by the Pizzicato team is a mathematical trick to decompose the total computation down into a number of slower, parallel, delta-sigma-like processes whose output can be combined into a single fast bitstream with minimal distortion (see **Figure 3**).

The first prototype used the SERDES port (the technology underlying SATA ports on computers) on a Xilinx Virtex-5 FPGA to output the serial bitstream. Software running in bespoke DSP cores in the FPGA builds baseband signals and then performs the Pizzicato processing. It outputs a 3 Gbps digital stream containing 14 GSM (Global System for Mobile Communications) base station signals spread over about 80 MHz in the 900-MHz cellular band. **Figure 4** shows two different views of this signal. Importantly, with the exception of the SERDES port, no part of the design runs quicker than 100 MHz, demonstrating how Pizzicato can be implemented in regular digital logic circuitry (ASIC, FPGA, or even CPU).

Potential Uses for a Software Radio

Unlike some previous attempts at limited software radios (i.e., fixed modulation schemes such as FM audio or carrier and symbol frequencies fixed at an integer fraction of the bit rate), Pizzicato is fully flexible. The demonstrator can generate any combination of signals with any modulation schemes in its 100-MHz pass-band, and the carrier frequency can be selected to a resolution of less than 1 Hz. Faster bit rates will offer higher carrier frequencies and wider passbands—SERDES at 28 Gpbs is widely available today.

Some interesting possibilities include fast hopping between disparate parts of the radio spectrum. Analog radios struggle to quickly hop between different carrier frequencies (there is an inevitable settling time or need to round-robin multiple synthesizers when hopping), whereas Pizzicato suffers no such limitations. Taken to the extreme, it might be possible to use spread spectrum techniques to produce a waveform (say) 10 GHz wide.

The signal from the Pizzicato demonstrator is no louder than a quiet access point (~0 dBm). Most applications would require a power amplifier (PA). This could be a conventional analog PA, but more interesting is the possibility of directly driving a Class-S PA (similar to a Class-D audio amplifier). Designs for Class-S PAs are just starting to appear.

Since Pizzicato uses general processing resource to operate, radio resource could be built on demand by an operating system and de-allocated after use.

Implications for Developers

Today, those developing fundamental radio technology are members of an increasingly small club. The cost of developing an analog radio custom chip (ASIC) can exceed $100 million. Most companies and hobbyists are only integrating off-the-shelf modules such as Wi-Fi or a 3G modem, or building simple "old-school" radios from discrete components.

Software radio could change this, allowing developers with little more than a computer and a grasp of signal processing concepts to create custom radio designs with complete freedom of modulation scheme, encoding, carrier frequency, and so on. Companies that develop digital technology (e.g., microcontrollers) will be able to combine most of a radio on-die without using exotic fabrication options.

Cambridge Consultants is currently considering options for partnering with a few early adopters, and ways of making the technology more widely available in the longer term (perhaps FPGA-based development kits).

It's Early Days

Like an early graphical monochrome LCD that foretold the end of the CRT's reign, Pizzicato suggests that analog technology will one day be phased out from mainstream radio applications. But, as with those early LCDs, it will take numerous iterations before Pizzicato can become a widespread technology.

The clearest limitation is that Pizzicato is only a radio transmitter. Although a pure software radio receiver is in development at Cambridge Consultants, it is far more complex and will take longer to mature. Also, some analog band-pass filtering is required to prevent interference to all other users of the radio spectrum. However, as the bit rate is pushed up, the requirements on this analog filtering can be relaxed.

It is the evolution over time that is key. Ever faster digital bit rates mean that Pizzicato is on a trajectory that automatically delivers benefits without redevelopment of the core technology—a stark contrast to the world of analog radio chips, which need to be "ported" to each new ASIC geometry.

The First Software Radio Product

It is impossible to predict with any certainty when wireless inventions will find their way into products. Consider the Orthogonal Frequency Division Multiplexing (OFDM) modulation scheme. It was invented back in the 1960s and even in the 1980s still seemed obscure. However, in the 21st century, it is a core technology underpinning Wi-Fi, LTE, and other wireless standards.

It could be five years before Pizzicato-like techniques find their way into ultra-flexible access points for future Wi-Fi variants or 10 years before handsets use exclusively software radios. It may even be that the flexibility and the performance offered by software radios allows future standards (5G onwards) to place demands on radios that would be impossible to meet today.

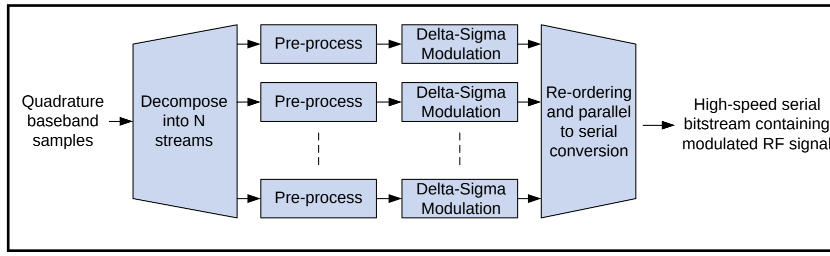


Figure 3: This is a simplified view of Pizzicato's processing stages.

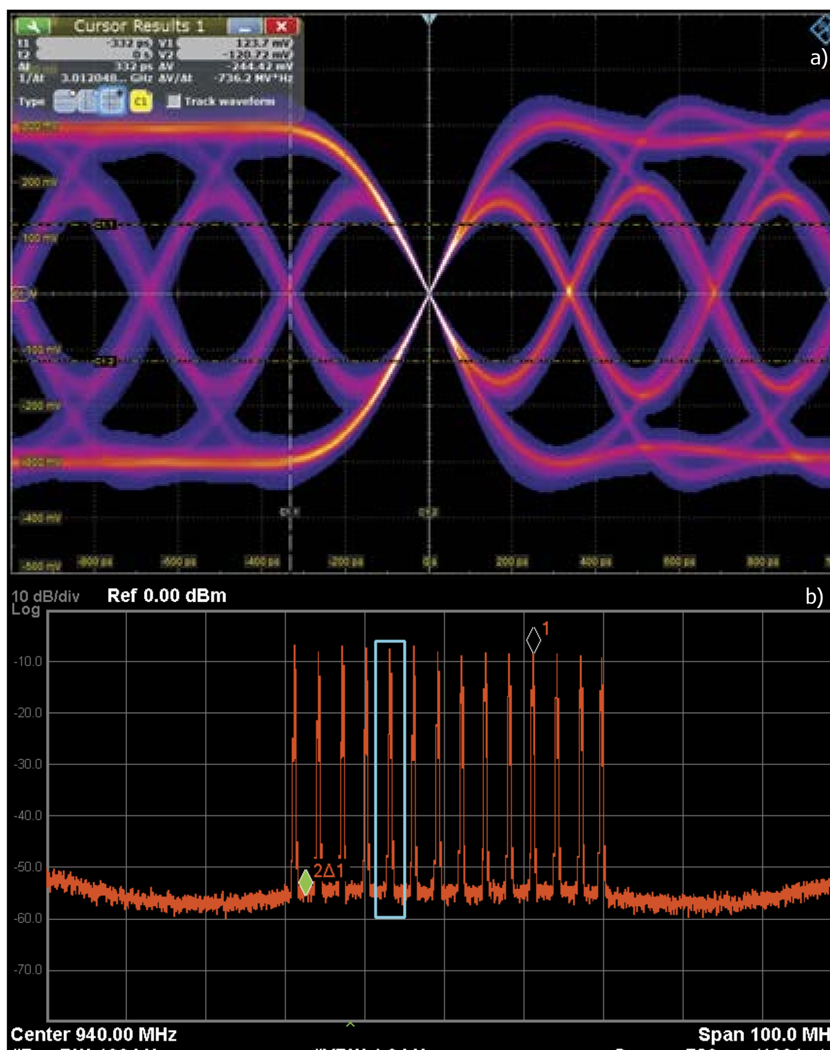


Figure 4: Here are two views of the same bitstream output from Pizzicato: the oscilloscope trace (a) and the spectrum analyzer trace (b) show the 14 base station signals (one highlighted in blue).

Acoustical Measuring with Rational Acoustics' Smaart

Photo 1: This flash screen appears when Smaart v.7 opens.

This month our Sound Control article focuses on the Smaart (System Measurement Acoustical Analysis Real Time) tool, which is a registered trademark of Rational Acoustics. Learn how Smaart was developed and receive an introduction to the system's uses.

By
Richard Honeycutt
(United States)

Smaart was developed by acoustical consultant Sam Berkow and touring sound engineer Alexander "Thorny" Yuill-Thornton II. Berkow and Yuill-Thornton founded SIA Software Co. in 1995 to produce Smaart. JBL then licensed the product from SIA. Smaart was first exhibited in October 1995 in New York City at the Audio Engineering Society (AES)'s 99th convention.

In 1999, EAW bought SIA and set it up as a division of the company. EAW hired Jamie Anderson, (formerly Meyer Sound's SIM and Technical Support Manager and UltraSound's Audio Engineer) to manage the product.

In 2008, Jamie and Karen Anderson, Adam Black, and Calvert Dayton established Rational Acoustics, which bought the Smaart brand, and are the current owners and developers. A major code redesign was incorporated in Smaart 7, which was introduced in 2010. This redesign resulted in reduced processing demands on the host computer.

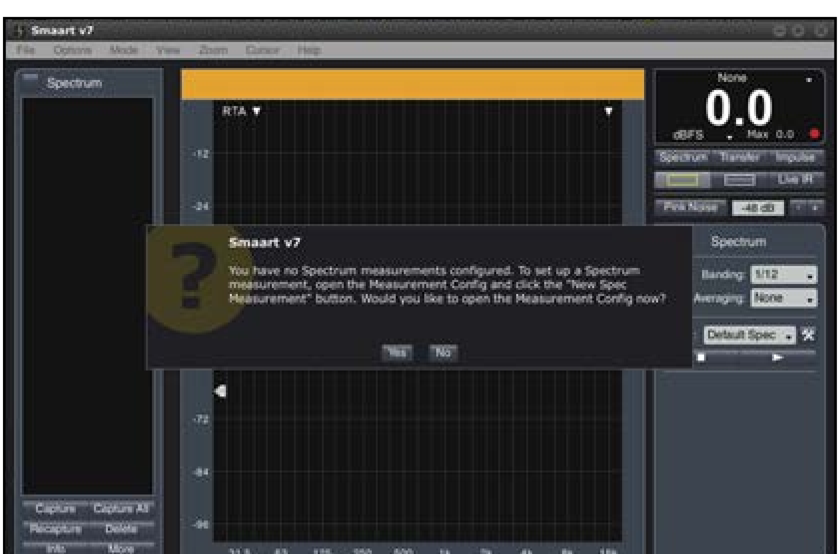


Photo 2: This opening screen appears after the flash screen.

Introduction to Smaart

Smaart is a dual-channel Fast Fourier Transform (FFT)-based software measurement platform. It has evolved through the years, from its beginnings as a Win 3.1/95-based tool to the current version 7.5, which uses object-based programming, enabling the user to run several simultaneous single-channel (spectrum) and dual-channel (transfer function) measurement engines, limited only by the PC's hardware capability. Smaart will run on either Windows or Macintosh computers.

Released in August 2012, Smaart version 7.4 featured an enhanced and redesigned impulse response (IR) mode that includes RT60 and early decay time (EDT) calculations (based on ISO standard methods), Clarity (C10, C35, C50, and C80), %ALcons (short and long), and a fully featured Speech Transmission Index (STI) measurement capability. Added to the IR Mode in the current version 7.5 is the ability to display the Schroeder integration curve, new generator sources, and a basic wave recorder.

Using Smaart

Photo 1 shows the flash screen that appears when Smaart v.7 opens. The first time the program is run, this screen will give the user an opportunity to register the software. After registering, the user will see the opening screen shown in **Photo 2**. If spectrum measurements are wanted, the user can click "YES" and configure the measurements. Since this column focuses on acoustical applications, we will examine the impulse response mode of Smaart. To enter this mode, the user can click on the "IMPULSE" button near the upper right-hand corner of the screen. Then, the display shown in **Photo 3** appears. (In general, any function accessed by a button on the Smaart screen can also be accessed through the menu. We will not describe menu access for commands that have button equivalents.)

Smaart enables the user to record either a single- or a dual-channel impulse response. Before recording a dual-channel impulse response, you must set up the signal generator function and define a transfer function pair. After clicking on the Options menu in the menu bar at the top of the screen, you will see the signal generator control panel shown in **Photo 4**. The photo shows the activated "Signal" drop-down box. Available signals include pink noise, pink sweep, sine, dual sine, and the ability to play a .wav file. The signal can be turned on or off using this panel. Once the signal has been selected, the signal button on the main screen is labeled with the signal type (notice "pink noise" on the main IR screen shown in **Photo 3**), and the signal source can be turned on or off by clicking on this button. The signal level can be adjusted by use of the + and – buttons beside the signal button on the main screen.

Clicking on the hammer-and-wrench icon on the IR screen brings up the Measurement Config window shown in **Photo 5**. The user can then click on "New TF Measurement" and Smaart will fill in the configuration table with default parameters, based upon the sound card in the computer. The user can then supply a name for the transfer function

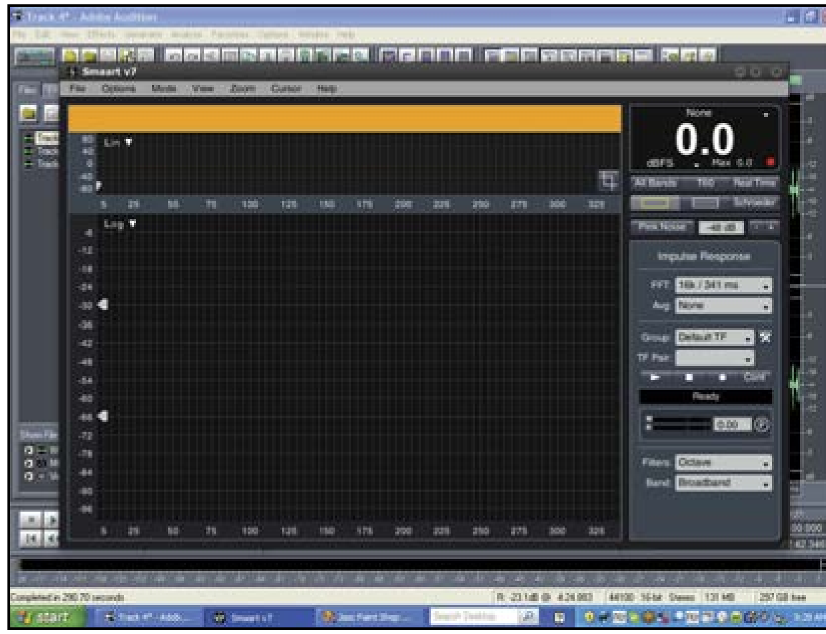


Photo 3: This is the opening screen for impulse response measurements.

Photo 4: The signal generator control panel enables selection of signal-source parameters.

Photo 5: The Measurement Config window enables the user to define transfer function pairs.

Photo 6: The recorded IR is shown in a time-domain view in the top window, and in log fashion in the bottom window.

Photo 7: The RT60 decay can be displayed for a chosen octave or 1/3-octave band.

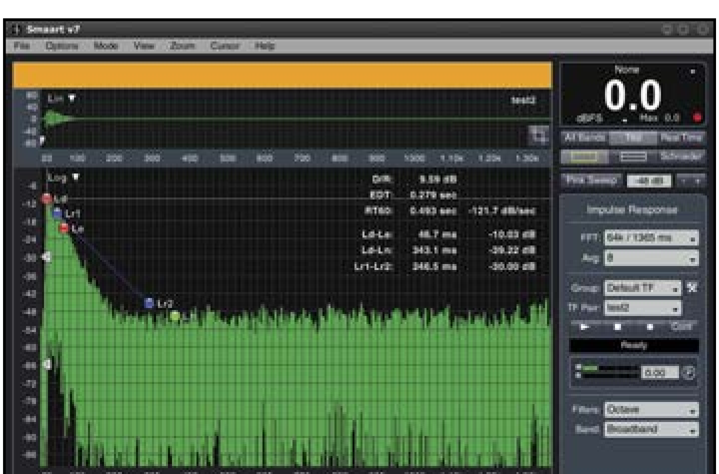


Photo 8: Pressing the Schroeder button adds the Schroeder reverse integration (gray curve) to the display.

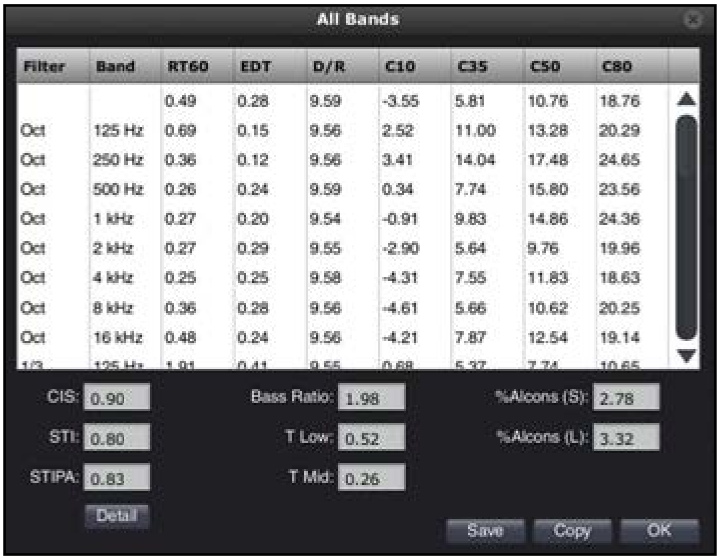
(TF) pair and click OK. If measurement configurations have previously been created, these will also appear in the table. The file tree at left allows creation of different measurement configuration groups, which can be selected and activated at will. Whichever configuration is selected will be listed by group and TF pair on the main IR screen (see **Photo 3**). FFT length and averaging can be specified via drop-down boxes on the main IR window.

At this point, the user can press the "play" button near the lower right of the main IR screen, and a dual-channel IR will be recorded. (For a dual-channel IR, the "measure" channel identifies the channel into which the test microphone is connected. The "reference" channel is the channel containing the signal provided to the test amplifier and speaker. Once the play button is pressed, the signal is turned on and the number of IRs specified by the "averaging" panel will be automatically performed. If the record button is pressed before the play button, a single-channel IR is recorded: the signal from the microphone is monitored and when an impulsive sound (i.e., balloon burst, starting-pistol shot, etc.) occurs, the impulse is then displayed. After either a single-channel or a dual-channel IR is recorded, a time-domain view of the impulse is typically shown in the upper screen, while a log view of the decay is shown in the bottom screen. The selection of linear, log, or Energy-Time Curve (ETC) display is made by clicking on the arrow in the upper left of either window, and choosing from the drop-down box that appears. **Photo 6** shows these displays.

Smaart Results

Once the IR is displayed, Smaart quickly calculates the results. The RT60 decay, calculated by either the T20 or the T30 method (depending on the IR's SNR), is plotted on the IR (see **Photo 7**). The octave or 1/3-octave band for which results are plotted is selected by the two drop-down boxes accessed in the lower right-hand corner of the main screen. In addition to the RT, the EDT, decay rate in decibels/second, and direct-to-reverberant (D/R)ratio are listed, along with the decay time and level values for the intervals between various markers:

- Ld = Level Direct. The arrival time of direct sound
- Le = Level Early (Decay). 10 dB down from the Ld marker on the reverse integration curve (see **Photo 8**). The slope between Ld and Le is used to calculate EDT.
- Lr1 = Level Reverberant 1. The top of the reverberant decay range, 5 dB down on the reverse integration curve from the Ld marker. (All of the level markers are user adjustable.)
- Lr2 = Level Reverberant 2. The end point for the reverberant decay slope. If there is sufficient dynamic range, it is placed 30 dB down the reverse integration curve from Lr1. If not, it is 20 dB down. Lr2 is one of

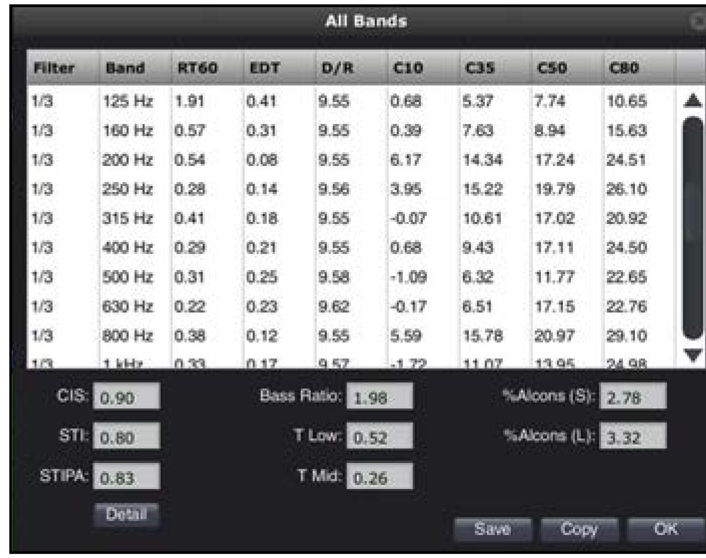


All Bands

Filter	Band	RT60	EDT	D/R	C10	C35	C50	C80
		0.49	0.28	9.59	-3.55	5.81	10.76	18.76
Oct	125 Hz	0.69	0.15	9.56	2.52	11.00	13.28	20.29
Oct	250 Hz	0.36	0.12	9.56	3.41	14.04	17.48	24.65
Oct	500 Hz	0.26	0.24	9.59	0.34	7.74	15.80	23.56
Oct	1 kHz	0.27	0.20	9.54	-0.91	9.83	14.86	24.36
Oct	2 kHz	0.27	0.29	9.55	-2.90	5.64	9.76	19.96
Oct	4 kHz	0.25	0.25	9.58	-4.31	7.55	11.83	18.63
Oct	8 kHz	0.36	0.28	9.56	-4.61	5.66	10.62	20.25
Oct	16 kHz	0.48	0.24	9.56	-4.21	7.87	12.54	19.14

CIS: 0.90 Bass Ratio: 1.98 %Alcons (S): 2.78
STI: 0.80 T Low: 0.52 %Alcons (L): 3.32
STIPA: 0.83 T Mid: 0.26
Detail Save Copy OK

Photo 9: Decay-related results for all bands can be displayed.



All Bands

Filter	Band	RT60	EDT	D/R	C10	C35	C50	C80
1/3	125 Hz	1.91	0.41	9.55	0.68	5.37	7.74	10.65
1/3	160 Hz	0.57	0.31	9.55	0.39	7.63	8.94	15.63
1/3	200 Hz	0.54	0.08	9.55	6.17	14.34	17.24	24.51
1/3	250 Hz	0.28	0.14	9.56	3.95	15.22	19.79	26.10
1/3	315 Hz	0.41	0.18	9.55	-0.07	10.61	17.02	20.92
1/3	400 Hz	0.29	0.21	9.55	0.68	9.43	17.11	24.50
1/3	500 Hz	0.31	0.25	9.58	-1.09	6.32	11.77	22.65
1/3	630 Hz	0.22	0.23	9.62	-0.17	6.51	17.15	22.76
1/3	800 Hz	0.38	0.12	9.55	5.59	15.78	20.97	29.10

CIS: 0.90 Bass Ratio: 1.98 %Alcons (S): 2.78
STI: 0.80 T Low: 0.52 %Alcons (L): 3.32
STIPA: 0.83 T Mid: 0.26
Detail Save Copy OK

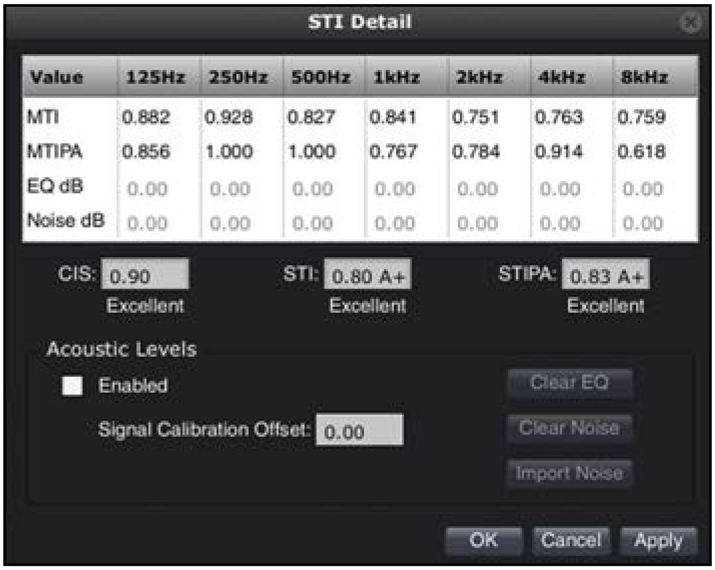
Photo 10: The 1/3-octave results are also tabularized.

the two markers that the user may sometimes want to adjust by hand; the other is Ln (below).

- Ln = Level of Noise. This is typically the most subjective of the five markers in terms of placement. The time at which the sound has decayed to Ln determines the start point for the reverse time integration curve, which is the basis for positioning all the other markers. Ideally, this will roughly correspond to the "saddle point" in the impulse response: where the decay slope merges into the more-or-less level noise floor. As of v.7.5, Smaart does a pretty good job of placing the Ln marker, but it may still benefit from user adjustment in some cases, particularly if the dynamic range of the measurement is marginal or there are significant distortion artifacts from a swept sine measurement or any other prominent anomalies in the noise tail of the IR being analyzed.

All decay-based results for all bands can be seen in tabular form by clicking the T60 button. The result is shown in **Photo 9**. These include the RT, EDT, D/R ratio, and clarity ratios. Although speech clarity (C50) and music clarity (C80) are the ratios most commonly used, C10 and C35 are presented as well. By scrolling down, the user can see 1/3-octave results, as illustrated in **Photo 10**.

At the bottom of the All Bands display, you can see Tlow, the low-frequency RT (an average of 125 and 250 Hz bands) and Tmid (an average of 500- and 1,000-Hz bands). Tlow divided by Tmid gives the bass ratio, which corresponds to the subjective impression of the room's support for low frequencies. Both %ALcons and STI intelligibility values are listed. The user can click on Detail to bring up detailed STI information (see **Photo 11**). In addition to the numerical results, qualitative ratings



Value	125Hz	250Hz	500Hz	1kHz	2kHz	4kHz	8kHz
MTI	0.882	0.928	0.827	0.841	0.751	0.763	0.759
MTIPA	0.856	1.000	1.000	0.767	0.784	0.914	0.618
EQ dB	0.00	0.00	0.00	0.00	0.00	0.00	0.00
Noise dB	0.00	0.00	0.00	0.00	0.00	0.00	0.00

Photo 11: Further information on the Speech Transmission Index (STI) can be displayed.

are also shown for Common Intelligibility Scale (CIS), STI, and STIPA.

In addition to serving as an IR recorder/analyzer, Smaart v.7.5 includes sound level measurement capabilities that may be of interest to those engaged in acoustical work. The Options button in the menu bar at the top of the Smaart screen provides a drop-down box in which SPL/LEQ options can be selected, bringing up the window shown in **Photo 12**. The weighting (none, A, B, or C) and speed (fast or slow) can be selected, as well as the LEQ interval in minutes. Logging can be enabled or disabled and the interval between logged measurements can be specified.

Overall Impressions

Although Smaart became famous as a platform for measuring transfer functions, spectra, and other electroacoustic parameters, the IR and noise functions make it quite useful for acoustical measurements. For more information, visit www.rationalacoustics.com/smaart/smaart-v-7.

Photo 12: Various options can be specified for measuring sound levels.

Build a Single-Ended Class-A Power Amplifier with 6C33 Tubes

The 6C33 tube attracts people's interest due to its design. Therefore, a single-ended Class-A power amplifier has a special charm. Due to the 6C33's low amplification, the driver stage should provide sufficient reserves. This one has been set up with the PCL 86 = 14 GW 8 vacuum tube, which is readily available.

By
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Single-ended Class-A power amplifiers are simultaneously advantageous and disadvantageous. Due to physical reasons, the efficiency is very low, which results in a relatively low effective output with a high power requirement. However, the unbeatable advantage is that the audio signal is not split up and put together again as it is with the push-pull Class-AB amplifiers. Therefore, crossover distortions, which often affect the sound, never occur.

However, you must take into account that each audio signal passes the zero-crossing and features quiet passages as well. In this case, crossover distortions have an impact (see **Figure 1**). The input signal is a pure sine wave (see **Figure 1a**). It is split into two half waves by the phase inverter and are put together again. If the characteristic curves of the two output tubes or of the transistors are not in a straight line on top of each other (see **Figure 1b**), the signal will be distorted in the zero-crossings (see **Figure 1c**). The distortions occur in a critical area. The only thing that can help is to set the idle current to a higher level, which leads to push-pull AB operation. The other approach is to use single-ended Class-A power amplifiers, which do not have this problem.

At hi-fi trade shows, high-efficiency horn speakers are typically operated with amplifiers producing more than 100 W at high volume. This extreme volume is justified by the achievable dynamics. Everyone can play speakers loudly. However, to produce an excellent sound at full dynamics even at low volume settings is an art.

To do so, the amplifiers must be hum free and low noise. At higher modulations, the distortion factor may moderately increase and should primarily consist of the second harmonic. High-efficiency loudspeakers operated with well-designed single-ended Class-A power amplifiers fulfill these requirements. The combination of single-ended

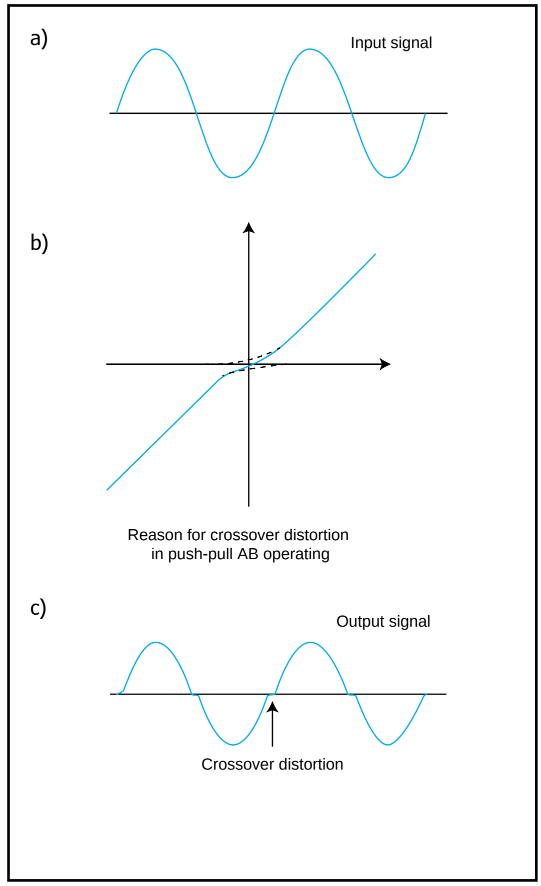


Figure 1: Crossover distortions can impact sound. The input signal is a pure sine wave (a). In an Class AB push-pull design, both characteristics of the output tubes for the negative and the positive half wave must be in line (b). Otherwise, signal will be distorted (c).

Class-A power amplifiers and high-efficiency full-range speakers—that are well balanced in terms of sound and correctly installed—has become increasingly popular. This type of power amplifier is ideal to work with loudspeakers in direct drive mode avoiding crossover circuits, which always cause phase distortions. Alternatively, two-way speakers with a minimum crossover are also well suited.

Loudspeakers and Crossovers

In the last 10 to 20 years, it has become common practice to build crossovers with impedance correction and complex filters. Beautiful frequency responses have been measured. However, it is not clear whether the speaker cabinets sound good or not. In the case of three-way speakers, a single crossover can easily contain more than 40 components. Therefore, the crossover price can rapidly exceed the cost of the loudspeaker chassis when using good components.

Loudspeakers represent an inductive and ohmic load to the amplifiers, featuring a true electromechanical life of their own, which does not improve the situation. But, if many inductances and capacitances are added in the crossover, different phase shifts develop over the entire frequency range, which causes reactive currents.

A reactive current means that in case of alternating current the energy oscillates between source and load. What is the negative feedback supposed to do in this case? Stereo amplifiers can't be completely realized without negative feedback, because it is needed to synchronize the two channels at any level and over the entire frequency range. Many people have searched for amplifiers that would cope with highly complex crossovers. However, these problems do not exist with broadband loudspeakers.

If you spend money on a quality loudspeaker chassis (e.g., two-way speakers), you can use a minimum crossover, namely an inductor connected in the path of the woofer and a capacitor installed to the tweeter. The amplifiers rarely have any problems with such speaker cabinets. If the efficiency of the loudspeakers is high and the losses of the minimum crossover are low, then low amplifier outputs are sufficient for producing high volume levels. It also follows that with such speaker concepts and single-ended Class-A power amplifiers, which are best built as monoblocks units, you obtain the best sound results. Hi-fi enthusiasts have proven this theory many times.

The 6C33 Power Tube

The 6C33 tube enables you to build an acoustically and optically attractive amplifier (see **Photo 1**). However, for operational safety reasons, the 6C33 should not be operated with too high anode voltage. With up to an anode voltage of slightly more than 200 V, the tube's life and operational safety are secured, but this is slightly at the expense of the performance.

There are different projects in which 20 W and more are achieved with one 6C33 in a single-ended Class-A circuit. Of course, higher amplifier power can be achieved with higher operating voltage and higher anode power loss. In this case, however, the tube's operating point more or less runs off. If you

Technical Data

Power	>12 W (clipping limit). THD = 3%
Frequency response at 8 W	10 Hz (–0.5 dB)
	33 kHz (–1 dB)
Frequency response at 5 W	10 Hz (–0.5 dB)
	56 kHz (–1 dB)
Harmonic distortion at 5 W	THD = 1%
	second harmonic = 0.73%
	third harmonic = 0.32%
	fourth harmonic = 0.24%
	fifth harmonic = 0.20%
Noise voltage unweighted	–72 dBV (20 Hz to 20 kHz) = 250 µV
Noise voltage	–77 dBV (A-weighted) = 140 µV
Input impedance	33 kΩ (see text)
Input voltage for maximum output level	1 V
Bias current of the 6C33	200 mA

Special parts, transformers, filter chokes and kits are available from EXPERIENCE electronics (www.experience-electronics.de).

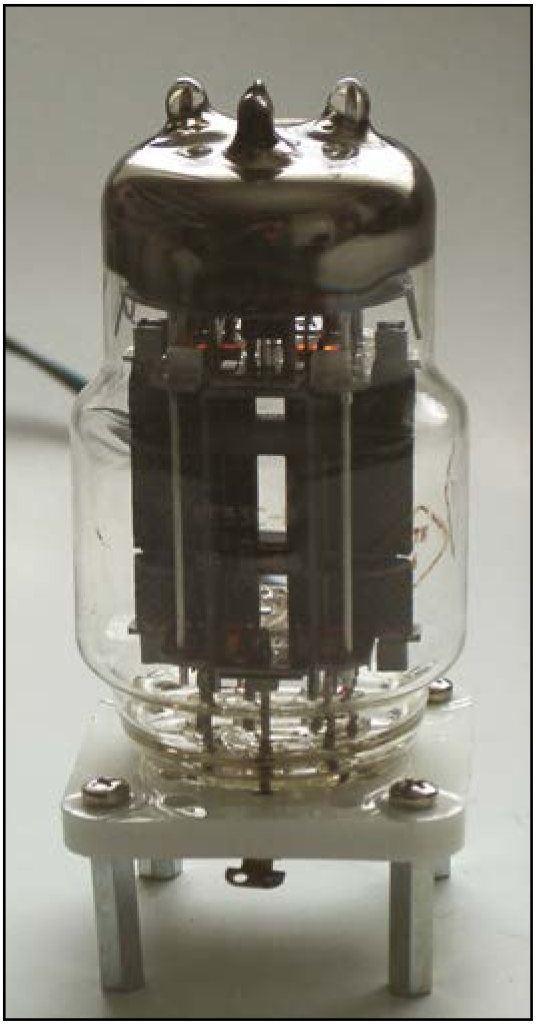


Photo 1: The Russian 6C33 is a military current regulator triode, which can provide up to 600 mA.

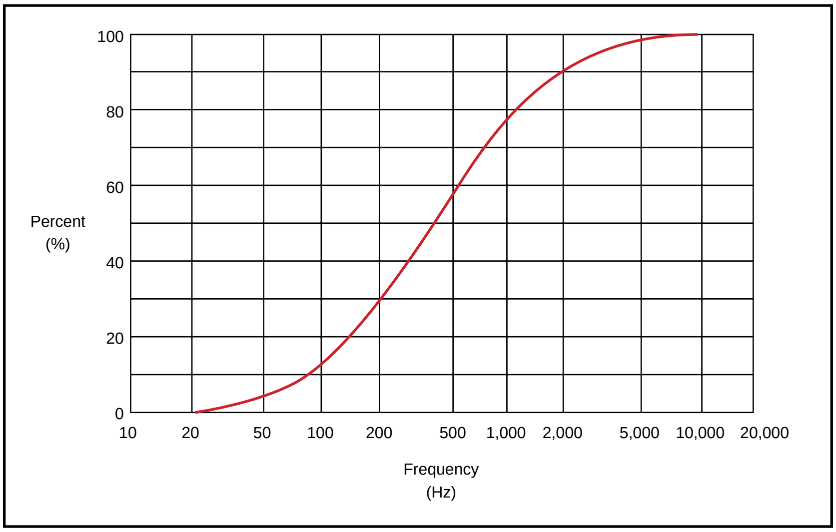


Figure 2: Different music signals vary in statistical power.

push the upper limits, there will be more failures. Therefore, a lower operating voltage and only about two-thirds of the anode power loss should be used.

Depending on the copper resistance of the output transformer, the operating voltage can be higher. With a bias current of 200 mA and a copper resistance of approximately 150 Ω, the transformer drops 30 V. This copper resistance can be reduced using a bigger transformer core with a bigger iron cross-section. This reduces the number of windings required and increases the wire cross-section, which reduces the copper resistance. However, due to the long winding length, the copper resistance doesn't drop to the desired level. Furthermore, the unfavorable winding geometry causes higher transformer capacitances that cut the upper limit frequency.

So, it is always a compromise and a matter of price. If you can place the highest requirements on the transformer, many things are feasible. However, everything also has a price! You can easily spend $250 to more than $350 for a good transformer built with the best materials and elaborate winding. The lower limit frequency is important as some church organs are able to produce their lowest tone at a frequency of 16 Hz. It is helpful if the amplifier is able to cope with that so there should be no significant drop at the upper frequency limit of 20 kHz. The achievable bandwidth depends on the output transformer's quality and on the amplifier's design.

Output Power Requirements

At this point, it is worth mentioning a few things about an amplifier's required output power performances. **Figure 2** shows the statistical power distribution of music signals. It shows that 70% of the performance uses frequencies up to 200 Hz and 80% at frequencies of up to 1 kHz. So, it would be sufficient if an amplifier had only 20% of its power at frequencies above 1 kHz. However, the problem is that, in case of an early performance drop, phase shifts would also occur, which would affect the sound in a negative way.

Furthermore, electronic music often has even higher performance peaks in the upper frequency range. But, it makes no sense to widen the amplifier bandwidth up to the 100-kHz range because there is no signal in that range and neither the speaker nor the ear would cope with these frequencies. With VHF radios, 15 kHz is the limit, and CDs as well as records end at about 22 kHz. It is also important that at 20 kHz a precise signal processing takes place. Therefore, no significant drop can occur. If the drop occurs too early, there will be phase shifts and a loss in level in the audible range, which significantly

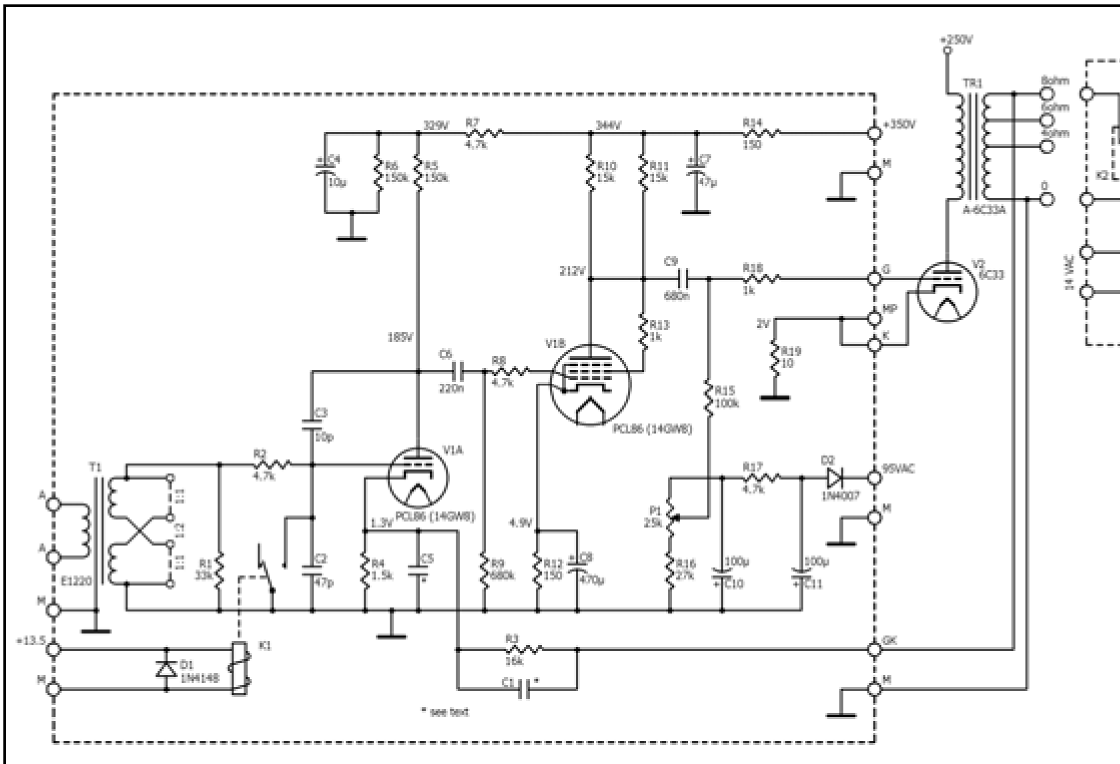


Figure 3: The amplifier's schematic shows a dotted frame, which indicates the portion that sits on the circuit board.

falsifies the sound.

Although the keynotes of the instruments and the voices require most of the power, the harmonics are crucial for the sound and the identifiability of the sound source. This means amplifiers must work precisely in this frequency range and the speakers and chassis must not destroy everything because of overloaded crossovers.

Due to the 6C33's moderate amplification, the preamplifier and the driver stage must sufficiently amplify the signal so that reserves are left for negative feedbacks and a sufficient input sensitivity, which is why I chose the PCL 86 (14 GW 8) for the preamplifier. The triode part corresponds to half an ECC 83 and offers high amplification. The pentode section, connected as a quasi-triode, offers further amplification and ensures the necessary low-impedance voltage swing for driving the 6C33.

The PCL 86 is operated with 350 V so that it will not prematurely limit, even during higher level control. With its 17 dB, the feedback factor of the negative feedback loop has been intentionally set relatively low. However, it can be increased at any time, which affects the input sensitivity at maximum level. You should not go significantly beyond 20 dB because then, even tube amplifiers tend to produce a "transistor sound."

Figure 3 shows the circuit diagram of the amplifier. The area marked with a dotted frame sits on the circuit board, the rest is freely wired. It makes

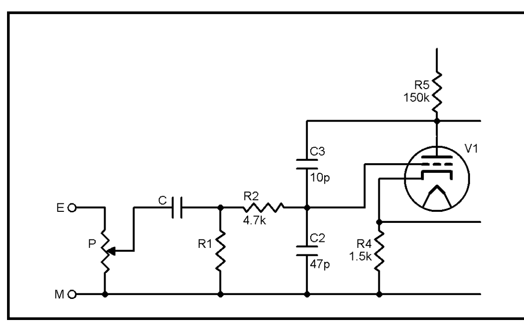


Figure 4: The input transformer can be replaced with a coupling capacitor.

no sense to mount the 6C33 to the circuit board because there is no print socket available for this tube and the heat emission due to a power dissipation of about 80 W would not be good for the surrounding components. The tube and the respective ceramic socket become quite hot during operation! The right dimensioning of the components is important for

About the Author

Gerhard Haas has written books as well as articles for Elektor Germany since 1995. *The Roehren Sonderheft* magazine was his creation, along with some partners. The first issue came out in 2005. Now, there are 10 issues containing a collection of interesting articles. Gerhard has also developed many electronic circuits and equipment. In the last two decades, he primarily worked on tube amplifiers and the circuitry around them, including the audio transformers, filter chokes, power transformers. He sold his company, Experience Electronics, several years ago to focus on designing tube amps and audio transformers, write articles for *Sonderheft* and finish his book *High- End Mit Roehren*. The first copy of his book was printed in 1995, and he is now working on an updated followup.

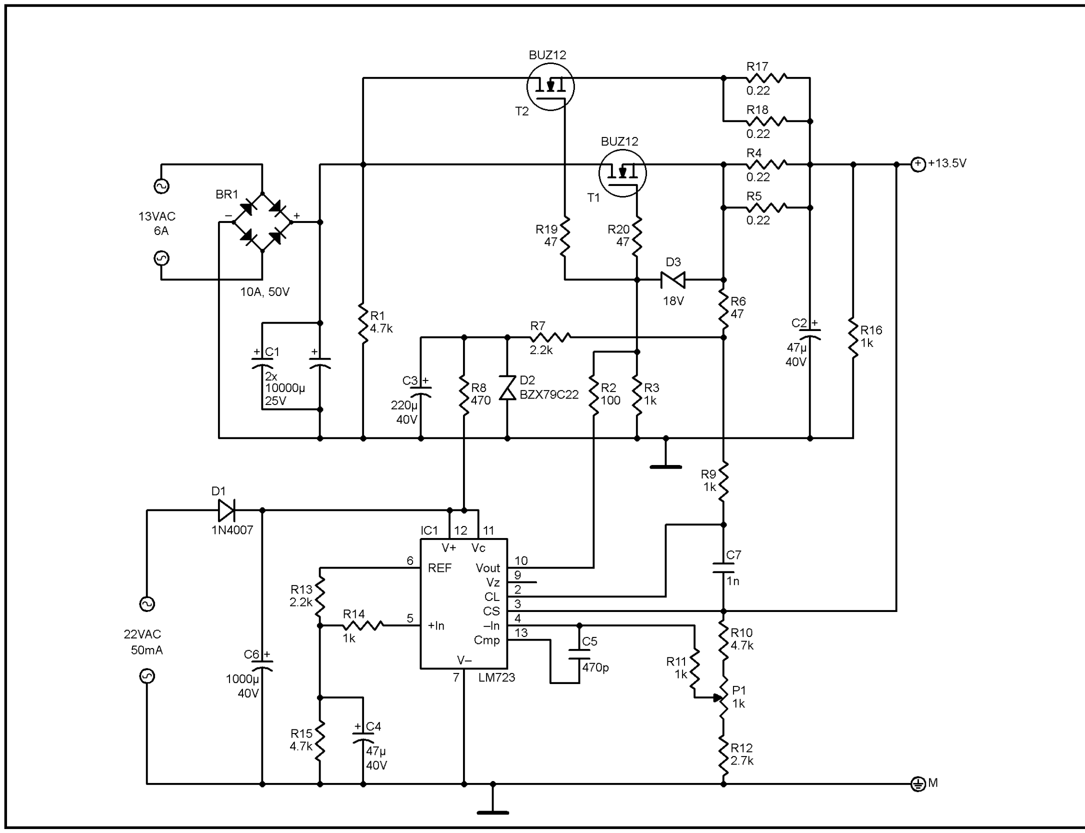


Figure 5: This is the DC heating circuit.

successful operation. The grid bleeder R15 of the 6C33 has relatively low impedance, which also requires a low-impedance drive. Therefore, the anode resistor of the PCL 86's pentode section is of low impedance.

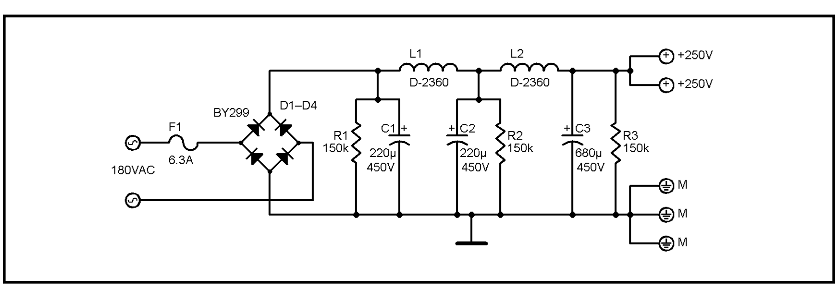


Figure 6: The 6C33's high-voltage supply is shown.

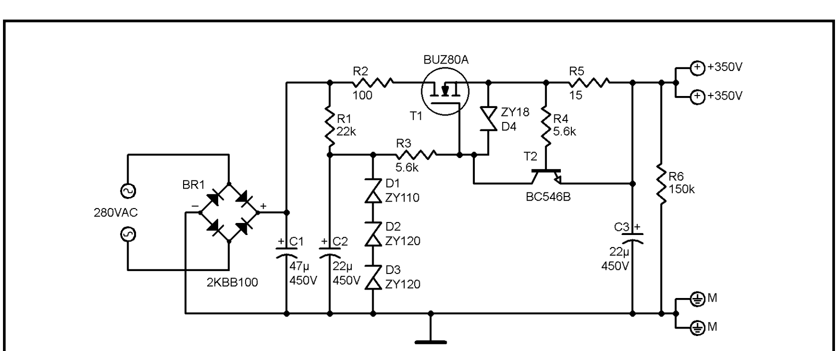


Figure 7: This is the driver stage's power supply.

Internet forums and some publications feature 6C33 circuits using grid bleeders up to 470 kΩ where clearly the tube's datasheet was not taken into consideration. It says that the maximum grid bleeder must be not more than 200 kΩ. If it is a little lower, it won't do any harm. You should also consider adding P1 in parallel connection with R16 to R17. Depending on the position of P1, you can add about 30 kΩ to R15.

If an overall low-impedance design is used, the coupling capacitors also need to be dimensioned correspondingly large, otherwise there will be a premature drop at the low frequencies in the preamplifier. At that point, the targeted good frequency response would already be destroyed. The development objective is for the amplifier to transmit 20 Hz at maximum power without noteworthy distortions, for which the output transformer represents a major contribution.

The high-frequency oscillating tendency is suppressed via C3. If, according to the circuit design, there still is a persistent oscillating tendency, C1 can be used. If the frequency response drops toward high frequencies, you can compensate for it using C5, which is a film capacitor. But normally, you can do without C1 and C5. To avoid a high-frequency

Amplifier (Figure 3)

Resistors, unless otherwise specified, are metal film 0.7 W, 1% tolerance, MO = metal oxide, 2 W, 2% tolerance

Capacitors

C1	10 to 100 pF, ceramic (only in case of oscillating tendency, see text)
C2	47 pF, Ceramic
C3	10 pF, Ceramic
C4	10 µF/450 V
C5	1 to 10 nF, MKT (see text)
C6	0.22 µF/630 V, MKP
C7	47 µF/450 V
C8	470 µF/25 V
C9	0.68 µF/630 V
C10,C11	100 µF/160 V

Miscellaneous

Rel1	SIL relay, 12 V, 1 × ON

1 Tube Socket
Nine-pin socket print ceramics

Circuit board
Epoxy resin glass fiber reinforced, 70-µm copper cover, 82 mm × 130 mm , (Board number EE-343)

Potentiometer

P1	Trimpot, 25 kΩ, Cermet

Resistors

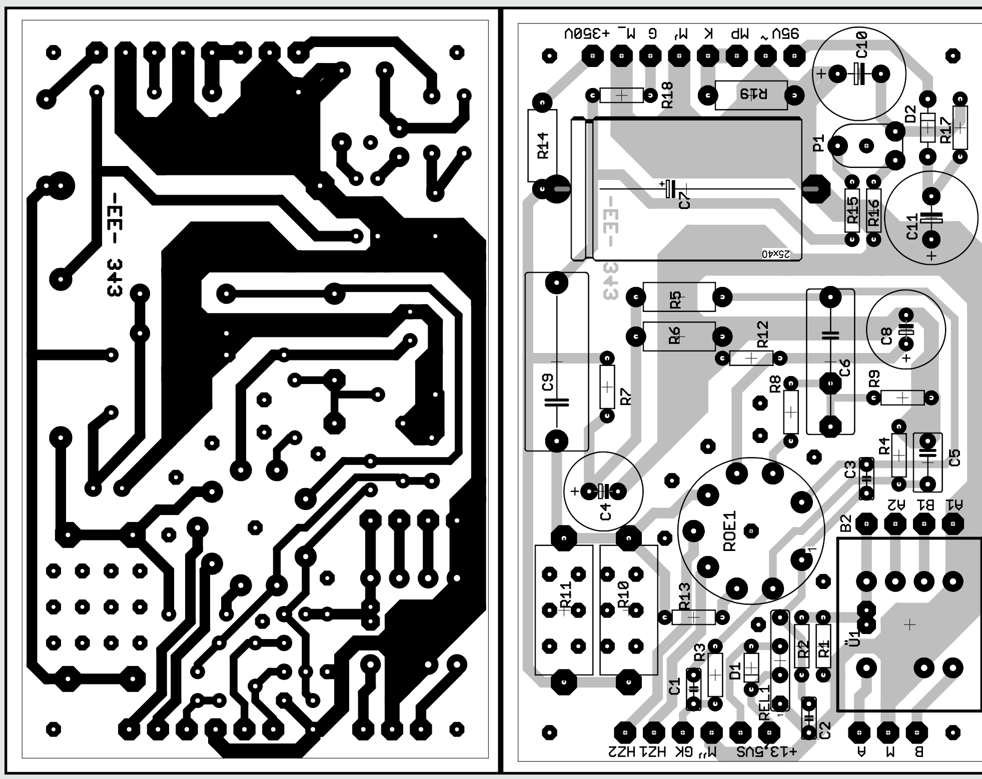
R1	33 kΩ
R2	4.7 kΩ
R3	16 kΩ
R4	1.5 kΩ
R5,R6	150 kΩ, MO
R7,R8	4.7 kΩ
R9	680 kΩ
R10,R11	15 kΩ, 4.5 W, 5%, MO
R12	150 Ω
R13	1 kΩ
R14	150 Ω, MO
R15	100 kΩ
R16	27 kΩ
R17	4.7 kΩ
R18	1 kΩ
R19	10 Ω, MO

Semiconductors

D1	1 N 4148
D2	1 N 4007
U1	E-1220 (see text)
U2	A-6C33 A (Ra = 600 Ω)

Vacuum Tubes

V1	PCL 86
V2	6C33 with socket



oscillating tendency, C3 is usually sufficient.

6C33's bias current is set to 200 mA using P1. To do so, the voltage drop at R19, measuring point (MP) against ground should be 2 V. When setting the bias current, it is important that the tube is heated up. The bias current slowly increases as the temperature increases. Due to the big cathodes and the tube's solid internal design, it can be set to the nominal value if it stays stable for at least 10 min.

Before operating the fully assembled amplifier, first allow it to heat up for 10 to 15 min. Prior to that, the bias current only achieves 75% of its nominal value. The best thing to do is to play back music because the loudspeaker's voice coils will also warm up. After the lead time, the full sound can be achieved.

Place Rel1 at the input because it enables you to mute the amplifier. The arrangement also leaves space for an input transformer. If you build monoblocks and longer wires between preamplifiers and power amplifiers are necessary, you can carry ground-free wirings. The transformer can be connected 1:1 or 1:2. In case of 1:2, a voltage gain of 6 dB is achieved. However, the input impedance drops to a quarter, which is about 8 kΩ.

For a linear frequency response, the transformer requires a real load of about 33 kΩ, for which R1 is used. When leaving out the transformer, a coupling capacitor should be used (see **Figure 4**). This also shows you how to connect a volume potentiometer. In this case, R1 should have about five to 10 times the value of the potentiometer resistance.

Depending on the preamplifier and the signal source, potentiometers between 10 kΩ and 100 kΩ can be used. If at all possible, you should opt for lower impedance, which is beneficial in terms of the SNR. The circuit board features the corresponding holes for soldering in the capacitor. In this case,

Power-Off Crackle Suppression (Figure 3)

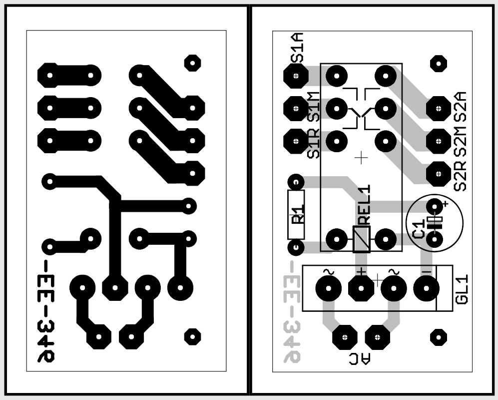
Bridge Rectifier
BR1 B 80 C 800 or similar

Capacitor
C1 10 µF/63 V

Miscellaneous
Rel1 12 V/40 mA, 2 ×UM gold-plated contacts

Circuit board epoxy resin glass fiber reinforced, 70-µm copper cover, 31 mm × 53 mm, (Board number EE-346)

Resistor
R1 150 Ω



use a film capacitor of at least 0.47 µF to avoid a premature drop at low frequencies.

The Power Supply

If you splurge on the amplifier circuit, don't attempt to save money on the power supply. I also recommend choosing a monoblock design. As confirmed in multiple listening tests, the extreme channel separation of mono power amplifiers offers plasticity, resolution, and spatialization during music playback, which is otherwise hard to achieve.

From a financial viewpoint, this is a one-time larger investment. However, if the amplifiers provide reliable service for many years, the initial expenditure can be put into perspective. Music can be enjoyed right from the start. As for the power supplies, they are designed to be easily built. **Figure 5** shows the circuit of the DC heating. **Figure 6** illustrates the 6C33's high voltage supply. **Figure 7** shows the driver stage's power supply.

Since the preamplifier and the power amplifier operate at different voltages—and because the power stage is operated at a lower voltage—two high voltage power supplies must be used. Due to the resulting power loss, it wouldn't make any sense to generate the lower operating voltage for the power stage from the higher operating voltage. In this way, more than 20 W can be converted into heat for each 6C33 tube.

Power Supply Design for Building Monoblocks—Parts List DC Heating (Figure 5)

Capacitors

C1	2 x 10,000 µ/25 V
C2	47 µ/40 V
C3	220 µ/40 V
C4	47 µ/40 V
C5	470 p, ceramic
C6	1000 µ/40 V
C7	1 n, ceramic

Diodes

D1	1 N 4007
D2	BZX 79 C 22
D3	ZY 18

Miscellaneous
BR1 Metal bridge, 10 A, 50 V (not on circuit board)
1 IC socket, DIL 14
1 heatsink, thermal resistance <4 K/W (provided that the housing is not used for cooling)
Circuit board 75 mm x 100 mm (Board number EE-212 A)

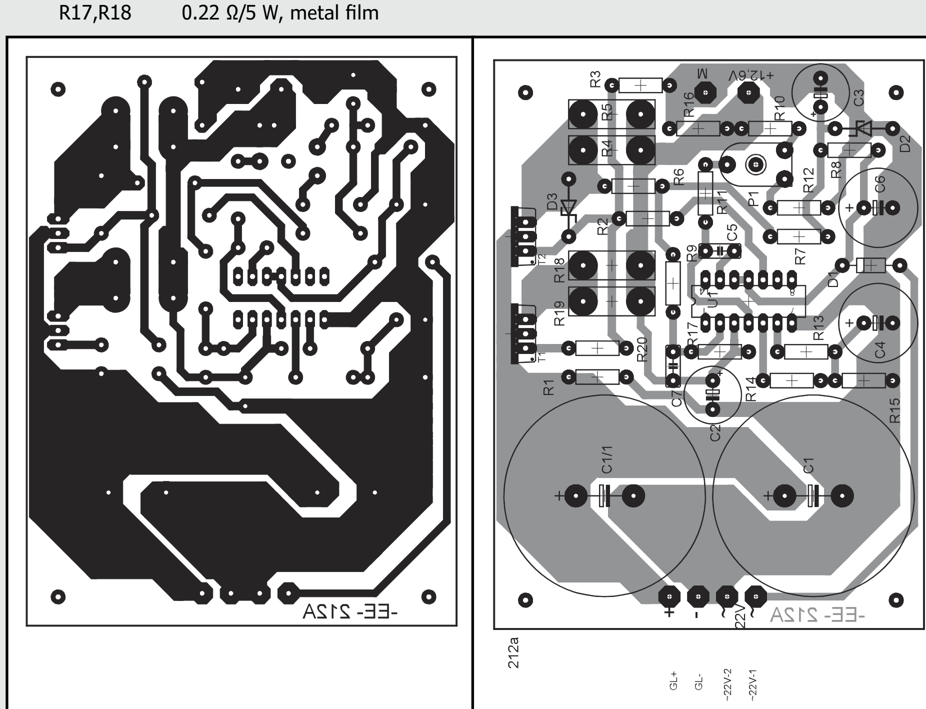
Potentiometer
P1 Trimpot, 1 kΩ, Cermet

Resistors

R1	4.7 kΩ
R2	100 Ω
R3	1 kΩ
R4,5	0.22 W/5 W, metal film
R6	47 Ω
R7	2.2 kΩ
R8	470 Ω
R9	1 kΩ
R10	4.7 kΩ
R11	1 kΩ
R12	2.7 kΩ
R13	2.2 kΩ
R14	1 kΩ
R15	4.7 kΩ
R16	1 kΩ
R17,R18	0.22 Ω/5 W, metal film

Transistors
T1,T2 BUZ 12

Voltage Regulator
IC1 723 DIL



Use two inductors and three electrolytic capacitors to filter the high voltage for the 6C33. Due to the lower current consumption, the high voltage for the PCL 86 can be comfortably generated using an active regulation circuit. Also the heating is regulated, stabilized, and features a soft-start function. Good filtering in the power supply combined with a clean design avoids hum. This is very important when operating the amplifier with the high-efficiency speakers for which it is designed.

The Driver Stage

The driver stage can also be used with many other power tubes (e.g., the EL 34, the 6L6, the KT 120, the KT 88, the KT 77, the 6550 and many others). Due to the higher amplification of the power pentodes, the negative feedback resistor R3 must be adjusted. If the open loop gain is too high, C8 can be omitted, which results in a reduction by about 6 dB. Consequently, the negative feedback doesn't need to be pulled so tight. It is sufficient if the amplifier can be fully driven with 1 V, which can be set via the negative feedback resistor R3.

It is essential to ensure that all the components are soldered on the component side. However, the tube socket must be soldered on the soldering side! By doing so, the circuit board can be installed so that the PCL 86 protrudes from the chassis. This design looks better and provides better cooling. To ensure better cooling, solder the two resistors R10 and R11 at a slight distance from the circuit board. They can also be fixed on the soldering side, which reduces the heating of the circuit board.

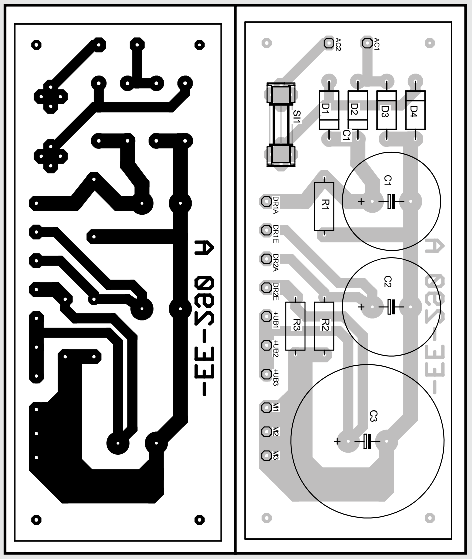
When Powering Off

Finally, when turning off the single-ended Class-A power amplifiers (especially if they are equipped with bigger tubes and feature more power), they can cause voltage jumps at the output because of the unbalanced supply for the power tubes. This could damage sensitive high-efficiency loudspeakers. Therefore, you should implement the power-off crackle suppression shown in **Figure 3**. It is marked with a dotted frame at the transformer output and is operated from the heating coil.

In idle state, the relay contacts are closed and they open shortly after switching the operating voltage on. After switching off, the relay immediately drops and short circuits the loudspeaker output.

I used a relay with two double-pole double-throw (DPDT) contacts so that only one power-off crackle suppression is required in a stereo amplifier. With mono power amplifiers, the second contact set can be connected in parallel with the first one, which increases contact reliability and improves the low impedance. The cables for the loudspeaker terminals should have a cross section of at least 0.5 mm², so the protection will work.

High-Voltage 6C33 (Figure 6)



Capacitors

C1,C2	220 µF/450 V
C3	680 µF/450 V

Miscellaneous

Fuse 0.63 A slow blow, with locking clips

Circuit board 55 mm × 192 mm, (Board number EE-290 A)

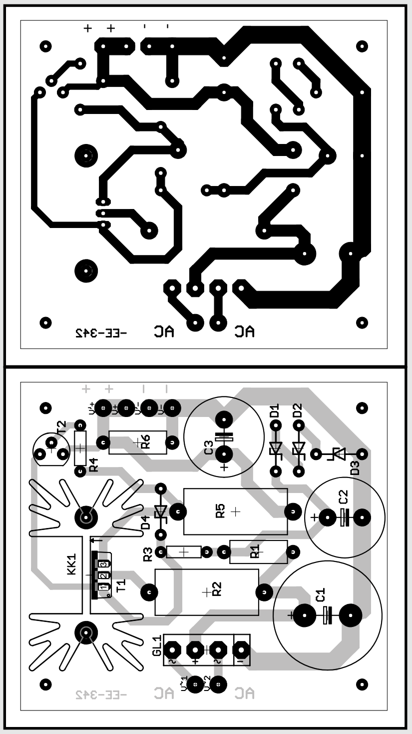
Resistors

R1,R2,R3	150 kΩ, MO

Semiconductors

D1–4	BY 299
Dr1, Dr2	D-2360, 2,3 H, 0.3 A

High-Voltage Driver Stage (Figure 7)



Capacitors

C1	47 µF/450 V
C2,C3	22 µF/450 V

Miscellaneous

GR1	2KBB100
1	Heatsink SK 104 with mica disk TO 220 and insulating nipple

Circuit board 72 mm × 81 mm, (Board number –EE-342)

Resistors

R1	22 kΩ, MO
R2	100 Ω, 4.5 W, MO
R3,R4	5.6 kΩ
R5	15 Ω, MO
R6	150 kΩ, MO

Semiconductors

D1	ZY 110
D2,D3	ZY 120
D4	ZY 18

Transistors

T1	BUZ 80 A (mounted with insulation)
T2	BC 546 B

Mains Transformer Monoblock

NTR-25

280 V/50 mA, 180 V/0.35 A, 95 V/50 mA, 22 V/50 mA, 14 V/5 A, core MD 85 B

Output transformer	A-6C33 A Ra = 600 Ω secondary 4, 6, and 8 Ω
Filter choke D-2360	2.3 H/0.3 A, RDC <30 Ω

The Right Filter (Part 2)

Crossover Frequency Tuning

My goal is to build a tunable near linear phase analog crossover, using analog components. Last month, I explained the method I used to get phase linearity over the audio band, without too much detail. This month, I will describe the hardware and briefly justify my design choices, focusing on phasor cells, tuning boards, and phase correctors dedicated to the crossover frequency tuning capability.

Photo 1: The two boards used in this tunable near-linear phase analog crossover project are shown.

By
Vincent Thiernesse
(France)

The main board in my design is composed of two parts because a single board would be too large. The two boards can be assembled together (see **Photo 1**). However, there is only be one board per channel. One of these two includes a management system performed by an Arduino micro module. It also includes human/machine interface via a digital scroll/button or IR remote. This system can be put in standby mode via a relay.

And, it contains the following slots:

- 5 slots for the phasor cells circuit boards
- 1 slot for the phase corrector
- 1 slot for the frequency board:
  - Fixed crossover frequency board offering higher quality and voltage swing
  - Tunable crossover frequency board for system engineering
  - An additional slot gives you access to pseudo-samples during the realization process.

You can also see the string of 20 1% weighting resistors.

The rest of the circuit addresses level adaptation. One relay per channel (K_2 and K_4) avoids powering artifacts. One relay per channel (K_1 and K_3) allows switching between the two impulse response configurations (i.e., filter 1, lower octaves, and filter 2, upper octaves). This switching is available with a fixed crossover frequency board as well as with the tunable crossover frequency board.

The parts list, schematics, and PCBs needed to build the main board, the digital rotary encoder, and the phasors can be found in the Supplementary Materials section on the *audioXpress* website (www.audioxpress.com).

I do not describe a power supply for this project. However, any power supply you select must deliver ±15 V/200 mA and 5 V/300 mA via a 5-pin DIN connector. You can, of course, use an ad-hoc design with other connectors. An external power supply configuration would give you a high 50-to-60-Hz rejection.

A digital scroller with a push-button includes hardware event management to lighten the load for the controller and avoid missed events due to timing

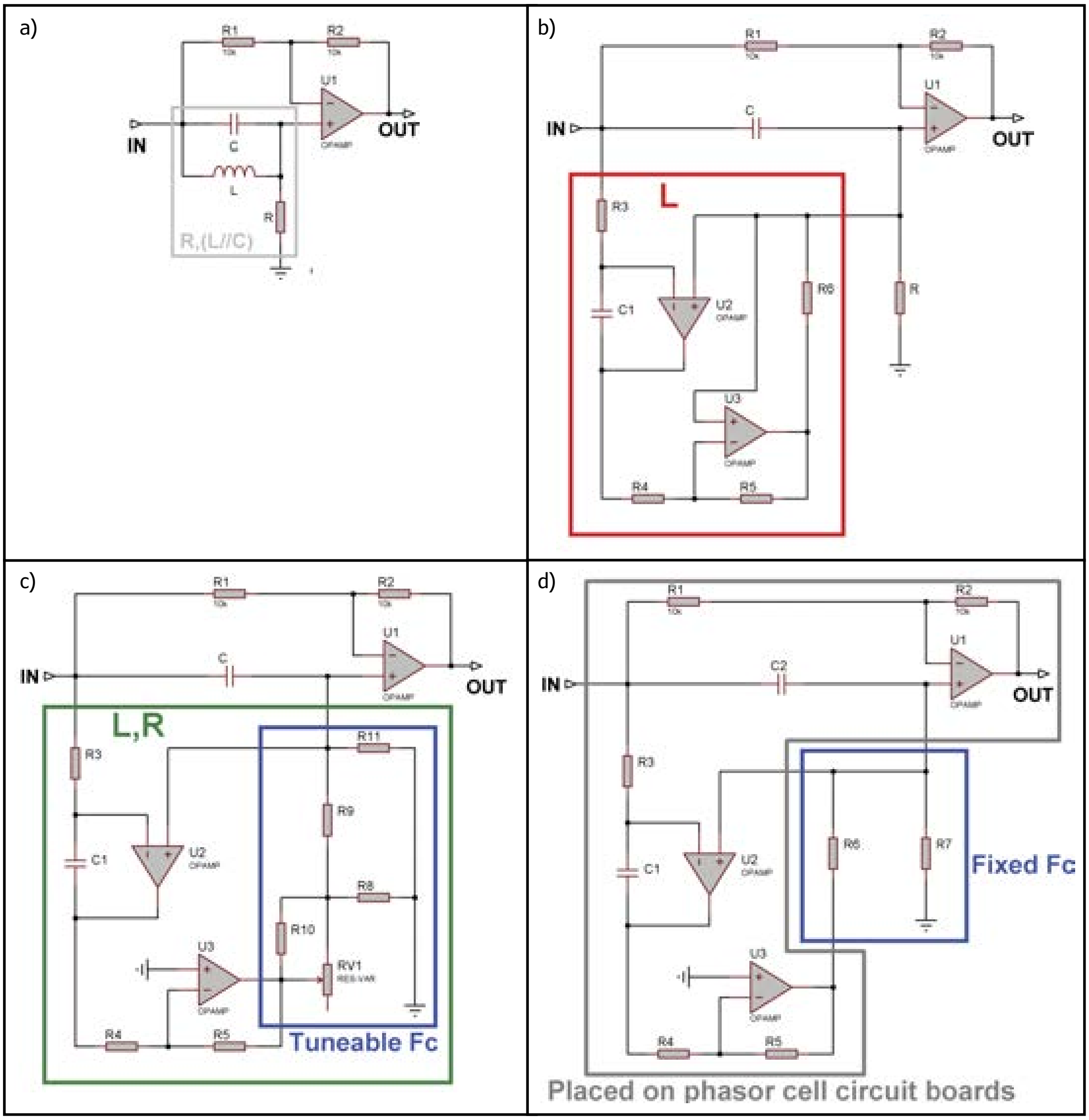


Figure 1: I have provided four tunable phasor cell topologies. The parts highlighted in blue are shown on the respective frequency circuit boards. Figure 1d's configuration is a consequence of Figure 1c's topology. For my design C1 = C.

constraints and enables "smart" physical integration of scrolling capability.

You might want to use high-quality capacitors and relays on that board. If you want to modify the main board, the Proteus files are available in the Supplementary Materials section of the *audioXpress* website.

Tuning the Frequency

There are at least three ways to design all-pass phasors using one only op-amp (see **Figure 1**). The challenge was to change the center frequency of each phasor cell with a single potentiometer per phasor with low damping factor variations.

I chose the design shown in **Figure 1a** because it was clear enough to focus on problems other than the circuit itself, and because it has the same topology as the 180° phasors used in my earlier projects.

The R,(L//C) circuit shown in **Figure 1a** is a notch filter. Subtracting the signal from this notch filter with a gain of two gives the the desired transmittance: a 0 to -2π phasing all-pass with damping factor being the one of employed R,(L//C) circuit.

The notch filter transmittance equation related to **Figure 1a** is:

$$\underline{B}(j\omega)=\frac{1-\left(\frac{\omega}{\omega_0}\right)^2}{1+2j\times m\times\frac{\omega}{\omega_0}-\left(\frac{\omega}{\omega_0}\right)^2}$$

with:

$$\begin{cases}\omega_0=\dfrac{1}{\sqrt{LC}}\\ m=\dfrac{1}{2R}\times\sqrt{\dfrac{L}{C}}\end{cases}$$

=> Corrector 1 and phasors cells isochronous transmittance:

$$\underline{T}(j\omega)\ =\ 2\times\underline{B}(j\omega)\ -\ 1\ =\frac{1-2j\times m\times\frac{\omega}{\omega_0}-\left(\frac{\omega}{\omega_0}\right)^2}{1+2j\times m\times\frac{\omega}{\omega_0}-\left(\frac{\omega}{\omega_0}\right)^2}$$

with:

$$\begin{cases} \omega_0 = \dfrac{1}{\sqrt{LC}} \\ m = \dfrac{1}{2R} \times \sqrt{\dfrac{L}{C}} \end{cases}$$

imposing:

$$\begin{cases} C = \dfrac{1}{2mR\omega_0} \\ L = \dfrac{2mR}{\omega_0} \end{cases}$$

The circuit shown in **Figure 1a** uses an undesirable inductor. It is replaced in the circuit shown in **Figure 1b** by a one-side active inductor. The introduction of two additional op-amps shouldn't create much noise and distortion as U2 works as an integrator.

Now we have: $L = R_3 R_6 C_1$ if $R_4 = R_5$

So,

$$m = \frac{1}{2R} \times \sqrt{R_3 R_6} \quad \text{and} \quad \omega_0 = \frac{1}{\sqrt{R_3 R_6} \times C}$$

Then,

$$\begin{cases} R = \dfrac{1}{24 \times \pi \times m \times C} \times \left(\dfrac{\alpha(m)}{f_{cross}} \right) \\ R_6 = \dfrac{1}{144 \times \pi^2 \times R_3 \times C^2} \times \left(\dfrac{\alpha(m)}{f_{cross}} \right)^2 \end{cases}$$

The first article in this series, "The Right Filter," (*audioXpress*, June 2015) details the significance of $\alpha(m)$. Note that $\alpha(m)$ also depends on the number of pseudo-samples per cardinal sine wave used for impulse response synthesis.

$$\alpha(m,N) = \frac{m \times N}{\tan\left(\dfrac{\pi}{2N}\right)} \times \left(\sqrt{1 + \frac{\tan\left(\dfrac{\pi}{2N}\right)}{m}} - 1 \right)$$

Where N is the number of pseudo-sample of half wave of the cardinal sine being impulse response of perfect low-pass filter. That gives:

$$\begin{cases} R = \dfrac{1.4365 \times 10^7}{f_{cross}} \\ R_6 = \dfrac{1.4055 \times 10^{11}}{f_{cross}^2} \end{cases}$$

with chosen components and:

$$R_7 = \frac{1}{1/R - 2/R_6} = \frac{1}{6.961 \times 10^{-8} \times f_{cross} - 1.423 \times 10^{-12} \times f_{cross}^2}$$

In case of a fixed crossover-frequency configuration, you have:

$$\begin{cases} R_{21} - R_{30}, R_{32}, R_{33} = R_6 \\ R_1 - R_{10}, R_{12}, R_{13} = R_7 \end{cases}$$

on a fixed crossover-frequency circuit board where R41-R50, R52, and R53 can be used for adjustments.

For example, a fixed 3-kHz crossover frequency =>

$$\begin{cases} R_6 \approx 15.6 \quad k\Omega \\ R_7 \approx 12.4 \quad k\Omega \end{cases}$$

The frequency board described in the first article in this series uses similar values. You need 48 identical capacitors. I bought a set of 50 5% capacitors. This is rather inexpensive and you can disregard less precise values.

The 5% precision of the capacitors relates more to the mean value than the standard deviation. Thus, it is better to get all the capacitors from the same

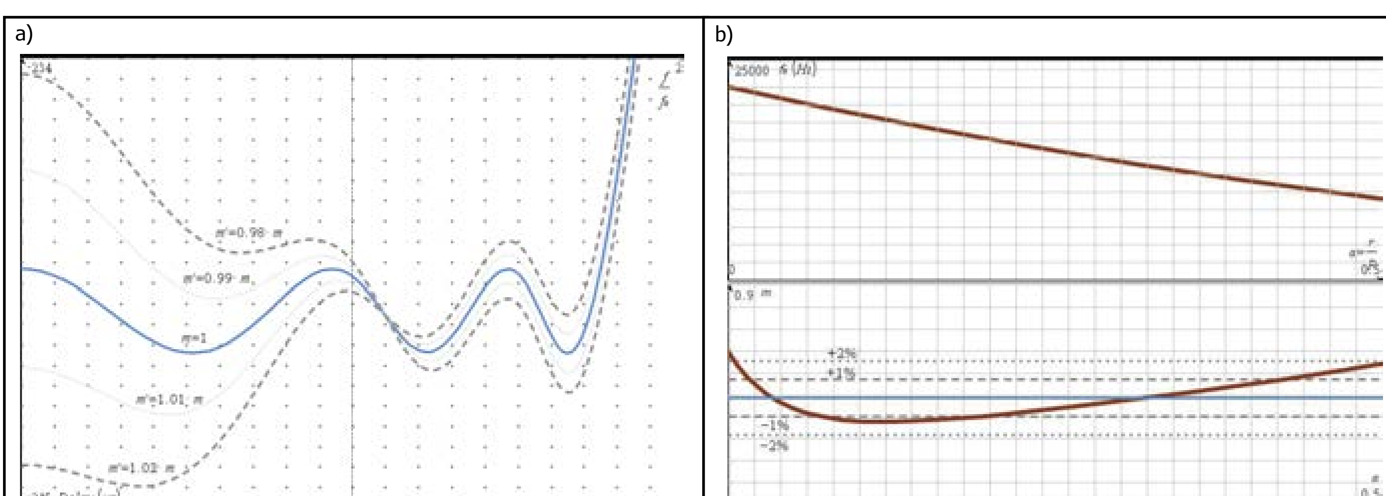


Figure 2a: The time delay flatness sensitivity is compared to damping factor variations (f_0 = 12 kHz); b: The center-frequency and damping factor variations (x = RV1/5 × 104) are shown.

manufacturing set to get homogeneous damping factors for the 10 phasor cells.

If we tune the crossover frequency, then we have to adjust R6. Resistor R, shown in **Figure 1c**, has to be tuned at the same time to avoid m variations.

Then we need:

$$\frac{\sqrt{R_6}}{R}$$

to remain constant regardless of f_0 tuning. In **Figure 2a**, the consequences of m variations on time delay flatness is a 2% damping factor deviation => 7.5 µs time-delay ripple.

Quantity	Parts References	Value
Integrated Circuits		
12	U1-U10, U12, U13	MCP4011-503E/SN (50 kΩ/SOIC)
Resistors		
1	R0	1 kΩ
12	R1-R10, R101, R102	12 kΩ
12	R11-R20, R111, R112	15 kΩ
10	R21-R30	56 kΩ
2	R121, R122	100 kΩ

Table 1: This is the parts list for the crossover-frequency tuning board shown in Figure 3.

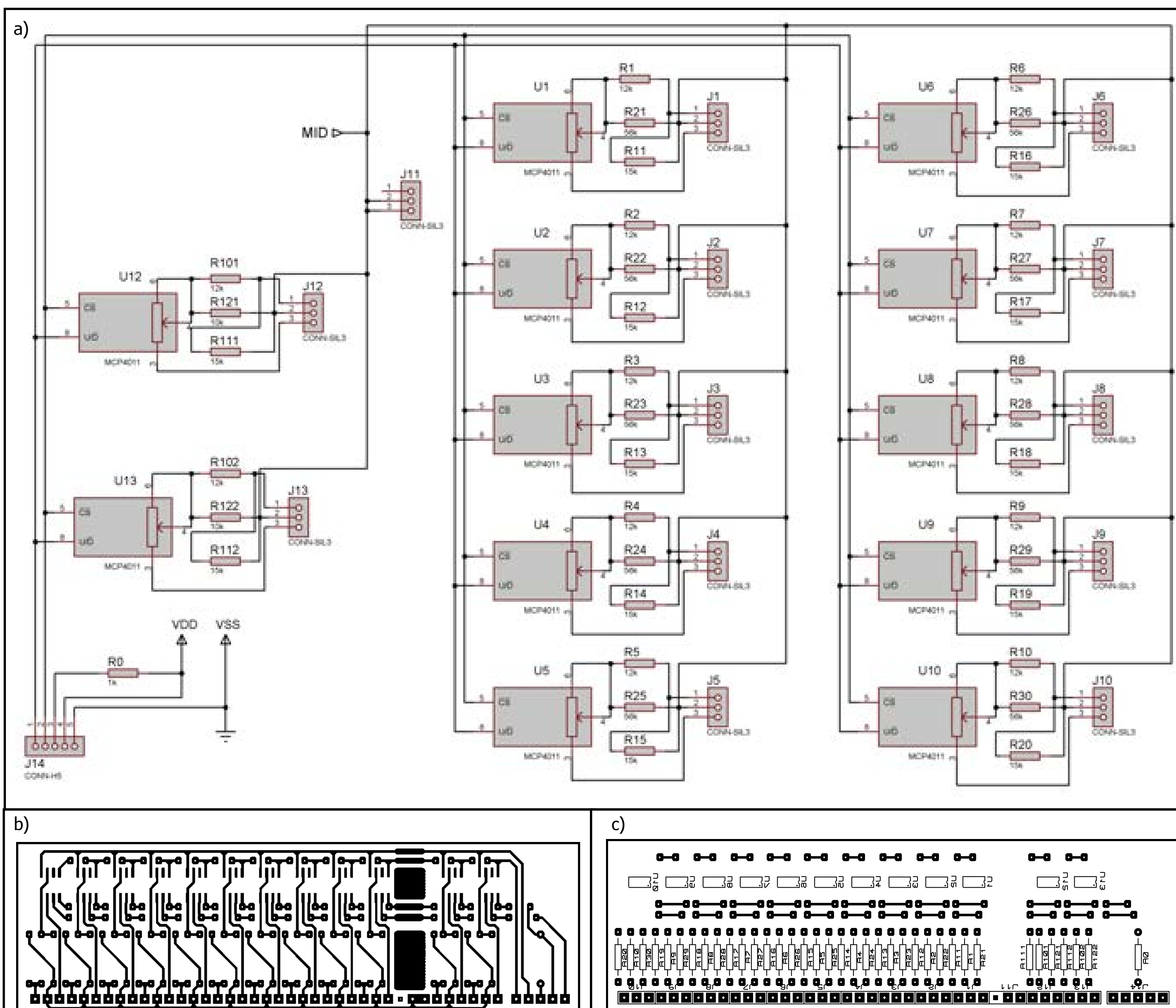


Figure 3: The schematic (a) and the PCB's front (b) and back (c) for the crossover-frequency tuning board is × 2 for the stereo filter. The dimensions are 4.625″ × 1.375″.

Table 2: This is the parts list for the fixed frequency board shown in Figure 4.

Quantity	Parts References	Value
Resistors		
12	R1-R10, R12, R13	12.7 kΩ
12	R21-R30, R32, R33	15 kΩ
12	R41-R50, R52, R53	180 Ω
Values for 3-kHz crossover frequency		
1	R60	1 kΩ

In **Figure 1b**'s circuit, using only one potentiometer to tune both R6 and R at the same time would result in too narrow acceptable crossover frequency tuning. So, the circuit shown in **Figure 1c** is an evolution of it, enabling wider crossover frequency tuning—more than an octave in practice. The circuit shown in **Figure 1d** is an adaptation of the design to enable the use of both a fixed crossover frequency board (detailed in the first article in this series) and the tunable crossover frequency board discussed in this article. Let us assume that a ±1.5 % variation is the maximum variation you can accept. **Figure 2b** shows the results I obtained.

Note that a high-pass response to a square signal is a visual indicator of phase linearity. You may verify that the two overshoots won't mismatch more than 20% over the crossover frequency range.

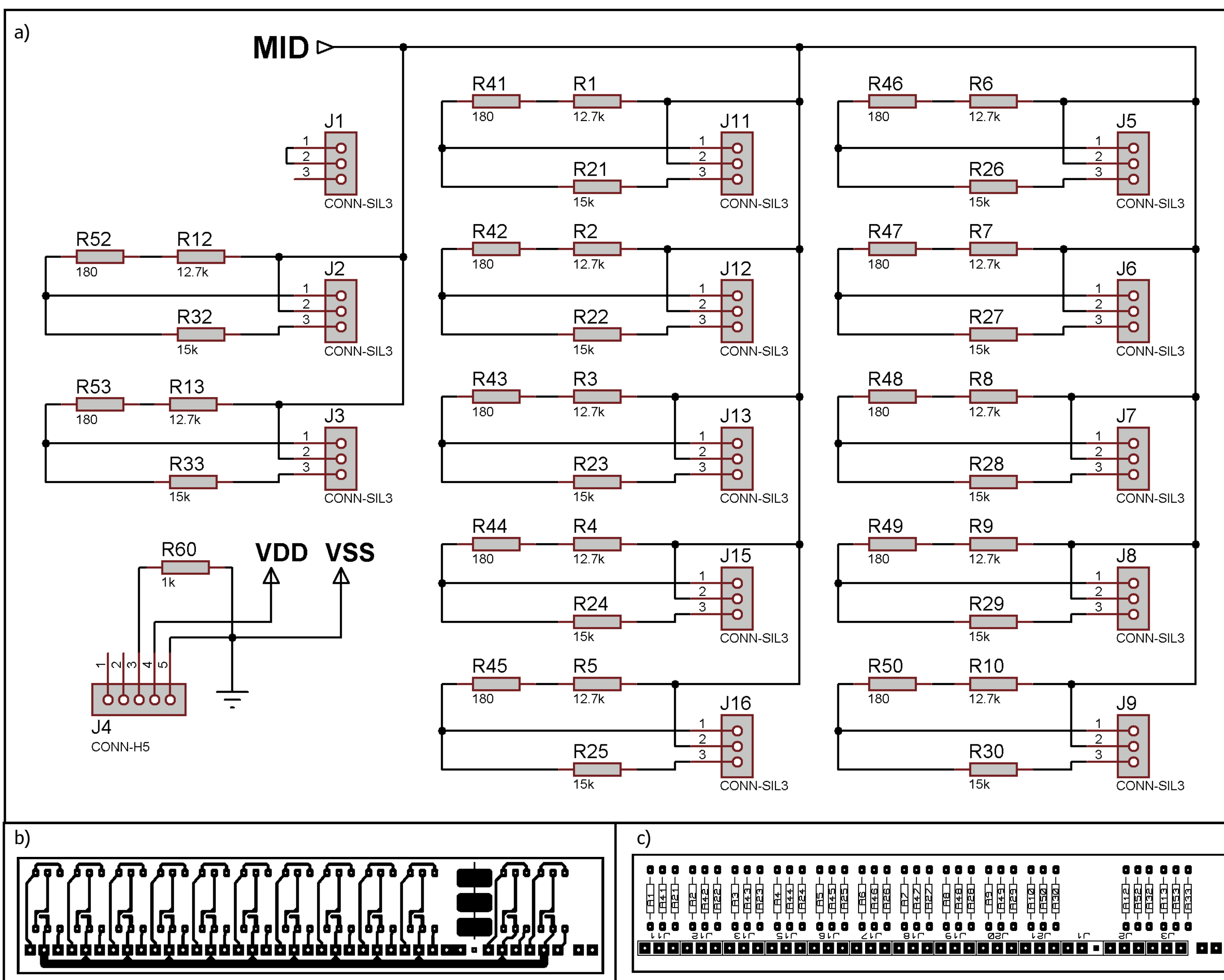


Figure 4: The schematic (a) and the PCB's front (b) and back (c) for the fixed frequency board is × 2 for the stereo filter. The dimensions are 4.21" × 0.81".

The crossover frequency tuning circuit board shown in **Figure 3** integrates 12 6-bit digital potentiometers: 10 for 10 phasor cells and two for phase corrector 1. I chose MCP4011/50 kΩ with a small outline integrated circuit (SOIC) package because of its size, low cost, and acceptable specifications. The SOIC package is also big enough for nondestructive 15-W iron soldering. **Table 1** shows the parts list for the crossover frequency tuning board.

These digitally tuned potentiometers have their input/output voltage swing limited by 0/5 V power

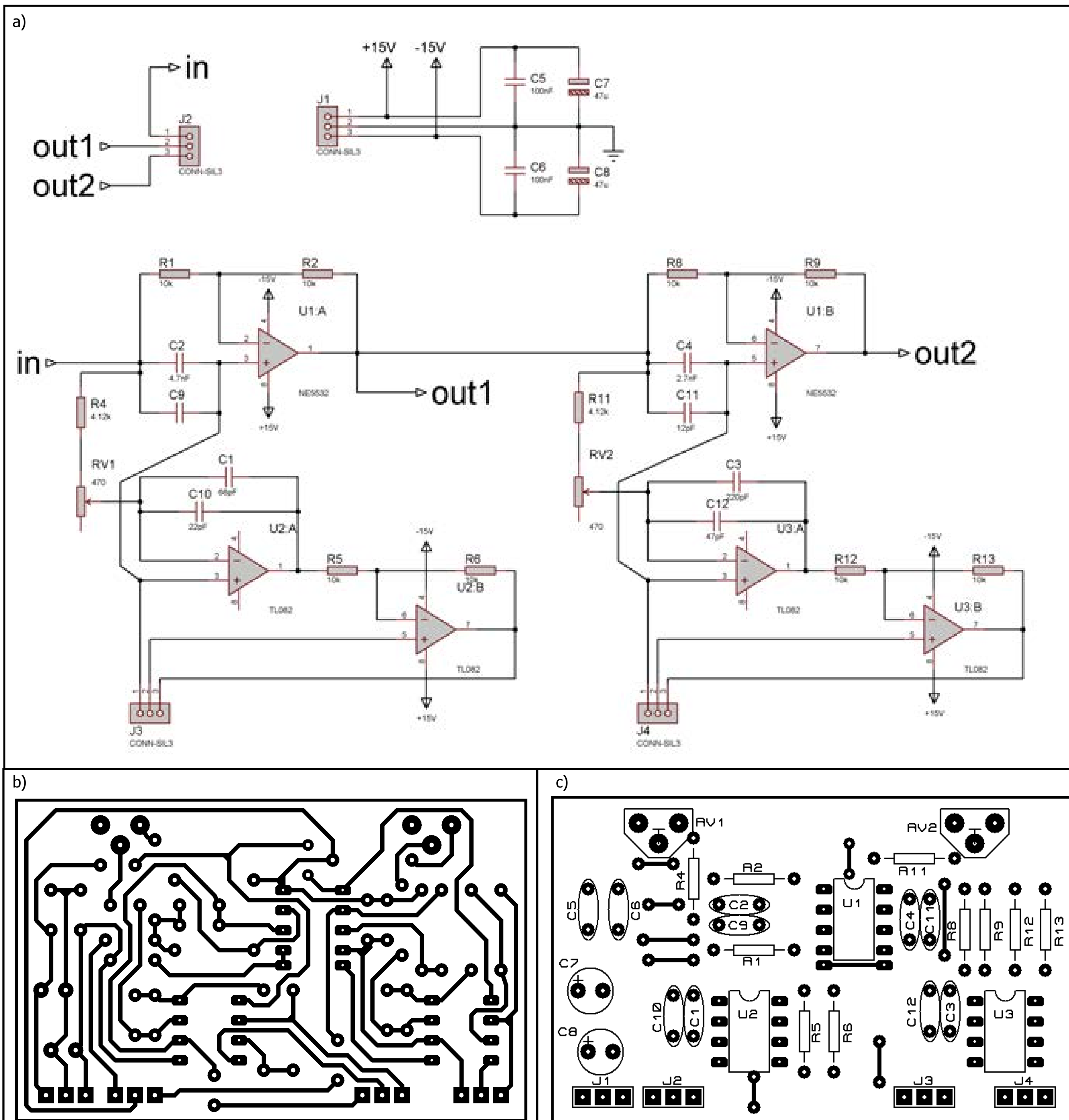


Figure 5: The phase corrector 1's schematic (a) and the PCB's front (b) and back (c) for second circuit board is × 2 for the stereo filter. The dimensions are 2.575″ × 1.6″.

Table 3: This is the parts list for the phase corrector 1.

Quantity	Parts References	Value
Capacitors		
1	C1	68 pF
1	C2	4.7 nF
1	C3	220 pF
1	C4	2.7 nF
2	C5, C6	100 nF
2	C7, C8	47 uF/25 V
1	C10	22 pF
1	C11	12 pF
1	C12	47 pF
Integrated Circuits		
1	U1	NE5532
2	U2, U3	TL082
Miscellaneous		
2	RV1, RV2	470 Ω
Resistors 1%		
8	R1, R2, R5, R6, R8, R9, R12, R13	10 kΩ
2	R4, R11	4.12 kΩ

Project Files

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

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.

rails. So, the virtual ground is set to 2.5 V provided by the motherboard. The selection of this virtual ground is done with jumper J11 on the circuit board (see **Figure 3**).

Figure 4 shows the fixed frequency circuit board. **Table 2** shows the parts list for the fixed frequency board.

Phase Corrector 1

Phase corrector 1 , which is the circuit board shown in **Figure 5**, is almost the same as the phasor cell's circuit. **Table 3** provides the parts list for phase corrector 1. Only the capacitors values differ. **Table 4** details how to change the capacitors values.

The central frequency of this phase corrector is tuned the same way as the 10 phasors cells. The damping factor of the second phasor cell is low: 0.1384. In some cases, that could cause uncontrolled oscillations because of stability issues. So, we have to make sure that such a low dumping factor won't cause uncontrolled oscillations.

Time Domain Simulations

Table 5 shows the results of the time-domain simulations of this filter with and without the phase corrector. I will discuss the phase corrector 2 next month. You may prefer it, if your audio source is not 20-kHz limited, as CD players are. Next, I will try to explain why the transient response may have undesired ripples, even if those ripples may not be directly audible, they may cause amplifiers and transducers to exhibit unexpected behaviors. So, I will try to provide a few solutions.

With the simulations shown in **Table 5**, we see that phase corrector 1 will provide a satisfying transient response as soon as the input signal is 20-kHz limited.

The simulations in **Table 6** show that phase distortion consequences, over the audio band, may not be that important for most common applications. But, they may be more important if you seek perfection.

Relation Between Phase Rotations and Step-Response Ripples

This short study intends to show the relations between phase distortion, with respect to phase linearity, and step response ripples, specific for unity gain systems. These relations could be summed up as follows:

- You get N ripples before the output signal step when total phase rotation is $N \times 2\pi$, such as frequencies distribution is weighed by

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

- Ripples are a kind of burst whose frequencies cover the non-linear phase band.
- Ripples are as small as nonlinear transition is narrow so,

$$\left|\frac{\partial T_{delay}}{\partial \omega}\right|$$

is important. I will provide mathematical examples in the third article in this series.

Calculations for Corrector 1's Capacitor Values	Phasor Capacitors		Parameter Ratios		Capacitor Ratios		Corrector Capacitors	
	C	CL	$\frac{m'}{m}=\sqrt{\frac{l}{c}}$	$\frac{f'}{f}=\frac{1}{\sqrt{lc}}$	$\frac{L'}{L}=l=\frac{m' \times f}{m \times f'}$	$\frac{C'}{C}=c=\frac{m \times f}{m' \times f'}$	CL′	C′
Cell 1	1 nF	1 nF	0.3086	1.1488	0.2686	2.82	267 pF	2.82 nF
Cell 2			0.1384	1.540	0.08985	4.693	90 pF	4.7 nF

Table 4: Capacitor calculations are needed for phase corrector 1.

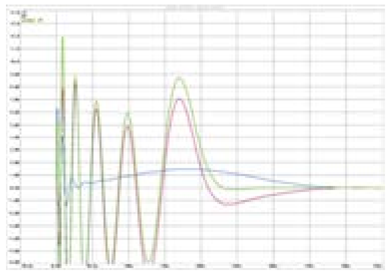
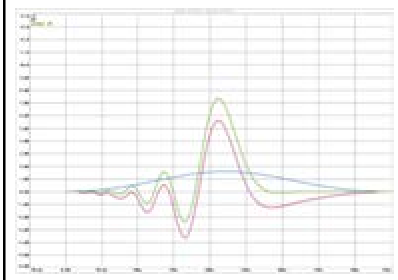
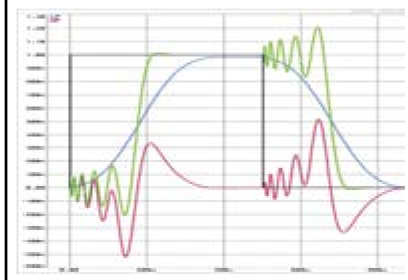
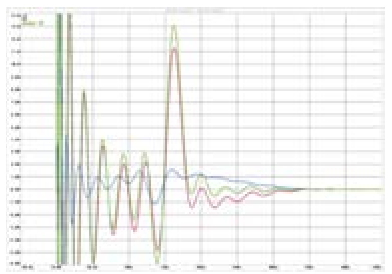
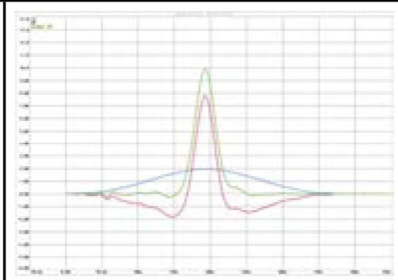
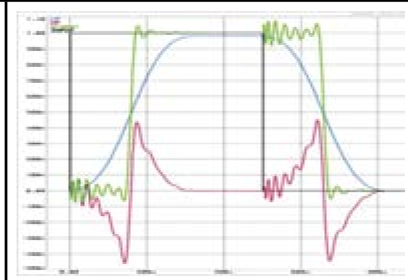
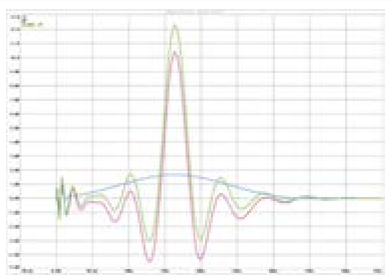
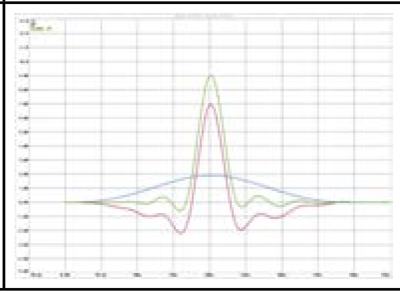
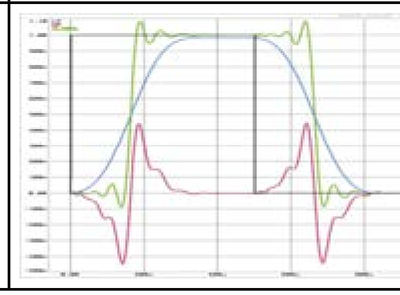
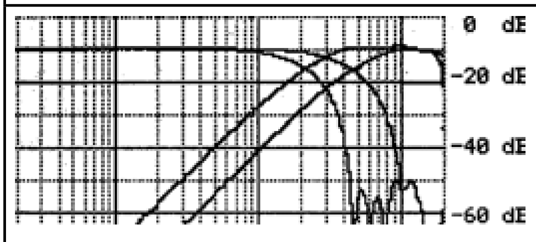
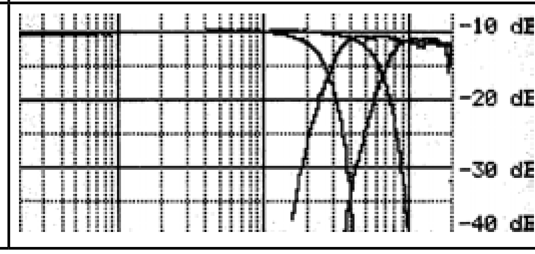
PROTEUS simulations	Full spectrum impulse	Gaussian impulse	Step response (20 µs rise-time)
Without a phase corrector			
Phase Corrector 1 (tunable)			
Phase Corrector 2 (fixed)			

Table 5: The full-spectrum impulses, Gaussian impulses, and step responses are shown.

Table 6: The tunable crossover frequency ranges from 1.5 to 3.5 kHz with filter 1 (a) and from 3 to 7 kHz with filter 2 (b). The phase linearity band is less than 20 kHz if f_C is less than 1.9 respectively 3.6 kHz.

Tunable Crossover Frequency	
a) Filter 1	b) Filter2
	

Assuming that a filter is linear in phase only until 20 or 40 kHz, these ripples won't happen if the input signal has no harmonics over 20 or 40 kHz. So, we need to find a solution to that problem.

One solution is to add a 20-kHz low-pass filter, possibly based on the same method. Too high a selectivity filter would create Gibbs overshoots. So, "in the Gaussian filter, we trust." To reach that goal, we need four phasors cells, with no phase corrector needed. The phase will be linear until about 0.6 × f_0 though frequency cutoff will be f_{cut} = $f_0/3$. Assuming that a Gaussian filter provides acceptable attenuation at 2 × f_{cut}, the problem is solved. Unfortunately, the complexity and the number of op-amps increase.

Increasing the corrector's complexity results in wider linear phase band and narrower nonlinear phase transition band. In fact, the more complex the phase corrector, the more abrupt the time delay breakdown is. This fact results in smaller unwanted ripples and these ripples are placed at higher frequencies. In the end, a simple passive low pass could filter these ripples. Unfortunately, such a corrector would require lower and lower damping factors for the phasor cells. The two consequences are the frequency response could overshoot on U2b, U3b, etc., which could limit the distortion-less output swing. Nevertheless, m as low as 0.1 still gives unconditional stability in practice. Perhaps I should have tried a three-cell corrector, but I have another solution.

Phase/Filter Corrector 2

I've been looking for a solution that would not increase the project's cost or complexity. The result is a structure that simultaneously performs phase correction and 20-kHz low-passing. This phase

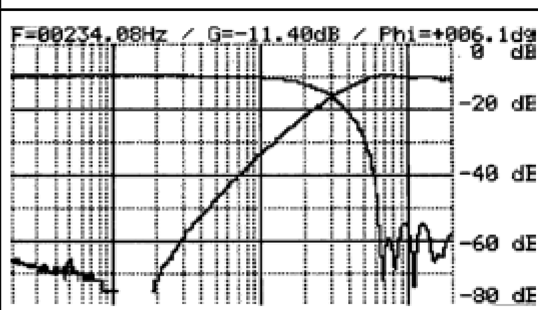
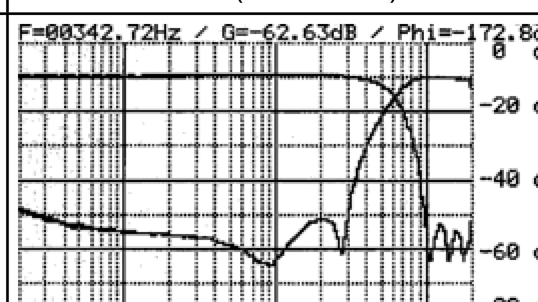
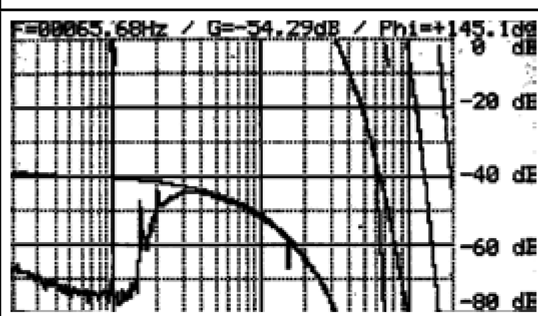
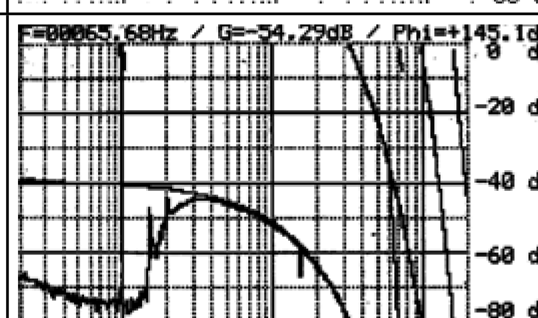
Fixed Crossover Frequency		
	Filter 1 (3-kHz Crossover)	Filter 2 (6-kHz Crossover)
Frequency response		
Phase response: HP + LP		

Table 7: The frequency response and phase response (high pass + low pass) are shown at a 3-kHz crossover for filter 1 and at a 6-kHz crossover frequency for filter 2. The domain measurements were taken using a homemade Bode plotter.

corrector will be described in the final article in this series.

Performances of this phase corrector appear in **Tables 5–8**. The actual crossover frequency is 2.5 kHz and not 3 kHz (as stated in the June *audioXpress* article). The filter's transient response won't be affected by unwanted ripples with this second phase corrector, but the crossover frequency won't be tunable until I, or one of our readers, finds a reliable solution.

Measurements

Measurements are very close to theory without anything to adjust. All trimmers can be placed in middle position. RV1, 101 for filter 1 and RV2, 102 for filter 2 can be adjusted applying a 600-Hz square wave. Look at the high-pass signal and then adjust them in both octave configurations to get equal DC levels.

Time domain measurements are so close to theory you could not recognize what is what…And time delay flatness conforms to specification (see **Table 7** and **Table 8**). There is nothing else to say about time-domain measurements except that they perfectly match previous simulations.

Conclusion

Next month, I will describe phase corrector 2, which simultaneously performs phase correction and 20-kHz anti-phase distortion filtering. I will also provide precise measurements of noise and distortion. The noise DSP is homogeneously spread over the audio band with an RMS value less than 40 µV. The total harmonic distortion (THD) is less than 0.01% but I need more precise measurements to provide the exact value.

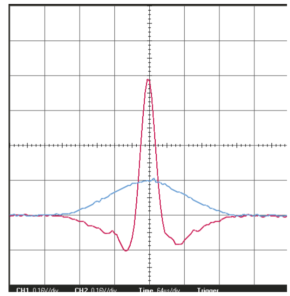
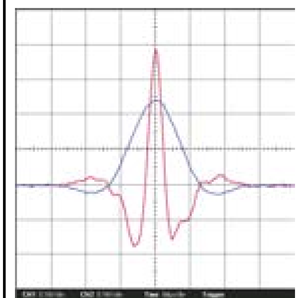
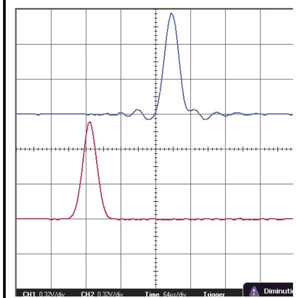
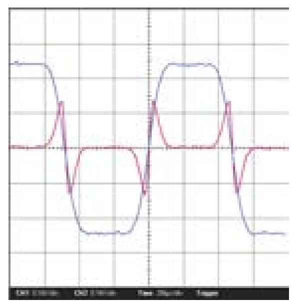
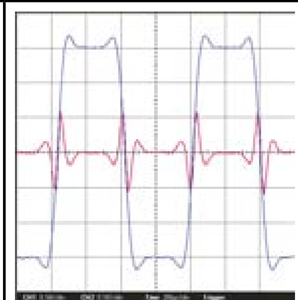
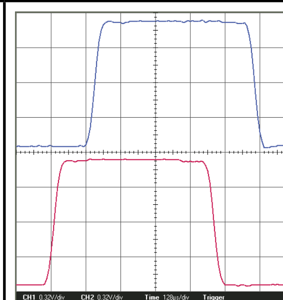
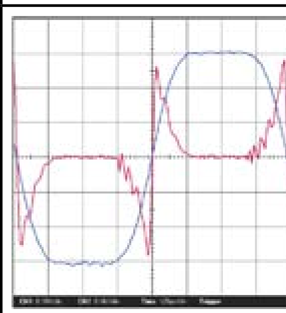
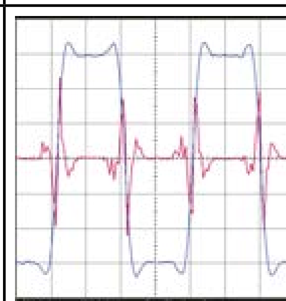
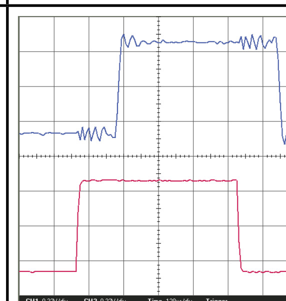
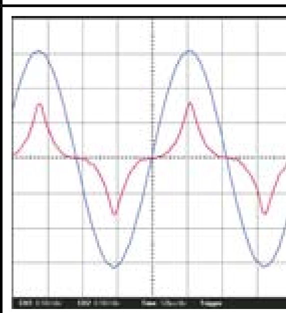
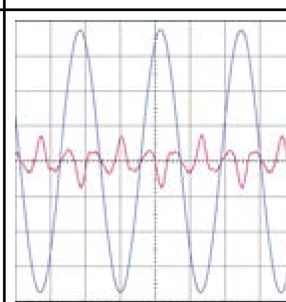
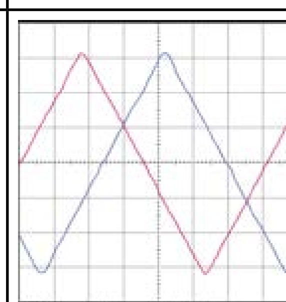
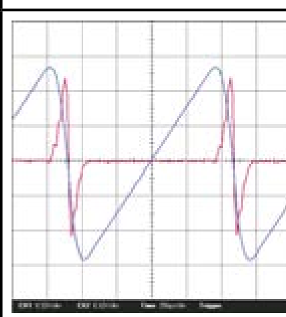
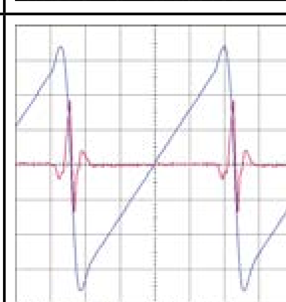
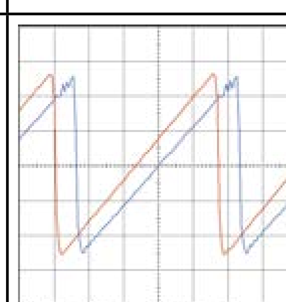
3-kHz Crossover Frequency	Filter 1	Filter 2	LP + HP (common for both filters)
Input response			
Square response (square convoluted with Gaussian)			
Square response			
Triangle response			
Saw response			

Table 8: The time domain responses, including the low pass + high pass, are shown for filter 1 and filter 2. The red trace shows the input and the blue trace shows the output.

Selecting the Operating Point for a Hollow-State Amplifier

In this article, we will see how to create a design for a hollow-state amplifier stage that will take into account the necessary factors. First, we will look at a triode circuit design. Then, we'll examine a pentode design.

By
Richard Honeycutt
(United States)

The first step in designing a hollow-state amplifier stage is selecting the device: a dual triode (e.g., a 12AX7/7025/ECC83), a pentode (e.g., a 6AU6A), and so forth. The second step is setting the operating point, or Q point, which is the combination of plate voltage, plate current, and control-grid voltage (and screen-grid voltage, for a pentode) at which the device will idle. There are four aspects involved in setting the Q point: maximum signal-voltage output, voltage gain, distortion, and plate power dissipation.

Creating the Design

The easiest way to create a design is to lift one from the tables included in a tube manual. In addition to one or two example circuits included in the datasheet for the triode or pentode, the *RCA Receiving Tube Manual* includes a set of design tables at the back.

Table 1 shows designs for the 3AV6, the 4AV6, the 6AV6, the 6EU7, the 12AV6, the 12AX7, the ECC83, the 20EZ7, and the 7025. **Figure 1** shows the circuit to which the design data applies.

The table headings represent components shown in **Figure 1**, plus voltage gain (V.G.). Ebb is the B+ voltage. The output voltage is the peak signal voltage at the plate with an input voltage just barely large enough to cause grid-current limiting. The cathode resistor value is given in ohms; and the plate resistor, in megohms. Capacitor values are in microfarads (μF), and are chosen for a low cut-off frequency of about 100 Hz. A low-impedance signal source and a very-high-impedance load (R_L>>R_P) are assumed.

Ebb	Rp (kilohms)	Rg (kilohms)	Rk (ohms)	Ck (uF)	C (uF)	Eo (peak volts)	Voltage Gain (V. G.)
90	100	100	4,400	2.7	0.023	5	29
90	100	220	4,700	2.47	0.013	6	35
90	100	470	4,800	2.3	0.007	8	41
90	220	220	7,000	1.6	0.012	6	39
90	220	470	7,400	1.4	0.006	9	45
90	220	1,000	7,600	1.3	0.003	11	48
90	470	470	12,000	0.9	0.006	9	48
90	470	1,000	13,000	0.8	0.003	11	52
90	470	2,200	14,000	0.7	0.002	13	55
180	100	100	1,800	4	0.025	18	40
180	100	220	2,000	3.5	0.013	25	47
180	100	470	2,200	3.1	0.006	32	52
180	220	220	3,000	2.4	0.012	24	53
180	220	470	3,500	2.1	0.006	34	59
180	220	1,000	3,900	1.8	0.003	39	63
180	470	470	5,800	1.3	0.006	30	62
180	470	1,000	6,700	1.1	0.003	39	66
180	470	2,200	7,400	1	0.002	45	68
300	100	100	1,300	4.6	0.027	43	45
300	100	220	1,500	4	0.013	57	52
300	100	470	1,700	3.5	0.006	66	57
300	220	220	2,200	3	0.013	54	59
300	220	470	2,800	2.3	0.006	69	65
300	220	1,000	3,100	2.1	0.003	79	68
300	470	470	4,300	1.6	0.006	62	69
300	470	1,000	5,200	1.3	0.003	77	73
300	470	2,200	5,900	1.1	0.002	92	75

Table 1: Tube manuals often included tables to simplify the design process. (This is Table A from the 1975 *RCA Receiving Tube Manual.*)

The left-hand column of **Table 1** shows the B+ supply voltage. As you would expect, the B+ voltage affects the maximum output voltage, which is listed in the next-to-right-hand column. The designs listed in **Table 1** set the triode's operating point for minimum distortion: the tube is operated in the most linear portion of its characteristic curve.

If the performance of the amplifier stage you want to design is represented by one of the lines on the table, you can jump directly to prototyping. But in many cases, optimizing a design requires you to optimize the Q point.

Optimizing the Q Point

First, we need the triode's plate characteristic curve. We'd like to have the transfer characteristic, the transconductance the μ, and the plate resistance plotted as a function of plate current, but these may not be available. The most commonly available set of curves is the plate characteristic. **Figure 2** shows this curve for the 12AX7A. Notice the straight red lines on the curve. These are called "load lines."

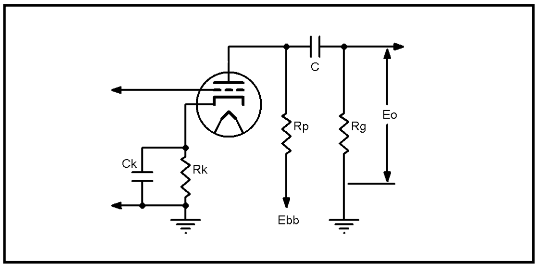


Figure 1: This schematic illustrates the component designations used in Table 1.

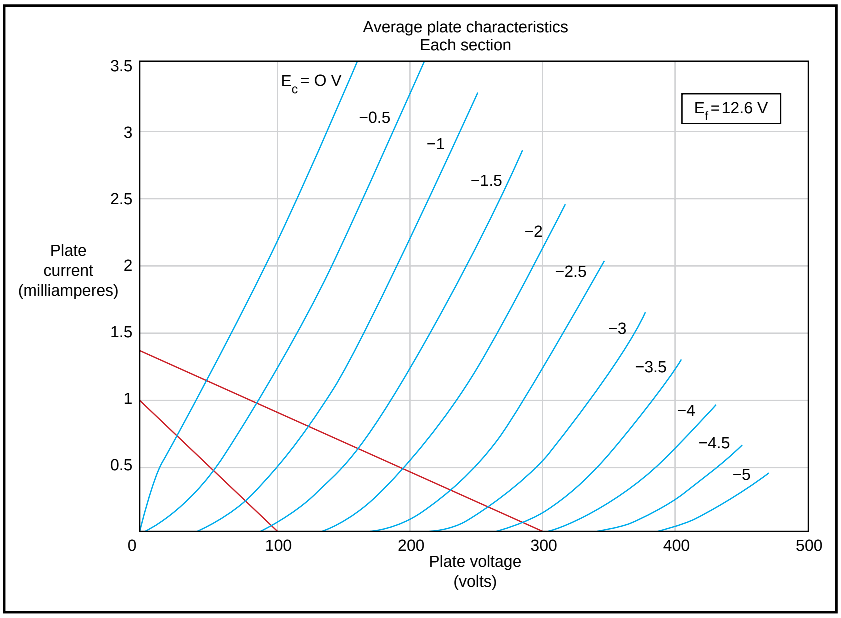


Figure 2: The most commonly available data curve in tube manuals is the plate characteristic, such as this one for the 12AX7A/7025/ECC83.

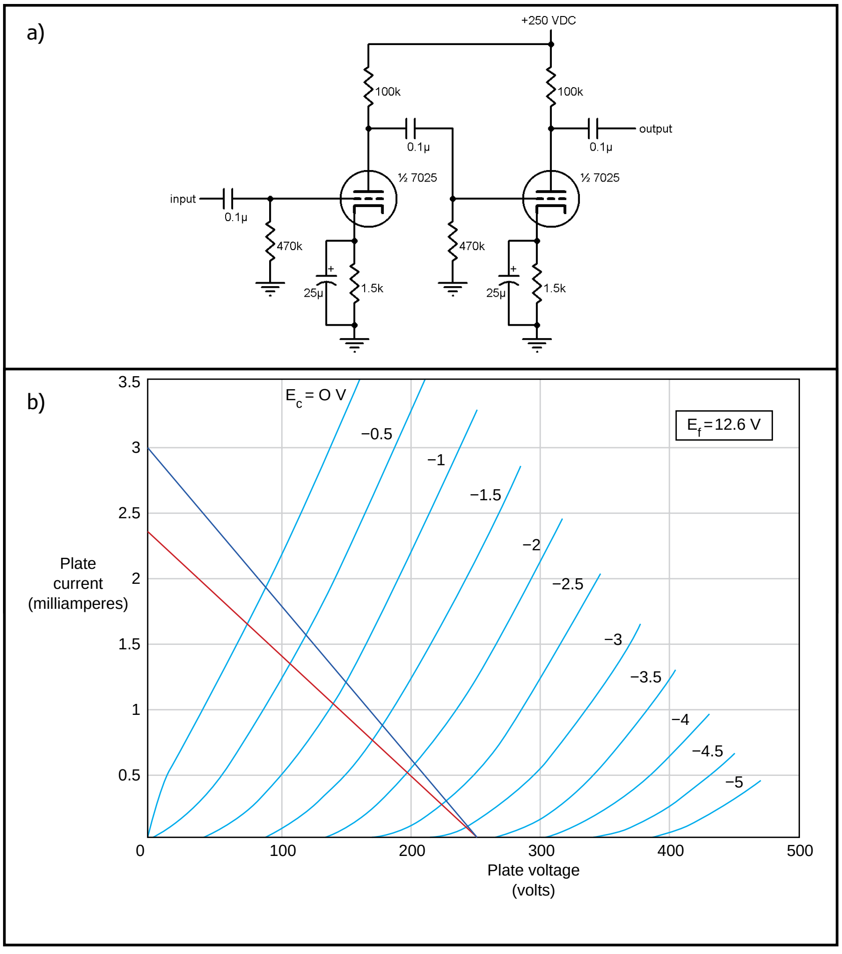


Figure 3a: The schematic is shown for a two-stage amplifier. b: The AC (blue) and DC (red) load lines for the amplifier's first stage illustrate the effect of the AC load on the tube's operation.

A load line is a graphical representation of all the combinations of plate voltage and plate current that a tube can have, given a certain B+ voltage and value of plate resistor. In **Figure 2**, the upper curve represents a circuit having a 300-V B+ and a 220-kΩ plate resistor. The lower curve is for a 100-V B+ and a 100-kΩ plate resistor. The vertical-axis intercept is the B+ voltage divided by the sum of the plate resistor and the cathode resistor values. It represents the maximum current that can flow in the given configuration. The horizontal-axis intercept is the B+ voltage, and represents the cut-off (zero current) condition. To draw a load line, you just mark these intercept points and draw a straight line between them.

The load lines shown in **Figure 2** only apply to DC conditions. When the tube has an AC signal, it sees two parallel paths to AC ground: the plate resistor and the AC load, which is often the input impedance of another amplifier stage. (Remember that any power-supply output is considered AC ground.) Thus to analyze AC behavior, we need to draw an AC load line. For the AC load line, the horizontal-axis intercept is still the B+ voltage, but the vertical-axis intercept is given by V_{B+}/R_{AC}. The AC load resistance is the parallel combination of the plate resistor and the resistance to ground fed by the output coupling capacitor for the stage. **Figure 3** shows a two-stage amplifier with the DC and AC load lines for the first stage. Note the difference in the slope of the load lines. This difference would be greater if the grid resistor of the second stage had a smaller value.

Maximum Signal-Voltage Output

The maximum signal-voltage output will be obtained when the control grid is biased so that the Q point is as close as possible to halfway between the B+ voltage and the plate voltage at which the AC load line crosses the zero-grid-voltage curve. (This plate voltage is sometimes called the "saturation" voltage, with saturation being defined as the point at which further positive increases in control-grid voltage do not produce corresponding increases in output current. However, true saturated operation is seldom achieved—or desirable—because grid-current limiting causes the signal to clip before saturation occurs. Grid-current limiting is caused by the flow of grid current through the parallel combination of the grid resistor and the signal-source resistance, building up a negative grid voltage that offsets the positive voltage being applied from the source. Only with very low-resistance sources is the effect of grid-current limiting small enough to allow the triode to truly saturate.)

Figure 4 illustrates the minimum and the maximum instantaneous plate voltages for stage one of the amplifier shown in **Figure 3**. The Q points shown by large dots would result in a quiescent DC plate voltage of 162 V, and a quiescent DC plate current of about 0.85 mA.

This would require a control-grid bias voltage of about −1.3 V (interpolated between the −1-V and the −1.5-V curves), and would give us an output swing of 250 V - 88 V = 162 V_{PP}. The necessary cathode resistor for a control-grid bias of −1.3 V at a plate current of 0.85 mA is 1.3/0.00085 = 1,529 Ω. **Figure 3** shows a 1,500-Ω resistor, which is close enough.

Figure 5 shows the AC and DC voltage gain calculations. This is one place where the significance of different AC and DC load lines shows up. The voltage gain at the Q point is given by:

$$A_V = \frac{\Delta V_P}{\Delta V_G}$$

For the AC gain, this amounts to:

$$\frac{196-142}{-.8-(-1.8)} = 54$$

The points for calculating the DC load line are shown in red, and the DC gain is:

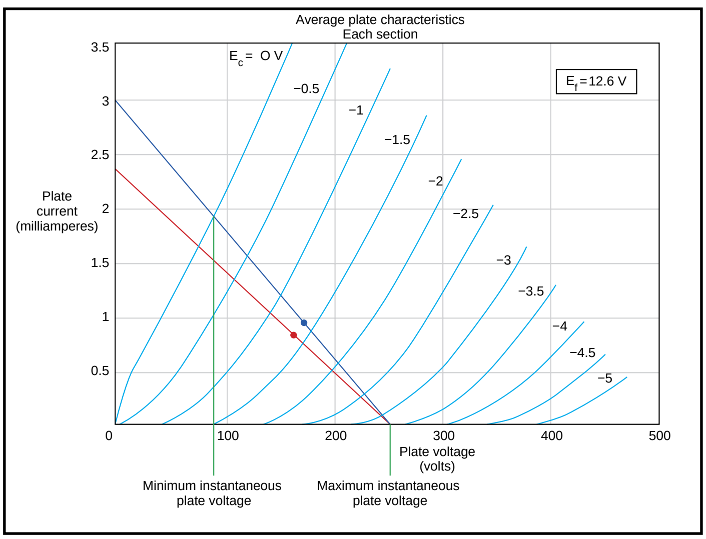


Figure 4: The instantaneous maximum and minimum plate voltages indicated by the AC load line reveal the maximum peak-to-peak output voltage.

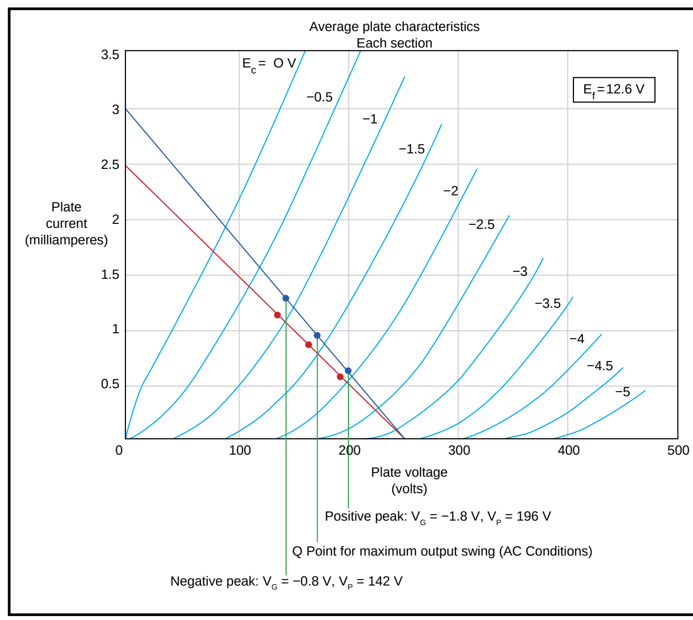


Figure 5: The instantaneous output voltages for 0.5 V positive and negative grid-voltage excursions can be used to calculate voltage gain.

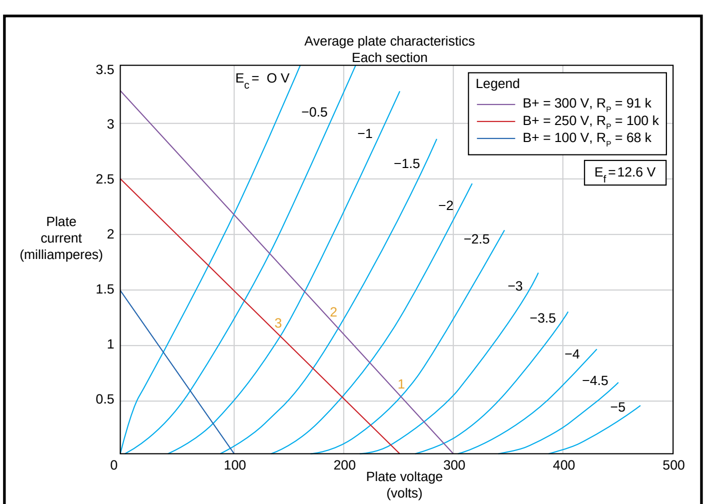


Figure 6: This is the set of curves and load lines from which the input characteristic shown in Figure 7 is derived.

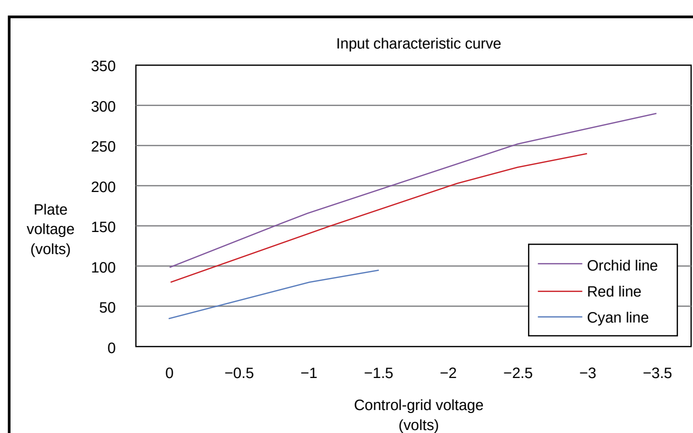


Figure 7: The different input characteristic curves show the behavior of the triode using the different load lines and Q points shown in Figure 6.

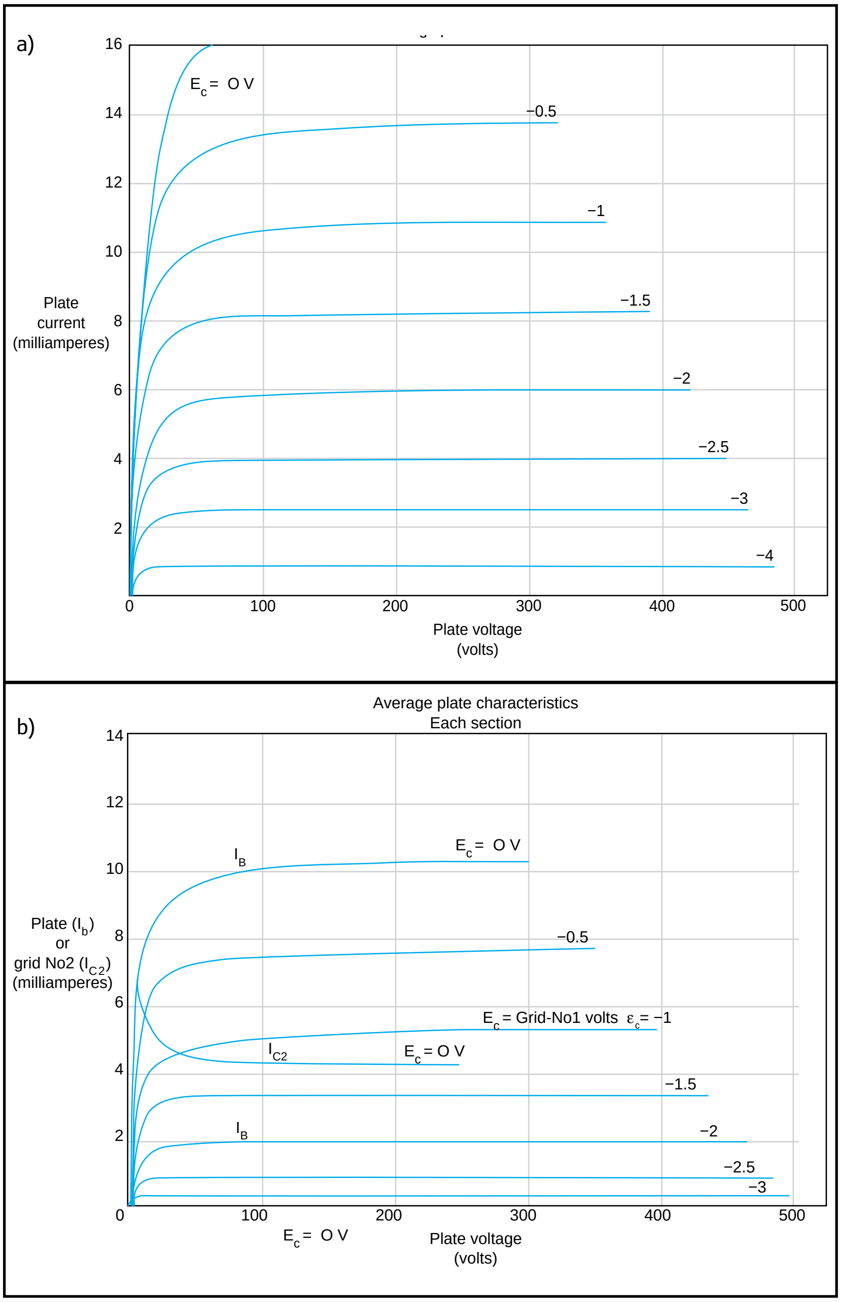


Figure 8a: This is the plate characteristic of a 6AU6A pentode using a 150-V screen voltage. b: This is the plate characteristic of a 6AU6A using a 100-V screen voltage.

Reference

The RCA Receiving Tube Manual, Radio Corporation of America (RCA), 1975.

$$\frac{192-135}{-.8-(-1.8)}=57$$

(The DC minimum and maximum voltages can be read from the graph.) Of course, the difference is caused by the loading introduced by stage 2's grid resistor. If the signal frequency is so low that the coupling capacitor has an appreciable reactance, this reactance would factor into the gain, reducing the loading effects and bringing the AC voltage points of the gain closer to the value of the DC gain. But through the voltage-divider effect, less of the stage 1 plate signal would be transmitted to the stage 2 grid, more than canceling the increased gain.

Distortion

Distortion is mainly caused by the same ΔV_G resulting in different values of ΔV_P. Thus, a Q point such that equal ± grid-voltage changes produce equal ± plate voltage changes gives you minimum distortion. Conversely, you can get more distortion by choosing a Q point that gives you unequal plate-voltage changes for equal grid-voltage changes. You can see this by looking at the spacing between the plate characteristic curves, and noticing where the spacing is, or is not, even. It is easier to see if we construct an input characteristic curve. We do this by looking at the load line and plotting the plate voltage versus the grid voltage. You can do this manually or using spreadsheet software.

Figure 6 shows a 12AX7A plate characteristic with three load lines, one of which has two different Q points. **Figure 7** shows the input characteristic for each load line. A perfectly straight input characteristic would indicate complete freedom from distortion. The more curved it is, the greater the distortion. So for the orchid load line, Q point 1 (250 V, using a −2.5 V grid bias) would give us some distortion; whereas Q point 2 (about 190 V, using a −1.5 V grid bias) would be cleaner. On the red line, Q point 3 (140 V, using a −1 V grid bias) is another clean solution. The slope of the input characteristic gives us the DC gain.

Maximum Plate Dissipation

Every hollow-state device has maximum plate dissipation. This is the power that can be safely dissipated by the plate without damage to the internal elements and supporting structures. For a Class-A amplifier such as we have been discussing, the plate dissipation is simply the product of the plate voltage and plate current at the Q point. A 12AX7/7025/ECC83 has a maximum plate dissipation of 1.2 W. For a quiescent plate voltage of 250 V, this implies a maximum plate current of 4.8 mA. Class-A amplifiers are rarely operated anywhere near the maximum plate dissipation, although when driven into clipping, (no longer Class-A), they may approach the maximum. One reason 12AT7s are sometimes chosen over 12AX7s is that they have a 2.5-W maximum plate dissipation, allowing them to safely drive lower-impedance loads such as reverb springs.

Setting a Pentode's Operating Point

Setting the operating point of a pentode involves one more decision than is required for a triode: choosing the screen-grid voltage. **Figure 8** shows the plate characteristic curves for a 6AU6A pentode, with screen-grid voltages of 150 V and 100 V, respectively. As you can see, increasing the screen-grid voltage increases the plate current. The linearity of operation is indicated by the evenness of the spacing of the plate-current curves as control-grid voltage is changed in uniform increments.

Note that in **Figure 8b**, the control-grid voltage increments are 0.5 V; whereas in **Figure 8a**, the bottom two curves are separated by a full volt. It is easier to see the effect of control-grid biasing on linearity by looking at the input characteristic shown in **Figure 9**. The curve is more nearly linear if the control-grid voltage is near –1.5 V. **Figure 10** shows the AC load line for a 6AU6A with a 50-kΩ AC load resistance (RP and RL in parallel), a B+ supply of 250 V, a screen grid at 100 V, and a control grid at –1.5 V, yielding a quiescent plate current of about 3.2 mA. Just by looking at the plate characteristic, you might not expect that this would be the best point to bias the tube for maximum linearity, but appearances can be deceiving!

If we had used 150 V on the screen grid, the most linear results would still be obtained with a quiescent plate current of about 3.2 mA, but a control-grid bias of –2.75 V would be required to obtain that current. Guitarists can think of the screen-grid voltage as a master volume, and the control-grid voltage as the channel volume: either control position affects the overall gain (or in the case of the grid voltages, either one affects the plate current).

You may also notice that the pentode, which can have much higher gain than a triode, also has a lower maximum peak output voltage at the same distortion level. And the pentode has more odd-order harmonics and higher-order harmonics than the triode, resulting in a very different sound.

The plate dissipation for a pentode is calculated in exactly the same way as for a triode. In small-signal amplifiers, plate dissipation is rarely a concern. Pentodes also dissipate power in the screen grids, but damaging levels of screen-grid dissipation almost never occur in normal operation.

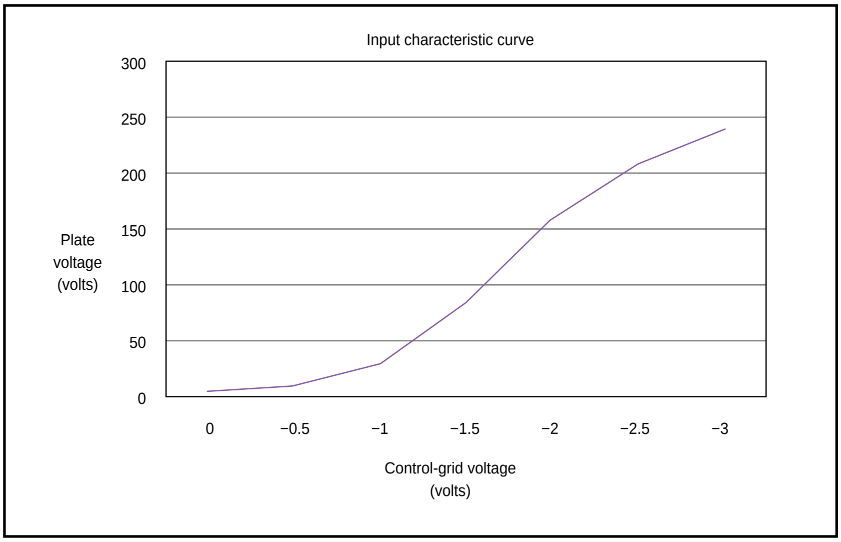


Figure 9: This is the input characteristic for a 6AU6A with a 100-V screen voltage.

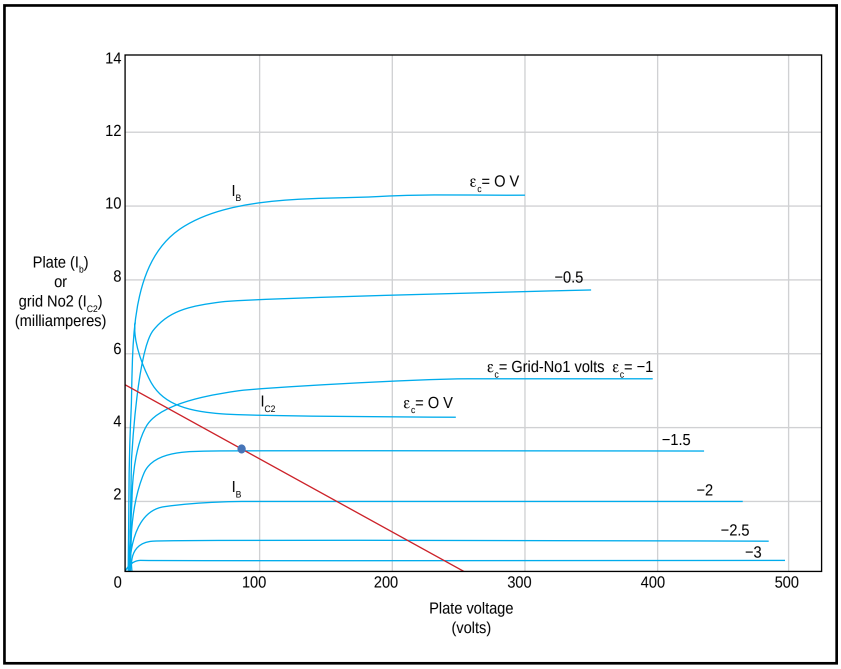


Figure 10: This is the load line from which the input characteristic shown in Figure 9 was derived.

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:

Summer NAMM 2015

July 9–11, 2015
Nashville Music City Center
700 Korean Veterans Blvd., Nashville, TN, US
www.namm.org/summer/2015

The Summer National Association of Music Merchants (NAMM) 2015 is expected to be one of the most vibrant shows for the music instruments and audio industry, with a strong emphasis on recording technology. The annual summer music industry gathering mixes a business-friendly environment with great entertainment and networking opportunities. The Summer NAMM event also provides a dynamic platform for emerging and established brands to create lasting impressions. Several companies, including Marshall Amplification, Washburn Guitars, Santa Cruz Guitar Company, Zildjian, DigiTech, Seymour Duncan, and Amati's Fine Instruments are returning to the show floor for the first time in many years. New in 2015, TEC Tracks offers educational sessions focused on recording and live sound production in an area adjacent to the NAMM Idea Center.

California Audio Show

August 14–16, 2015
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.

SET Expo 2015

August 23–27, 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.