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The Best Sound You Will Ever Wear

Recently I wrote an editorial for *The Audio Voice* (our weekly e-newsletter) on “why you need to look at Apple Watch and wear it.” In that article, I introduced a few of the reasons why we cannot ignore Apple’s new Watch.

Like the iPhone, it’s not about the object, the new platform, or the design. It is all those things put together and the way they interact to revolutionize so many different areas, from security to payments, from communications to health monitoring. And that’s just the visible intention—as if it wasn’t enough—on Apple’s road map.

I believe the Apple Watch is just the beginning of a “wearable” revolution in electronics and that the implications will be massive. Remote control is the most obvious application, available from day one on the Apple Watch. We already heard from audio companies, such as Apogee Digital, who are releasing an Apple Watch remote control app for recording with the new Apogee/Sennheiser ClipMic Digital solution. But we should also think about less obvious things.

The Apple Watch, in itself, features wireless communications and audio capabilities; complementing the computing power we carry in laptops (yes, I think they will still be around), tablets, and smartphones. It can be used as a direct replacement for a mouse, for instance, allowing simple gesture recognition. We’ve also seen it used for gesture interaction with musical instruments—check out the Artiphon Instrument, recently funded on Kickstarter (www.artiphon.com). You will see how it can be used to play “air-violin.”

As biometric devices and sensor-hubs, the implications for wearables are huge, and Apple is right when it decided to start promoting health and fitness applications. And of course, it will be huge for authentication, wireless payments, access control, security, and entertainment.

All the companies who have recently developed tablets and smartphones apps to remotely control their music players, portable wireless speakers, or music servers will have new options because the best remote control is something we can wear, not carry in a pocket or place on a tabletop.

But let’s move on to real audio applications, beginning with audiology. Two major hearing-aid companies, ReSound and Beltone, recently announced new apps for Apple Watch allowing preset and personalized listening scenarios, monitoring, control, and calling. Also, Israeli company Glide, announced an app to deliver live-video messaging on the Apple Watch. The “hands-off” function works with the iPhone, but if it can stream video, it certainly will do better with audio.

In today’s electronics industry, technology innovations are created in parallel with simple evolutionary development. And that’s based mostly on “development platforms.” Examples include Audinate’s work with the Dante technology and development platform, over which professional audio companies are creating a completely new range of audio-over-IP (AoIP) applications. Another example is Valens Semiconductor and its HDBaseT technology, currently powering connectivity in residential and commercial AV installations, and replacing multiple cables and interfaces (HDMI, Ethernet, USB, control signals, and up to 100 W of power) with a single Ethernet cable (using just the physical layer).

Apple Watch is “another platform” extending the company’s ecosystem of operating systems, software development tools, APIs, and hardware hosts. Will Apple be the only company leveraging this strategy? Certainly not, but its lead, which can be seen in the 64-bit architecture and the fact that the company controls all its hardware, software, and services, will enable Apple to maintain a significant edge over the competition.

For most audio companies, it becomes obvious that there’s plenty to be gained on leveraging that platform, and that’s why the Apple Watch is relevant.

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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Electrostatic Headphones

In this month's article, we review the current state of electrostatic headphone technology and provide a bit of electrostatic history.

By
Mike Klasco and Steve Tatarunis
(United States)

Photo 1: Arthur Janszen, a young naval engineer designed an electrostatic tweeter element and made it ready for commercial production. This is an ad for the JansZen Electrostatic Tweeter circa 1952. (Photo courtesy of Retro Vintage Modern Hi-Fi, itshifi.com)

Conventional headphone drivers, ribbon planars, and most air-motion transducers (AMTs) are electrodynamic. As with most loudspeaker drivers, they all use a magnetic structure and some sort of voice coil (the ribbon planar uses a flat printed circuit coil). On the other hand, electrostatic headphone drivers are similar to condenser microphones, and work more like a capacitor. Many, but not all, are used in an open-back configuration, and most require a high-voltage power supply, although we will also explore low power or self-biased electrets.

With the electrodynamic speaker dominating the market, why bother with electrostatics? Realistically, electrostatics require special amplifiers with some serious bias voltage, which makes mobile operation a challenge, not to mention expensive. The answer as to why electrostatics have a place in the sun is the electrostatics' combination of desirable properties. Electrostatics provide more transparent sound due to the diaphragm being uniformly driven over its entire surface, low moving mass resulting in "fast" response, and very low distortion.

Due to these positive attributes, electrostatics have the inherent ability to produce an extended top-end and flat frequency response with low distortion. The benefits of high-resolution sound reproduction are audible, measurable, and rarely equaled by few competing technologies (e.g., ribbon planars). For readers who want more background information about electrostatics, *audioXpress* published a three-part series in the July, August, and September 2014 issues (see Resources). For those who want more extensive information, Segment Publishing offers two books—*Electrostatic Loudspeaker: Design and Construction* by Ronald Wagner and *The Electrostatic Loudspeaker Design Cookbook* by Roger R. Sanders.

Some Electrostatic History

Before World War II, Chester W. Rice and Edward W. Kellogg at General Electric researched electrostatic speakers, but the large diaphragm size needed for low-end response and lack of appropriate diaphragm materials at the time made the electrodynamic cone speaker the low-hanging fruit and electrostatic development languished until after the war.

In 1947, Arthur Janszen, a young naval engineer, took part in a research project. The Navy was interested in developing a better instrument for testing microphone arrays. Janszen believed that electrostats were inherently more linear than cone speakers, so he built a model using a thin plastic diaphragm treated with a conductive coating, which exhibited remarkable phase and amplitude linearity. By 1952, he had an electrostatic tweeter element ready for commercial production (see **Photo 1**). Although far from the first electrostatic headphone, Janszen's basic design was a model for what must be the world's most geeky looking headphone, the Jecklin Float which was introduced around 1975 (see **Photo 2**).

Electrostatic headphone history begins with STAX, Ltd., the most well-known and respected electrostatic brand (see **Photo 3**). STAX introduced the SR-1 headphone back in 1959 and has focused on electrostatic headphones ever since. Outside of Japan, the STAX brand was quite obscure in the 1960s. Even in Japan, its customers were limited to the "audio maniac" contingent.

Koss was one of the main headphone brands in the 1960s, coining the term "Stereophones," and the company really shook up the industry when it launched the ESP-6 in 1968. This was the first mass production electrostatic headphone. It sold for approximately $160 (about $1,100 in today's dollars), which is an unheard of price point at the time.

To maintain compatibility with AM/FM receivers of the day, Koss utilized step-up transformers inside the earcups. For best results, the headphones could be used with an adapter ("energizer") that enabled them to be hooked up to an amplifier's output terminals or to the receiver's speaker-outputs (usually the receiver's second set of speaker outputs). The ESP-6s sounded pretty good, but they were a closed-back design, heavy, and uncomfortable. With positive customer feedback about the sound, but poor feedback on everything else, new models contained an external "energizer box" for connection to a receiver as well as improved comfort, ergonomics, and performance (see **Photo 4**).

Koss continued making the ESP headphone line through 1970s, 1980s, and 1990s with one or two new electrostatic models produced in each decade. The current Koss electrostatic model is ESP-950. It's the company's first electrostatic model to be released with a fully functional amplifier instead of an energizer and is positioned as a transportable system that consists of headphones, an amplifier, various cables, a battery, and wall wart adapters.

Photo 2: Jecklin Float is an iteration of the Jecklin earspeaker frame, designed by Joerg Jecklin.

Photo 3: The SR-009 is STAX's new flagship electrostatic headphone. Photo courtesy of STAX, Ltd.)

Photo 4: Koss was one of the main headphone brands in the 1960s, coining the term "Stereophones." The company really shook up the industry when it launched the ESP-6 in 1968. (Photo courtesy of Preservationsound.com)

However at $1,000, it has some formidable competition.

In the early 1970s, the Koss electrostatics only had a couple of true electrostatic contenders for competition. One was from Stanton/Pickering, introduced in 1971, these electrostatics were lightweight and wild looking—picture "futuristic Tupperware on steroids." One of this article's authors, Mike Klasco had the opportunity to spend some quality time with them. (After school, he worked in an audio store that had them on demonstration).

While Pickering had some early experience in the mid-1950s with electrostatic top-end tweeters, the Stanton electrostatic headphones missed the mark. These headphones were lightweight, but so was the sound. They had no dynamic range or any bass and they soon disappeared from the market.

Speaking of "space cadet" headphones, the aforementioned Jecklin Float also appeared in 1975. They were essentially a pair of electrostatic panels that hung on either side of your head. From that same era, the Marantz SE-1S electrostatic headphones and companion EE-1 "energizer" were also quite popular and affordable.

Photo 5: Each Sennheiser Orpheus HE90 set comes bundled with a dedicated tube amplifier, the HEV 90. (Photo courtesy of Sennheiser)

Electrets

In the September 2014 issue of *audioXpress*, we explored a class of electrostatics, the self-biased electret. Electret headphones are a specific class of electrostatic headphones. Electrets are unique in the method by which a fixed electrostatic charge is placed on the diaphragm. It is essentially a static electric charge. There are two main mechanical differences between an electrostatic headphone and an electret electrostatic headphone—the diaphragm material with a conductive coating and a high-voltage supply bias.

There have been a dozen or so electret headphone designs over the years. Sennheiser's Unipolar 2000 was probably the first open-back electret headphone on the market in the 1970s. Japanese brands such as Audio-Technica, Panasonic, Pioneer, Technics, and STAX also offered electret headphones in the 1970s.

Early electret headphone drivers did not have adequate sensitivity to be driven by the headphone outputs of receivers (which were notoriously padded down by resistors). The headphones required their own external supply amplifiers and would definitely not work with the meager output of today's smartphones.

Over the years, there have been quite a few pure electret headphones, but they have all disappeared, in part, because good results and high sensitivity can be obtained with the modern $1 to $3 factory cost headphone electrodynamic drivers. The exception has been the STAX electrets. STAX's main focus is true electrostatics. But, it has offered several self-biased electret headphones over the years and still offer them in today's product line. STAX also still produces the electret headphone drivers for these designs.

STAX is truly a "condenser-centric" company. Its first product was a condenser microphone in 1950, followed by a unique condenser phonograph cartridge in 1952 and an electrostatic tweeter in 1954. By 1960, STAX was shipping its first electrostatic headphone, which is the world's first electrostatic headphone. Throughout the 1960s, more electrostatic products followed, including more headphones and several full-range electrostatic speakers, the ESS series. By the late 1970s, STAX had developed and had commercialized self-biasing electret phonograph cartridges and electret headphones. Following was various open-backed, closed backed and off-the-ear (Earspeaker) configurations. More recent developments at STAX include electrostatic earphones and the acquisition of the firm by Edifier in China.

Other Notable Products

Perhaps the most legendary electrostatic headphones are the Sennheiser Orpheus HE90, which cost $16,000 when they were introduced in 1991 and typically go for double that today on the used equipment market. The story goes that in the early 1990s, Sennheiser gave its engineers a mission: make the best headphones, irrespective of price and outdo those Japanese guys (STAX) on all fronts. The Orpheus HE90 was made in very limited quantities, with a total of 300 produced. Each set comes with the HEV 90, a purpose-built tube amplifier (see **Photo 5**). Sadly, such an ambitious start in electrostatics was followed (and ended) by the Sennheiser HE 60 "baby Orpheus" electrostatics, which cost much less—and partially because of its less than awesome amplifier—did not make quite the same impact on the reviewers nor the market as its bigger brother.

KingSound, a Hong Kong company known for its full-range electrostats, showed its new electrostatic headphones last year at High End Munich. The series included the solid-state M-10 electrostatic amplifier or the tube variant, the M-20, along with the KS-H1 and the KS-H2 electrostatic headphones. The headphones are light and comfortable but in $1,000 to $2,000 price range, the headphone plus amplifier has some serious competition.

The Irvine, CA-based company, ENIGMAcoustics, is known for its patented self-biased electrostatic (SBESL) technology. Article author Mike Klasco had a chance to check out its flat-panel speaker products at High End Munich in 2014. He received a private demonstration of a headphone prototype in ENIGMAcoustics' demonstration suite at the 2015 Consumer Electronics Show (CES), and also at its R&D facility in Taiwan. ENIGMAcoustics recently announced the launch of two products in the high-end headphone category, the Dharma D1000 Hybrid Electrostatic Headphone and the Athena A1 Triode Vacuum Tube Hybrid Headphone Amplifier. The bias voltage is derived from the signal itself in a proprietary manner, making the Dharma "an easy to use, convenient, and with its light weight, a very comfortable headphone," according to the company.

The Dharma has a unique two-way (woofer and tweeter) hybrid electrostatic design (see **Photo 6**). Bass and mid-bass are handled by a proprietary dynamic membrane, with stiffness, mass, and internal damping optimized for use with ENIGMAcoustics' electrostatic high-frequency driver. Transition between the electrostatic and dynamic drivers is accomplished via a phase coherent first-order crossover.

Photo 6: The Dharma has a unique two-way (woofer and tweeter) hybrid electrostatic design. (Photo courtesy of ENIGMAcoustics)

Wrap-Up

Electrostatic headphone designs have been with us for more than 50 years. They are still relatively expensive and exist outside of the mainstream of dynamic driver consumer headphones, but for those of us who have experienced their open and natural sound, there's little argument that they represent the "best in class." There is a strong future potential for new self-biased electret headphones, as there have been recent advances in this technology (as can be seen in the ENIGMAcoustics products). Future development work may enable full-range electrets with more excursion enabling less expensive and complex implementations.

Next month, we take a deeper look at the challenges involved in using acoustic test instrumentation as a tool for the objective evaluation of headphone sound quality.

Resources

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Sound Cards for Data Acquisition in Audio Measurements

In this column, I will look at some of the available ways to create a low-cost system (emphasis on "low-cost") for lab-grade audio electronics measurements and provide some examples. I will skip the superb but expensive systems to concentrate on solutions within easy reach of budget-conscience hobbyists. The goal is to demonstrate how you can generate useful data, similar to the graphs and charts in the audio review magazines, but without the expense.

By
Stuart Yaniger
(United States)

I will confess to being a measurement geek. I firmly believe that anything you can hear, you can measure. The challenges lay in choosing appropriate measurements, properly performing them, and making reasonable interpretations.

A Bit of History

The basic measurements in audio electronics are voltage, current, impedance, noise, power, and distortion. Many years ago, when I was young and dinosaurs roamed the earth, the basic tools for these audio measurements were voltmeters, sine and square wave signal generators, and oscilloscopes. If your test bench was particularly well equipped, you might have a distortion meter (or two since the measurement of harmonic distortion and intermodulation distortion usually required separate instruments) and a test microphone. With those simple tools, audio amateurs performed all the basic measurements needed to construct the relatively unsophisticated circuits of the day. The measurement process was slow, tedious, and often uncertain.

Professionals used a few more advanced tools (e.g., impedance bridges, spectrum analyzers, and wave analyzers), but for the amateur, these were hideously expensive. For example, the Nicolet spectrum analyzer I used as a whippersnapper (state-of-the-art for its time) would sell for more than $200,000 in today's currency. It also outweighed me, so it was not the most convenient instrument to use. At the expense of adding a lot of tedium and time to the measurement of frequency spectra, you could use a somewhat less expensive and much less bulky wave analyzer (e.g., the Hewlett-Packard 3581A), but it still carried a rather prohibitive cost. Typically, we could only view the tools of professionals with burning envy.

Likewise, speaker measurements were done with swept sine waves or (if you were particularly cool) warble tones, point-by-point, graphed by hand. These manual methods suffered from the need to improvise an anechoic chamber (and it was rarely satisfactorily done). High noise levels, and again, time and tedium were major impediments. Commercial breakthroughs such as Crown's Tecron Time Delay Spectrometry (TDS) analyzer or the use of multiplex and impulse techniques were a blessing for professionals, but again, cost was a deterrent for most amateurs.

With the advent and evolution of the personal computer (PC), a revolution took place. One of the first shots fired was Bill Waslo's innovative IMP audio analyzer—about 20 years ago. The IMP was

Photo 1: It took some surgery to fit a PCI sound card into a low-level computer! I had to add a few additional components to use a normal-sized PCI card with my low-profile computer.

a sophisticated speaker testing system that could test loudspeaker frequency response and complex impedance using impulse response (later upgraded to MLS), and connected to a PC via the printer port.

Recognizing that sophisticated signal processing power was now available on inexpensive chips and that much of the cost and bulk of professional spectrum analyzers was the user input interface and display, the IMP cleverly took advantage of the early PCs' newly available graphics and input capabilities to provide a remarkably sophisticated tool for amateur speaker builders. (In this context, I have used PC in the general sense, not limited to specific brands or operating systems, though in this case, the IMP was only compatible with IBM-type computers running DOS.)

When I first saw Waslo's article, "The IMP Audio Analyzer," I literally ran to the mailbox to send off a check for the kit. Finally, some of the incredible capabilities I had at work for scientific measurements could be available to me at home for my audio hobby! And, I wasn't disappointed. The unit greatly increased my measurement capabilities.

On a commercial level, data acquisition systems (attachments to the computer that converted analog measurements into data accessible by the computer's CPU for calculation and display) started to become widely available to the professional community, but they were expensive and required some sophisticated user skills in programming them to get usable measurement results.

Although the audio amateur and semi-professional electronic design and construction market for test equipment is miniscule, we can often leverage technology from other, larger markets. The next step in the revolution was the widespread introduction of sound cards to PCs, which targeted computer gamers. The sound cards interfaced the computer with playback transducers (headphones and loudspeakers) and microphones.

Some clever people recognized that the sound cards were, in essence, data acquisition systems themselves. They could take analog data and convert them to computer-readable files or take computer-readable files and convert them to analog outputs. The development and refinement of the appropriate software and hardware has vaulted amateur measurement capabilities into arenas formerly the exclusive domain of the pros, and inadvertently, pushed professional audio equipment manufacturers to seek new performance levels, measurement capability, and user-friendliness.

Computer-aided measurements can, unfortunately, be an efficient way of quickly amassing huge amounts of nonsense, so I'll also try to warn you away from the more common pitfalls so that your measurements are more likely to be meaningful.

Hardware Basics

There are several "flavors" of sound cards out there (Sound cards intended for the music recording market are usually called "audio interfaces," so don't let my use of "sound card" limit you).

Built-In Sound Cards: Most modern computers come with built-in sound capability, generally using on-board chipsets. In my experience, the performance of these systems has been mediocre, but it's easy and quick to check.

One problem with a built-in solution is that often the only accessible analog input is a microphone level input that might even be mono, which is the case for the Lenovo laptop I used to write this article. You'll need a line-level input to really get the most out of your built-in sound card. If that's available, a few minutes doing a loopback test (i.e., connecting output to input) with something such as the RightMark Audio Analyzer (which we'll discuss when we get to software options) will determine if your equipment has sufficiently low distortion and noise or whether you'll have to purchase an aftermarket solution.

Note that for loudspeaker measurements, the sound card performance can be quite a bit worse than for electronics measurements without messing things up. That's because loudspeakers tend to have significantly higher distortion than electronics, and generally, we're not looking at very low level signals. So if your primary interest is loudspeaker frequency response measurements, the built-in sound card in a laptop is likely good enough. However, it will still require an external amplifier, a test mike, and most likely a mike preamp.

PCI Sound Cards: A major step up is a high-performance aftermarket card that plugs into the PCI slot of a desktop computer. It is often asserted that plunging the sound card into the depths of a computer is a sure-fire recipe for noise pickup. Maybe so, but I've used two different PCI sound cards in five different computers and only had one issue with noise among all those combinations. My experience may not be universal, though, so see if you can buy on a return basis.

The main market for the PCI format is music composition and gaming, so there's a wide range of available performance and pricing. 192-kHz sample rates are widely available.

Professional Tip: Do not buy a low-profile computer if you're planning to fit a normal-sized PCI sound card into it. Otherwise, your setup may look like mine (see **Photo 1**). Despite the horrible appearance, this actually worked pretty well.

As an example, **Figure 1** shows the intermodulation performance of a prototype phono stage with the spectrum acquired via the sound card. Note the nice quiet noise floor. The new PCI Express (PCIe) cards are suitably low profile and will obviate the need for the sort of radical surgery I had to perform.

External Sound Cards: Most of these are made for musicians and bedroom recording. That means they're mass produced at a relatively low cost. Putting aside audiophile snobbery, this is not a bug, it's a feature. Communication with the computer is most commonly through the USB port, though there's a few Firewire and Thunderbolt interfaces out there.

The potential upside of going external is removing the sound card's potential of noise contamination from being inside the computer case. There's also superior portability, since you're not tied down to a desktop machine as you are with PCI interfaces. The downside is potential noise and ground-loops from USB or Firewire power. Because of the intended market for external sound cards (amateur recording) and communications speed limitations with USB and USB 2.0, they are overwhelmingly available only with 48-kHz and 96-kHz sample rates (which I will discuss in greater detail later). However, we're starting to see some nice hardware with higher sample-rate capabilities starting to come down in price.

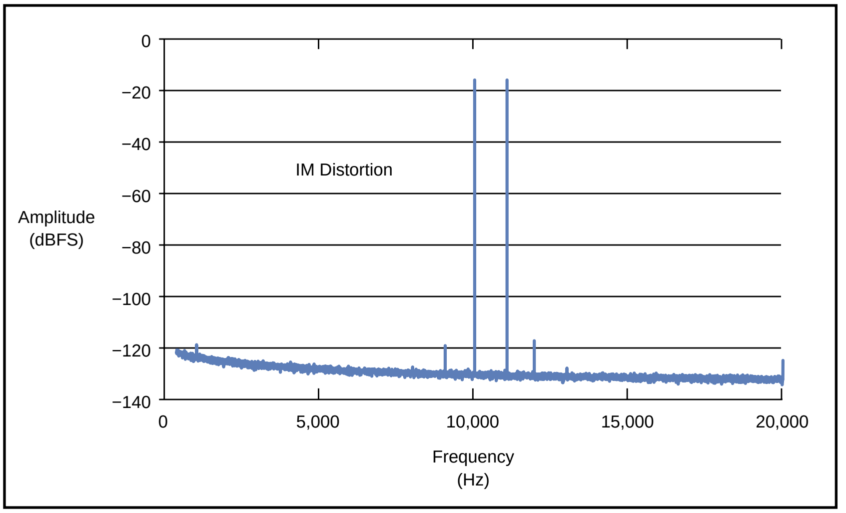

Figure 1: This intermodulation (IM) distortion spectrum of a phono preamplifier was acquired using the sound card shown in Photo 1. Full Scale = 4 VRMS.

Article Series Overview

In coming months, I'll provide an overview of some of the hardware and software options. Following that, we will consider what external hardware it takes to translate that into a useful instrument. Along the way, I will provide some examples of actual measurements and what it took to obtain them. I'll also discuss ways to design and build your own simple interfaces and jigs to do common measurements.

In addition, I'll review some specific pieces of commercially available software and hardware so you can get a good idea of the trade-offs and choose what's most appropriate for the measurements you want to do. Finally, we'll consider some advanced issues in measurement and how they can impact your results and the way the data are presented. Next month: Choosing a sound card and some computer tips.

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Classical Binaural Experiences

This is the seventh article in our multi-part series that takes readers on a continuing quest for realistic recorded sound. The series, which began in the December 2014 issue with mono, then stereo, and ambisonic sound, and focuses this month on binaural sound.

By
Ron Tipton
(United States)

Photo 1: The Neumann KU 100 Dummy Head Microphone is a replica of the human head with a microphone built into each ear.

Classical binaural recordings are made with a pair of microphones that simulates a human head. In some cases a "head" model is used. One example of a head model is the Neumann KU 100 Dummy Head Microphone (shown in **Photo 1**). It is a replica of the human head with a microphone built into each ear. Another example is the Kall binaural transducer shown in **Photo 2**.

Photo 2: The Kall binaural transducer simulates human hearing. The head has a special microphone built into each ear canal. (Photo courtesy of Kall Binaural Audio)

A less costly approach is a pair of small microphones that plug into your ears such as the Sound Professionals model SP-TFB-2-BEC (see **Photo 3**). An even less expensive option is a pair of in-ear Roland CS-10EM microphones. In all cases the microphones hear and record what you as a listener hears. That is, all sound that comes from the front, sides and rear.

When the recording is played back through a pair of headphones, the listener hears a reproduction of the original sound with all of the spatial location information preserved. As an example, I have included the classic2.wav file (which can be found in the Supplementary Material section of the *audioXpress* website). It is a classical binaural recording from the Kall Binaural Audio website. The recording demonstrates the 3-D effect when listened to using headphones.

Head Models

But first, the "head" model needs some explanation. It refers to the Head-Related Transfer Function (HRTF), a model that describes how a sound from a specific point will arrive at the ears. We humans estimate the location of a sound source by comparing the sound's arrival times and loudness as heard by our two ears. Because our ears are so close together, it would seem at first glance the sound difference would be negligible. However, our auditory sense has evolved to the point where we are adept at 3-D sound location. Each person has a different pair of HRTFs, one for each ear, but it's possible to construct an average or generic HRTF that is reasonably close for the majority of the population.

An HRTF involves a mathematical relationship between the input and the output functions. If we have an input and an HRTF, we can produce an output. Looking at **Photo 4**, we see a group of loudspeakers around an HRTF head. Assume the loudspeakers are reproducing a concert where each loudspeaker position was occupied by a microphone during the performance. By recording the head output, we have a two-channel copy of the multichannel pickup with all of its spatial information intact. The number of channels (loudspeakers) in the photo is perhaps an extreme case, but the setup works for any number of channels, provided the microphone and loudspeaker positions match.

Photo 3: The Sound Professionals SP-TFB-2-BEC binaural microphone places the mic element next to the ear canal, within the Pinna, for the most realistic recording possible from the perspective of the person wearing them.

Converting to Classical Binaural

Longcat Audio Technologies developed a digital audio workstation (DAW) VST plug-in that takes a stereo input file and converts it to classical binaural using a "generic" head model. The plug-in is not available but the trial version, which works for 30 days, is included in the Supplementary Material in the longcat.zip file along with a longcat-h3d.mp3, which is an example conversion from Longcat Audio's website. I recommend listening to it through headphones.

Bauer stereophonic to binaural DSP is a free VST plug-in for creating a binaural file from a stereo file for headphone listening. It, too, uses a generic head model. I have tested it in Sony Audio Studio and I prefer the "C.Moy" setting of the Default slider. It adds some crossfeed between the right and left channels, which seems to make the sound more lifelike. This plug-in, a webpage (htm) file with some interesting information about the plug-in and a converted MP3 file are included in bs2b.zip in the Supplementary Material. The htm file describes the plug-in theory with some nice details of how the control settings affect the sound.

The bs2b plug-in is simple, while the Panorama plug-in from Wave Arts with its 24 control knobs, is definitely not simple (see **Photo 5**). It is priced at $129.95 from Sweetwater, but a demo is available. I've included a demo copy along with some useful information and two example MP3 files in the panorama.zip file in the Supplementary Material. The reviews of this plug-in are mixed. With this many controls to adjust, it's apparently easy to go wrong and it seems the supplied presets are not good places to start. It is generally agreed that your input stereo file should be at least 24-bits at 96 ksps to minimize the degradation inherent in the processing. Maybe you can have too much of a good thing.

A Binaural Room Simulator

Jeroen Breebaart took a different approach to this conversion when he wrote the Isone Pro and Pro

Photo 4: The many loudspeakers grouped around a HRTF head, enable us to convert an array of sound sources to a two-channel binaural output. (Photo courtesy of NIRO Surround Sound Systems, www.niro1.com)

Photo 5: The 24 control knobs on the Panorama plug-in are not easy to adjust for a pleasing result. The manual explains each knob group, but the user is left to experiment.

Surround plug-ins. He used a binaural room simulator (see **Photo 6**). I've had fun with his different presets using Audio Studio as the VST host. This plug-in is no longer available but the demo (included in the Supplementary Material in the isone.zip file) seems to run fine. Updated versions are available from ToneBoosters as TB Isone and Isone Surround. I've included the Windows XP demo version in the Supplementary Material and the Windows 7 (and Mac versions) can be downloaded from the ToneBoosters website. A license for the ToneBoosters versions is reasonably priced at $25.

Binaural Music Sources

Radio France is providing binaural music for headphones through the www.bili-project.org website. I watched (it's also a video) and listened to Claudio Monteverdi's *Vespers*, the binaural effect was pleasing through headphones and even enjoyable through loudspeakers, although it was reduced to just stereo. If you have a slower-speed Internet connection, you will get better results going to www.lemouv.fr/rone-apache-binaural. The listening experience can be excellent with the lower bandwidth needed by the music-only format.

The British Broadcasting Corp. (BBC) does occasional headphone binaural programming on its Radio3 website (www.bbc.co.uk/rd/projects/binaural-broadcasting). You will find an immense amount of information about its ongoing binaural projects but

Photo 6: This screenshot of the original Isone Pro plug-in is functionally similar to the updated ToneBooster's version with areas for HRTF Adjustment and Room Acoustics setup.

Project Files

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

Resources

Bowers & Wilkins, www.bowers-wilkins.com/Society_of_Sound/Society_of_Sound/Music/sos-binaural-recordings.html.

HDTracks, www.hdtracks.com.

Kall Binaural Audio, www.kallbinauralaudio.com.

Last.fm, www.last.fm/tag/binaural.

Sony Audio Studio, www.sony.com.

SoundCloud, soundcloud.com/groups/3d-audio-binaural.

Sweetwater Corp., www.sweetwater.com.

ToneBoosters, www.toneboosters.com.

Veclip.com, www.veclip.com/tag/audio-binaural.

Wave Arts Inc., www.wavearts.com.

Sources

Koss headphones
Koss Corp. | www.koss.com

Neumann KU 100 Dummy Head Microphone
Neumann | www.neumann.com

CS-10EM Binaural microphones/earphones
Roland Corp. | www.rolandus.com/products/details/1081

Binaural microphones
Sound Professionals | www.soundprofessionals.com/binaural_mic.html

bs2b Plug-in
SourceForge | www.sourceforge.net

TDL Model 444A headphone amplifier
TDL Technology, Inc. | www.tdl-tech.com/data444A.htm

not much example music. Clicking on the Radio tab at the top of the home page brings up a recorded music list but it's mostly stereo.

A web search for Binaural Music Downloads provides a large collection of sources. I've listed a few of the more interesting sites in Resources. These files are mostly MP3, for a smaller file size, but they are binaural through headphones. A good source for more general binaural information is www.binaural.com.

Binaural Listening

Because headphones are preferred for listening to classical binaural, I have some comments about what works for me. Ear buds are lightweight and convenient but it's difficult to get good bass response from a 0.25" diameter or smaller diaphragm, regardless of what the ads say.

A pair of Koss ESP-6As have a good frequency response and come with an individually measured graph of the sound that gets to your ears. However, they weigh 2 lb and require an amplifier capable of driving their low impedance. (I designed the TDL model 444A Headphone Amplifier for low-impedance loads, which includes my ESP-6As.)

You can listen through loudspeakers, but in general, you won't experience 3-D sound. However, you can come close to the headphone experience with a pair of carefully positioned (4" to 8") monitor speakers, if you keep your head in the "sweet spot" centered between the speakers and 3′ to 4′ in front. Even if the sound is more spatial with headphones, I don't think many people enjoy being tied to them, even the wireless variety, so closely spaced loudspeakers may be more pleasant.

Processing is available to convert binaural into ordinary stereo while preserving the 3-D sound using normally spaced stereo loudspeakers, but I'm getting ahead of myself. I'll be describing the work of Dr. Edgar Choueiri and Dr. David Chesky (and others) and Fraunhofer Cingo in a future *audioXpress* article called "Binaural from Loudspeakers."

Author's Note: In my "Playing with Ambisonics" article from audioXpress *April 2015, my description for building a small first-order microphone array failed to mention that although the Panasonic WM-61A is an omnidirectional cartridge, mounting them with sealed rears makes them semi-cardioid. Also, their responses should be matched, but they are sufficiently inexpensive so you can purchase extras.*

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.

Genelec 8351 Acoustically Coaxial SAM System

Rethinking a Three-Way Monitor

Introduced during the 137th Audio Engineering Society (AES) Convention in Los Angeles, CA, the new Genelec 8351A Acoustically Coaxial Smart Active Monitor (SAM) System is a tri-amplified studio near-field monitor with an ingenious design that introduces enhanced directivity control at lower frequencies in a relatively small speaker. This brilliant design reflects the Finish brand's "outside-the-box" thinking and will help improve reference audio in recording studios.

Photo 1: It is important for a studio monitor to work equally well in vertical or horizontal orientations. The 8351 is a significant accomplishment for Genelec.

Introduction by
João Martins and Vance Dickason
Article by
Aki Mäkivirta, Jussi Väisänen, and Ilpo Martikainen
(Finland)

The mid- and high-frequency section of this three-way design is comprised of a coaxial 5" midrange with a 0.75" tweeter. The clever part, however, is the woofer section in which Genelec's engineering staff incorporated two 8.5" × 4" slot-loaded racetrack-shaped low-frequency transducers at the top and bottom of the enclosure. While this intuitively looks as if it might not be optimal in both horizontal and vertical planes, the opposite is true (see **Photo 1**).

When we look at the horizontal isobaric directivity plot shown in **Figure 1** and the vertical isobaric directivity plot shown in **Figure 2**, they are reasonably matched in performance. Obviously, it is important for a studio monitor to work equally well in both possible orientations. However, most monitors do not, so this is a significant accomplishment for Genelec.

Every sub-system in the 8351—electronics, the drivers, and the mechanical assembly—is designed by Genelec's R&D team and built entirely in-house at the Genelec factory in Iisalmi, Finland, to ensure accuracy and reliability.

In this article, we share the design process as explained by the authors—Aki Mäkivirta, Jussi Väisänen, and Ilpo Martikainen—in their paper "Design of an Acoustically Coaxial Three-Way Monitor," which was presented at the 28th Tonmeistertagung VDT International Convention, November 2014. The Genelec 8351 SAM System was initially introduced the prior month, in October 2014, at the 137th AES Convention in Los Angeles (see **Photo 2**).

The Acoustically Coaxial Design

The three-way acoustically coaxial monitor uses a minimum diffraction co-axial midrange/tweeter driver seated in a directivity control waveguide. The innovative solution allows acoustically concealing the woofers seated in the front enclosure under the waveguide, creating a large continuous front baffle surface for mid and high frequencies.

The two woofers, placed at the enclosure ends, acoustically combine to extend controlled directivity to bass frequencies. This acoustically concealed woofer system enables a compact enclosure with controlled directivity across the full audio band and directivity control on par with physically larger studio monitors. Horizontal and vertical characteristics are similar, enabling use in any orientation. For example, to minimize console top reflections, the monitor is vertically placed for high vertical directivity. If the side walls are close, the monitor can be horizontally mounted, reducing acoustic interaction with the walls. Other room influences can be minimized with smart signal processing features.

This design responds to the current needs of today's recording and mixing environments, which are shrinking in size and becoming more uneven in their frequency responses. At the same time, the production work must be done within the constraints of tighter budgets, yet with increasing quality requirements for the end product.

More coloration, larger differences between rooms, multiple types of production and repurposing of rooms during the day or week require reproduction systems with high neutrality for accurate editing work, and high sound pressure level (SPL) capability when required (see **Photo 3**).

Although a two-way active monitor is the industry's ubiquitous reliable workhorse, a three-way monitor design presents the performance optimum in many ways. A three-way monitor is needed for applications where:

- Very high-quality audio is required
- Accurate sound stage imaging is wanted
- It is necessary to hear subtle acoustic details, even at high SPL
- Accurate control of monitor directivity is necessary

The three-way monitor design enables refined selection and optimization of individual driver characteristics (i.e., sensitivity, linearity, output capacity, and directivity) because of the narrower frequency range for each driver (see **Photo 4**). Optimal crossover frequencies can be chosen based on the acoustical constraints given by the drivers and the enclosure characteristics. A larger surface area of the waveguide can enhance directivity and achieve the best performance in the presence of challenging room acoustics typical of small or acoustically compromised rooms.

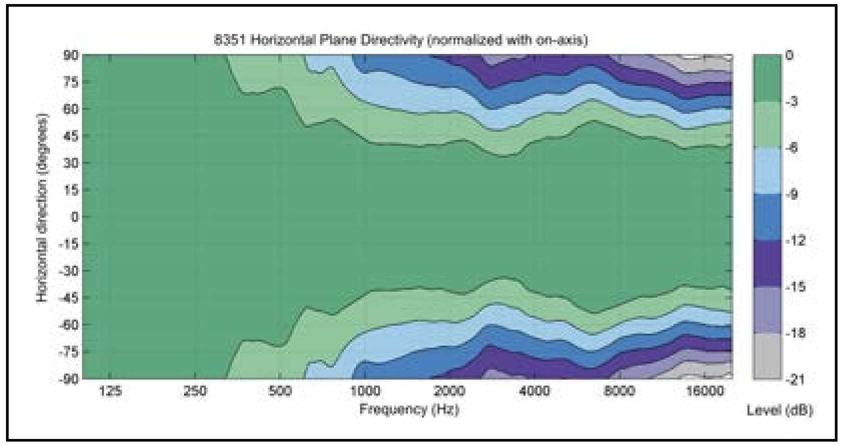


Figure 1: The 8351 monitor's horizontal plane directivity (normalized with on-axis) is shown.

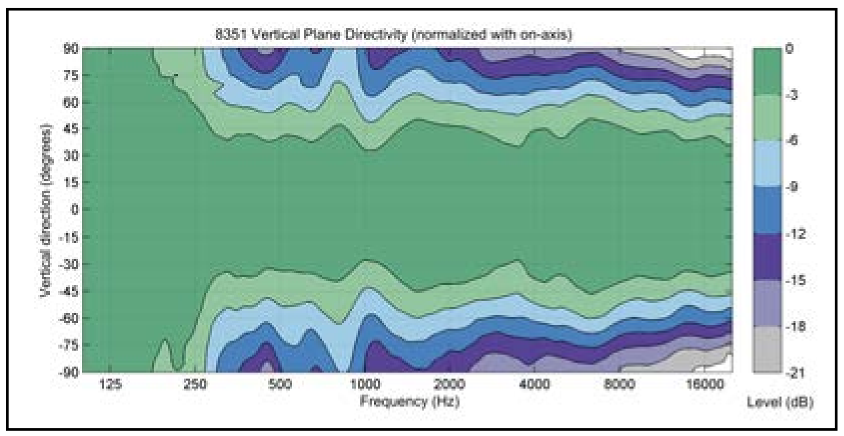


Figure 2: The 8351 monitor's vertical plane directivity (normalized with on-axis) is detailed.

Photo 2: Genelec's R&D Director Aki Mákivirta introduces the 8351 monitor at the 137th Audio Engineering Society (AES) Convention in Los Angeles, CA.

Photo 3: The Genelec 8351 monitor's acoustic performance reaches a peak SPL per pair ≥123 dB (referenced at 1 m) and a maximum SPL (half space, on axis, at 1 m, average 100 Hz to 3 kHz), with a frequency response from 32 Hz to 30 kHz (–6 dB), 38 Hz to 21 kHz (±1.5 dB).

About the Authors

Ilpo Martikainen is founder and chairman at Genelec. Jussi Väisänen is lead acoustic designer at Genelec. Aki Mäkivirta is the R&D Director at Genelec.

Genelec is a manufacturer of active loudspeaker systems based in Iisalmi, Finland. It designs and produces products especially for professional studio recording, mixing and mastering applications, broadcast, and movie production.

Photo 4: The 8351 monitor's response can be adjusted using six parametric notches, two low-frequency shelving, two high-frequency shelving, a 80 ms delay, and a 30-dB level adjustment. The connections are 1 x XLR analog in, 1 x XLR digital in, 1 x XLR digital out.

Benefit of Coaxial Monitor Concept

The coaxial driver arrangement has an important acoustic benefit. A multi-way monitor loudspeaker has two drivers simultaneously reproducing sound across the crossover range of frequencies. A conventional multi-way design has driver locations distributed across the front baffle. The summation of the outputs from any two drivers has been designed to be in phase in the primary listening direction (the acoustical axis). When the two outputs are in phase, the combined system output remains flat. For off-axis locations, the two drivers are not in phase because the distances to the drivers have changed from the design geometry. This can produce strong coloration in audio. The frequency of this coloration also changes when the listener moves further off axis. Placing drivers coaxially can eliminate this coloration for off-axis positions as the geometry remains the same for the off-axis positions.

The crossover performance has been further improved by time-aligning the two coaxially constructed drivers at the crossover frequency, producing the benefit of maintaining the time domain waveforms. This is particularly important for driver constructions with which the physical height of the tweeter and midrange drivers are significantly different.

Evolution of the Coaxial Driver

The concept of implementing a multi-way driver system coaxially is not new. In the 1940s, Altec Lansing launched the 601 coaxial driver with a section horn tweeter coaxially located in a 12″ woofer. This was followed closely by the Tannoy Dual Concentric design. By the 1970s, the UREI 813 three-way coaxial monitor became popular in audio monitoring.

When the woofer cone surface forms the waveguide for the high-frequency radiator, intermodulation distortion tends to be larger than it is for designs with a fixed horn located in the woofer's apex. The intermodulation can be reduced by directing low frequencies to a separate woofer driver, using the coaxial driver as a midrange-tweeter radiator. The intermodulation distortion problems typical for these designs can be significantly reduced with a physically displaced woofer in a three-way arrangement.

The fixed horn tweeter coaxial designs suffer from diffraction of sound around the horn's mouth. The sound diffracting around the horn's mouth travels to the woofer cone, then reflects backward and sums with the direct sound with a fixed delay and frequency specific level, creating sound coloration.

When we look at the small three-way monitoring products available today, we see four primary design types. The first design type is the conventional non-coaxial driver arrangement, in which individual drivers are arranged in the loudspeaker enclosure's front baffle, displaced from each other. When three drivers are crammed onto the compact front baffle area, typically there is no surface area left for directivity control in the midrange and woofer frequencies. It becomes difficult to arrange the drivers on the front baffle so that crossover coloration problems at the two crossover frequencies can be avoided for off-axis sound.

The second type is the approximately coaxial driver arrangement, which has an island containing a conventional tweeter and midrange driver on top of a woofer. In some approximately coaxial systems, all three drivers have been stacked but the driver axes may not be coaxial. Approximately coaxial designs allow smaller physical size for the monitor's front baffle but usually have problems with the drivers' height differences and the delayed addition of the diffracted audio to the direct radiation, coloring on-axis audio. Directivity matching can be problematic and cause coloration issues for the off-axis audio.

Today, the coaxial operating principle is enjoying a degree of renaissance as coaxial drivers can implement physically smaller multi-way monitoring systems. Both coaxial driver construction principles can be found in three-way monitors.

The two coaxial driver technologies used in three-way arrangements include variants of the concentric tweeter-midrange driver and the horn-loaded tweeter system (fixed tweeter horn) suspended coaxially with the midrange driver.

However, several characteristic problems remain with three-way monitor designs that use coaxial drivers. It is difficult to optimally design the seating and surface shapes of the coaxial driver. This can lead to poor control of diffraction and colorations.

Although the concentric arrangement tendency to cause intermodulation distortions can be alleviated by using a separate woofer, it is usually not possible to sufficiently optimize the midrange cone shape to make the midrange work as a high-performance waveguide for the tweeter. Seating of the midrange cone to the driver chassis and the joint between the tweeter and the midrange typically introduce diffracting edges in the design.

The fixed horn tweeter approach for a coaxial driver reduces the intermodulation problems but has other challenges. There is usually some frequency dependent amount of sound diffracted around the edges of the tweeter horn's mouth opening.

Figure 3: The minimum diffraction coaxial (MDC) design—a 130 mm (5") coaxial driver with midrange and a 19 mm (0.75") tweeter eliminates all diffraction sources, using an acoustically optimized seating of the tweeter and a smooth acoustically continuous surface from the tweeter outward. This design enables good blending to the directivity control waveguide.

This sound energy takes time to travel to the midrange cone surface and is then reflected back, and eventually sums with the direct tweeter sound a short time later. This introduces SPL direction and frequency-dependent variations, making it challenging to achieve flat frequency response on more than one axis for these designs.

Today's coaxial three-way monitors try to alleviate the inherent problems of the coaxial drivers by digital signal processing. Although it can improve the flatness of the frequency response, signal processing cannot correct for problems with directivity control, colorations in the off-axis positions, or the tendency of a coaxial driver to produce distortion.

Figure 4: Acoustically concealed woofers radiate low frequency through acoustically optimized openings in the ends of the front baffle. This enables the whole front baffle to be used as a waveguide.

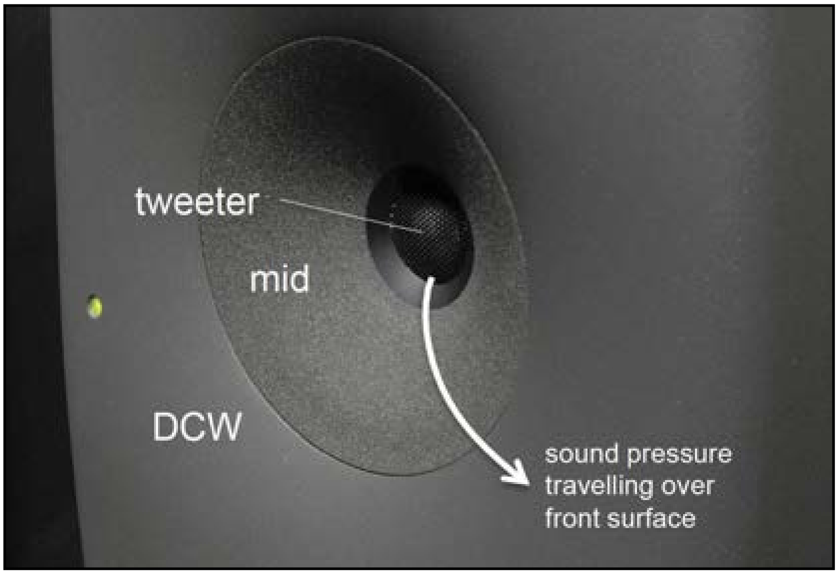


Figure 5: Genelec's minimum diffraction coaxial (MDC) driver enables excellent integration with the directivity control waveguide (DCW), avoiding diffractions and generating a flat frequency response on and off axis.

Acoustically Concealed Woofers

Genelec's novel approach consists of using a new acoustically coaxial minimum diffraction three-way monitor loudspeaker design. This design uses the entire front baffle as a directivity control waveguide by hiding two woofer drivers under the front baffle.

The monitor uses two 215 mm × 100 mm (8.5" × 4") woofers displaced out on the two ends of the enclosure front. The woofers are housed in a shared bass reflex enclosure having a reflex port opening to the enclosure's back. The total woofer surface area and the magnetic motor capacity have been maximized. The woofers occupy the remaining front baffle surface outside the minimum diffraction coaxial (MDC) driver unit, hidden under the front surface (see **Figure 3**). The total woofer diaphragm area is close to that of a 255 mm (10") woofer.

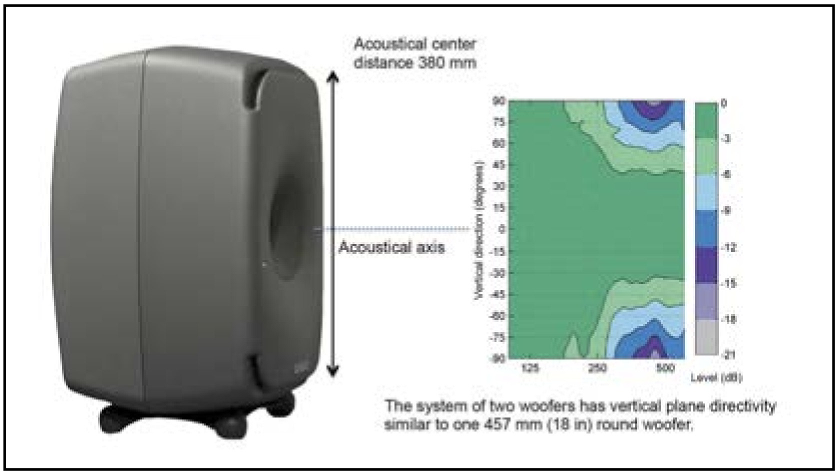


Figure 6: The diagram shows how the woofer system directivity extends to low frequencies.

The two woofers are located under acoustically optimized extensions of the front baffle. Woofer radiation happens through the openings, one for each woofer. This acoustical concealment is the cornerstone, enabling a compact coaxial monitor with good directivity control. Two spaced woofers create acoustic directivity extending to woofer frequencies. The directivity produced by the concealed woofer concept is considerable (see **Figure 4**).

The distance between the centers of the two woofers is 268 mm. However, the radiation occurs out of two acoustically optimized slots. These two slots are spaced with a distance of their acoustic centers of 380 mm.

This corresponds to 0.56 wavelengths at the highest operating frequency, which is 490 Hz at the woofer-to-midrange crossover point. This distance is short so the system of two woofers has directivity characteristics similar to one large round woofer driver with a diameter of 457 mm (18") in the vertical plane. On the horizontal plane, the directivity is determined by the longer axis diameter of the oval woofers.

Directivity Control

Matching directivities of the drivers in the acoustically coaxial three-way monitor improves the system performance.

The woofer system's combined acoustic axis created by the two spaced drivers radiating though the slot openings coincides with the acoustic axis of the minimum diffraction tweeter-midrange coaxial driver. This matching creates an acoustically coaxial three-way system with the added benefit of a woofer system with controlled directivity (see **Figure 5**).

The diffraction-free coaxial driver blending with the directivity control waveguide of the enclosure front creates a continuous optimized acoustic surface for the high frequencies radiated by the tweeter as well as the midrange. The coaxial driver design does not have any acoustic discontinuity, such as a gap or a step change of the acoustic surface profile, either at the joint between the tweeter, the midrange inside edge, or at the outer edge of the midrange. The coaxial driver functions without acoustic diffraction effects through its operating frequency range.

The large waveguide enables good control of directivity across the operating frequency range of the tweeter-midrange system. The waveguide geometry has been optimized to minimize sound coloring diffractions.

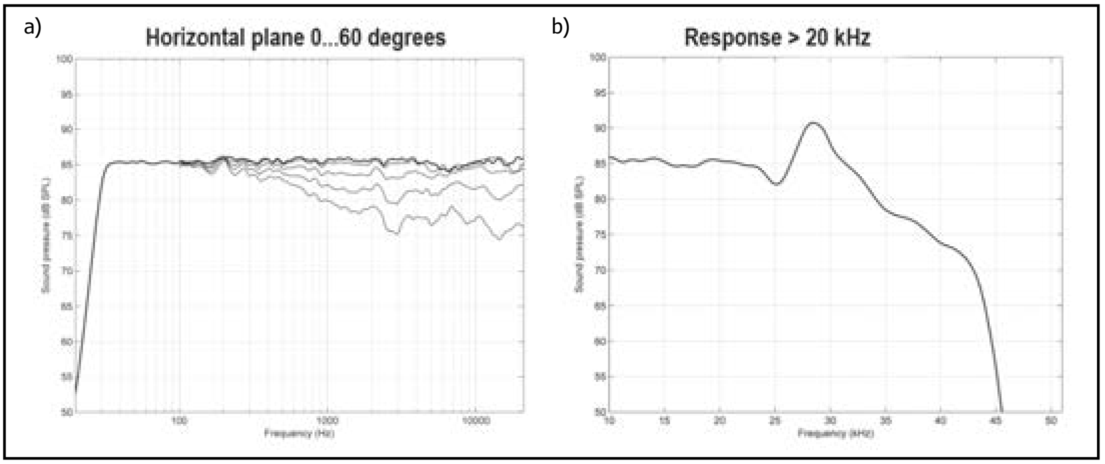


Figure 7: Frequency responses of the acoustically coaxial monitor design on the horizontal plane, on the acoustical axis, and at 15°, 30°, 45°, and 60° off axis (a). Typical extension of the response to frequencies above the audio range are shown (b).

The compact three-way design presents a waveguide surface area equal to that found in large size three-way main studio monitors. The directivity control remains down to 300 Hz in the vertical direction and down to 700 Hz in the horizontal direction (see **Figure 6**). Considering the enclosure's compact external dimensions, the monitor has excellent ability to control directivity, typically only found in monitors with significantly larger enclosure sizes.

Characteristics

The new acoustically coaxial three-way design uses two woofers 215 x 100 mm (8.5" x 4") and a minimum diffraction coaxial driver that contains a 130 mm (5") midrange and 19 mm (0.75") aluminum dome tweeter. These are powered by two 150-W Class-D amplifiers (a woofer and a midrange) and one 120-W Class-AB amplifier (tweeter). The crossover frequencies are 490 Hz and 2.6 kHz.

The external dimensions are similar to a typical two-way, with a 452 mm (17") height, a 287 mm (11") width, and a 278 mm (11") depth.

Analog and digital audio sources can be directly connected to the monitor. The analog input is capable of accepting signals up to +24 dBu. The analog input is converted for signal processing and reproduction. Monitoring of digital audio in AES/EBU format supports sampling rate up to 192 kHz. Digital audio is sample rate converted to ensure synchronization with all sources. An AES/EBU thru output provides daisy-chaining to more monitors.

The digital audio AES/EBU and analog audio input signals are filtered with digital signal processing at an internal 96-kHz sampling rate. There is a 48-kHz-wide audio path from the inputs to the acoustic output. Input-to-output system latency is approximately 5 ms in mid and high frequencies.

The low corner frequency of the coaxial monitor system is 32 Hz. Signal processing has an electronic bandwidth of almost 50 kHz, allowing system equalization to extend above the audible range (see **Figure 7**). The high corner frequency is typically 30–40 kHz (±6 dB). The on-axis frequency response shown in **Figure 1** and **Figure 2** is flat to ±1.5 dB (38 Hz to 21 kHz).

Because of the large directivity control waveguide effectively covering the monitor's entire

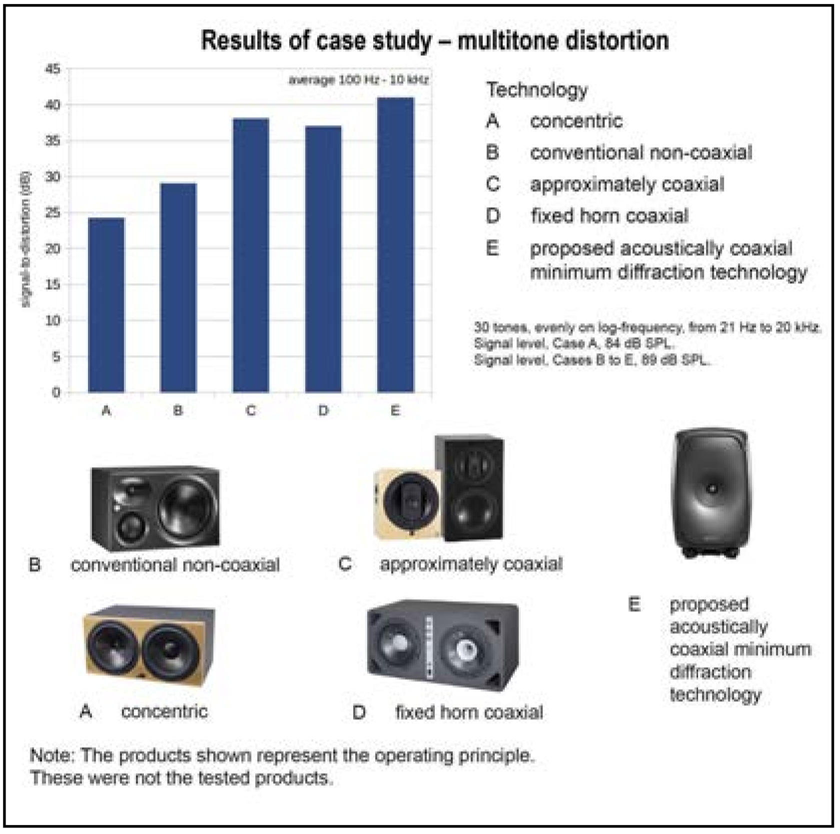


Figure 8: Multitone signal-to-distortion for case examples of different technologies: (A) concentric, (B) conventional non-coaxial, (C) approximately coaxial, (D) fixed tweeter horn coaxial, and (E) proposed technology. The classical concentric system is showing a fairly strong tendency for intermodulation distortion and a low signal-to-distortion ratio. This may be due to the driver's individual characteristics and may not reflect a universal principle.

Photo 5: The Genelec 8351A Acoustically Coaxial Smart Active Monitor (SAM) is used in a studio environment (http://parkstudio.se/info.htm) in Sweden, where Stefan Boman, the engineer and producer and also part owner requested a trial. (Photo courtesy of Lars-Olof Janflod)

front baffle, the frequency response neutrality can also be maintained for the off-axis directions. This implies that the color of sound radiated into the room, and audible as the late energy in the room, also maintains a relatively neutral character.

In a typical monitoring environment, the instantaneous peak output SPL exceeds 123 dB (referenced at 1 m) per a pair of monitors. For an individual monitor, the maximum short-term SPL measured in half space, averaged within the frequency band 100 Hz to 3 kHz exceeds 110 dB (referenced to 1 m distance). The long-term maximum SPL is limited by the overload protection to about 105 dB SPL.

The acoustically coaxial three-way monitor also features flexible signal processing tools (Smart Active Monitoring). These tools add a layer of intelligence to the monitors. The monitors connect to a control network. All aspects of the settings in a multi loudspeaker system can be adjusted at the listening position. Automated aligning of monitors and subwoofers with regard to level, timing, and equalization of room responses can reduce room influence on sound and improve accuracy of monitoring.

Performance

The performance of the proposed acoustically coaxial monitor construction was evaluated against the typical performance of the established implementations of compact three-way monitors. A multi-tone measurement signal was used. Multi-tone measurements can provide relatively realistic measurements of the overall linearity because the signal is a complex multi-frequency signal containing a wide bandwidth, essentially presenting a load similar to a realistic wideband audio signal. The test signal contains 30 tones distributed evenly on a log frequency scale across the audible range, from 21 Hz to 20 kHz.

The case examples have included the typical technologies currently applied in compact three-way monitor designs. The technologies were represented by case example products of the concentric type tweeter-midrange system (A), a conventional separate driver arrangement (B), approximately coaxial approach with the tweeter located on an island suspended over the tweeter and a displaced woofer (C), a fixed horn-type tweeter coaxial with the midrange driver (D), and the proposed acoustically coaxial minimum diffraction driver system (E) shown in **Figure 8**. It is worth noting that in this case study, technologies A and D were in fact two-way systems. This may affect the distortion figures for these two cases.

The multitone distortion to audio signal ratio was measured for each of the case examples and the distortion within the frequency range 100 Hz to 10 kHz was considered. It is worth noting that when the frequency range was limited to 500 Hz to 10 kHz, the results remained relatively the same. The signal level for Cases B to E was 89 dB SPL, and for Case A, 84 dB SPL.

It may also be noted that the conventional separate driver three-way system did not show any

Resources

Altec Lansing, "DUPLEX Loudspeaker System," http://alteclansingunofficial.nlenet.net/Duplex.html, September 2014.

J. Borwick, *Loudspeaker and Headphone Handbook* (3rd edition), Focal Press, 2001.

M. Dodd, "A Wide Dispersion Constant Directivity Dual Concentric Driver," Convention Paper for the 92th Audio Engineering Society (AES) Convention, Vienna, Austria, March 1992.

E. Dupont and S. Lipshitz, "Modeling the Intermodulation Distortion of a Coaxial Loudspeaker," *Journal of the Audio Engineering Society* (*JAES*), 2009.

Genelec, "8351A Three-Amplified SAM Monitor System," www.genelec.com/products-menu/products-by-series/sam-series/8351a.

P. Klipsch, "A Note on Modulation Distortion: Coaxial and Spaced Tweeter-Woofer Loudspeaker Systems," *Journal of the Audio Engineering Society* (*JAES*), 1976.

Tannoy, "Dual Concentric Loudspeaker Drive Units," www.44bx.com/tannoy, September 2014.

advantage in terms of the intermodulation distortion although theoretically, it should have an advantage over the coaxial systems.

Approximately coaxial, fixed horn coaxial and the proposed technology were similar in terms of the complex intermodulation performance but will show clear differences in terms of the off-axis coloration and on-axis frequency response flatness.

As the case study examples are believed to display state-of-the-art performance, it could be established that the proposed method showed favorable performance in terms of distortion characteristics.

Applications

Acoustically coaxial minimum diffraction design with concealed woofers implies that the three-way monitor can be used equally well in either orientation, vertical or horizontal. The acoustically coaxial three-way monitor's neutral acoustic character is retained for off-axis directions in either orientation.

Coaxial design also enables the listening distance to be unexceptionally short. The minimum listening distance for the proposed design is smaller than 0.5 m (1.5′). Freedom of orientation with the very short minimum listening distance means that an acoustically coaxial three-way monitor can work in near-field and mid-field applications.

The acoustically coaxial three-way monitor is slightly more directional along its long axis. This can provide benefits (e.g., when the console top reflection must be minimized). The monitor can be placed vertically, and the higher vertical directivity reduces the console top reflection level.

Since all the signal processing is done in the digital domain, it allows for excellent frequency response control. The signal processing can be tuned for the installation location and helps in system alignment (see **Photo 5**). An automated software-based procedure is available for the room alignment and equalization. The flexible room influence compensation reduces room coloration and improves accuracy of monitoring.

The compact acoustically coaxial three-way monitor offers a high-quality monitor option for space- or size-limited applications, where normally a two-way monitor would be chosen.

The Right Filter

This project details my "tunable near-linear phase analog crossover" design. Despite the fact that digital filters are more commonly used because they allow linear phase performance, I built my device using analog components because many people still prefer analog systems. Theories exist about how to perform linear phase crossovers (e.g., the well-known Lipshitz-Vanderkooy crossover). The presented method is easy without need for precise, expensive capacitors. Moreover, the cross-frequency will be tunable over more than two octaves.

Photo 1: Here is a look inside my tunable near-linear phase analog crossover project.

By
Vincent Thiernesse
(France)

In the first article in this series, I describe my method and give you the tools to build your own design. In the second article, I will describe the hardware for the project and the final touches.

The hardware project includes 1.8-to-7-kHz crossover-frequency tuning with a remote or hardware-fixed crossover-frequency from 1.8 to 7 kHz and a 3-kHz fixed crossover-frequency configuration with a fifth-order 20-kHz cut-off filter optimized for high quality.

Introduction

Overtime, digital filters have progressively replaced analog filters, but when the two systems do the same thing, you might prefer the analog one for several reasons—both subjective and objective. The Lipshitz-Vanderkooy crossovers are known to be near-phase linear. However, they are expensive and difficult to do because they require precise components due to the separate concept of the low-pass filter and an approximation of pure delay. The subject seems to have been widely abandoned since the emergence of digital filters. Nevertheless, recent studies about phase linearity have renewed the argument for analog filters.

With my design, I intend to prove that my method provides robust practice results with respect to component tolerances. The basic idea, which is not really new, is to simulate finite impulse response (FIR) digital filters with analog phasor networks where the pure delays provided by ideal temporal sampling are replaced by analog phasor cells as approximations of pure delays.

This approach differs from many classic approaches in which a near-linear phase low-pass filter is synthesized with physical independence with a pure delay (or a close approximation of it) whose subtraction from the low-pass results in the high-pass. However, the problem is obtaining practical results that are "not too far from theory" because the phase of both structures has to very nearly match.

Here, the low-pass and the high-pass will emerge from the same structure to ensure a near-perfect match of the low-pass and the high-pass in terms of gain flatness, phase linearity, and phase synchronization between low-pass and high-pass, even without taking strict design precautions, because the frequency/phase response won't depend on perfect-matching phase between these two structures. Precise adjustments are also not required (see **Photo 1**).

Some Analog Methods

First, we will focus our interest in subtractive analog filters, phase linear or not. Bessel filters are known to provide low-pass filters with near-linear phase. When your low-pass Bessel filter is ready, you need to perform a pure delay or an analog approximation of it. The subtraction of the pure delay, or an analog approximation of it, from the near-linear low-pass filter is supposed to give you the high-pass. The problem is the way you need to synthesize a pure delay and how it will match in practice with the low-pass in term of phase synchronization, which is required to keep your frequency response close to theory.

You might use a digital solution to perform the pure delay. That ensures phase linearity, but it involves sampling and quantising the signal through which the high-pass signal must travel. It is also not low constraint, even with today's technology.

You might think about an analog approximation of pure delay, which leads to exponential complexity with respect to the circuit's desired linear-phase bandwidth and involves several active components. In both cases, and especially in the second case, the problem will be to get the low-pass and the delay to have almost exactly the same phase, with better than a 1° similarity. To reach that goal, you'll have to use precise and expensive capacitors.

This is probably the main reason why it is hard to find this type of design. Typically, we find variants of a filter such as the one described in an *Elektor* article, "Active Phase Linear Cross-Over Network," written by Elektor lab/editorial staff (September 1987). This filter, which is not a Linkwitz-Riley filter, is subtractive but it doesn't have linear phase characteristics. The high-pass is obtained by subtracting the low-pass filter from a phasor but neither is linear in phase. The phasor is a unity gain phase copy of the low-pass phase. Further discussion could show how complex it would be to have a phasor with the same phase as the low-pass, supposedly phase-linear, until the

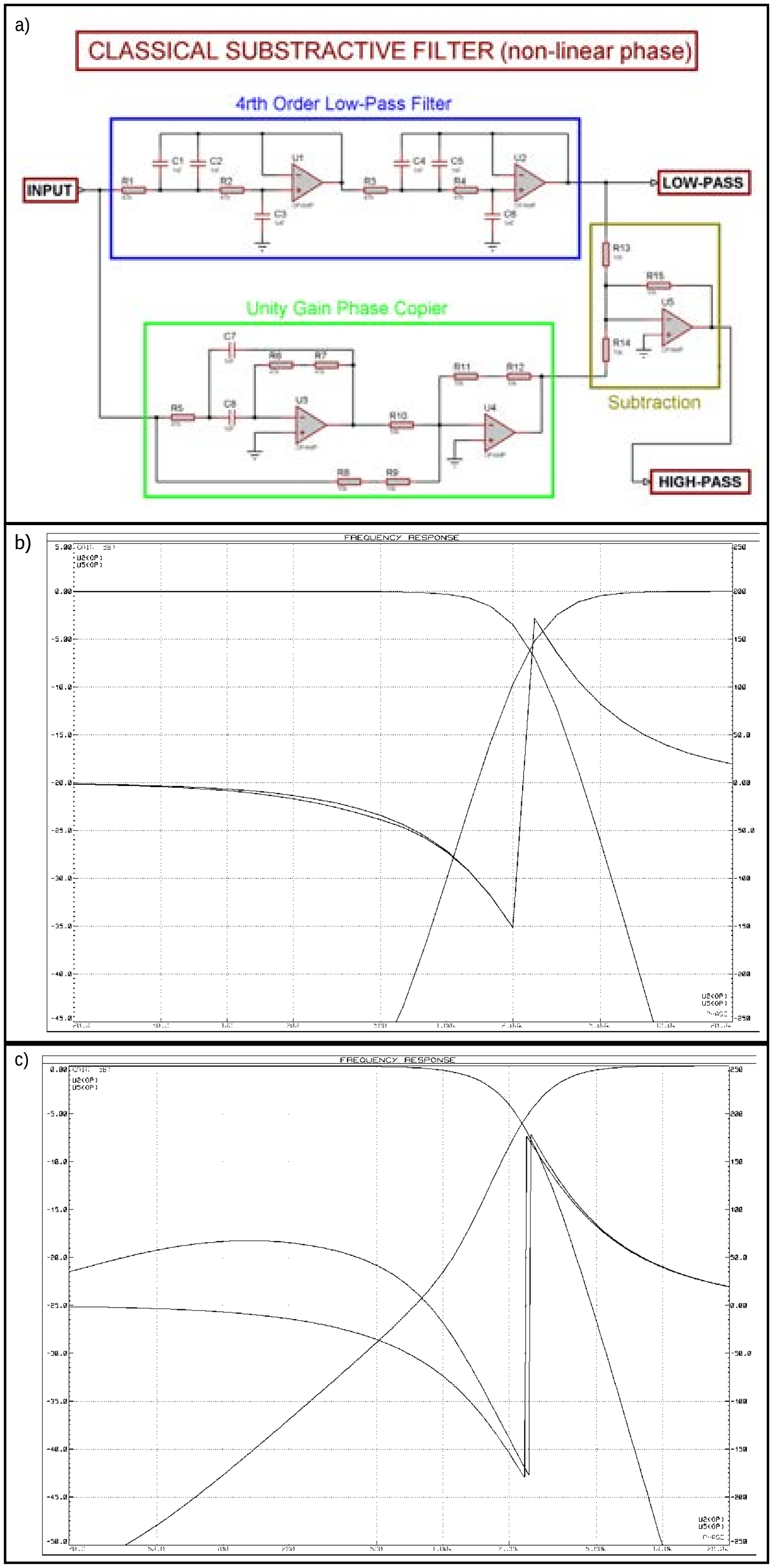


Figure 1a: This schematic details a classic substractive filter (nonlinear phase). Included are two simulations: one uses the exact component values (b) and the other uses 5% precision capacitors (c).

crossover frequency (e.g., 2 kHz) and remaining linear in phase until 20 kHz.

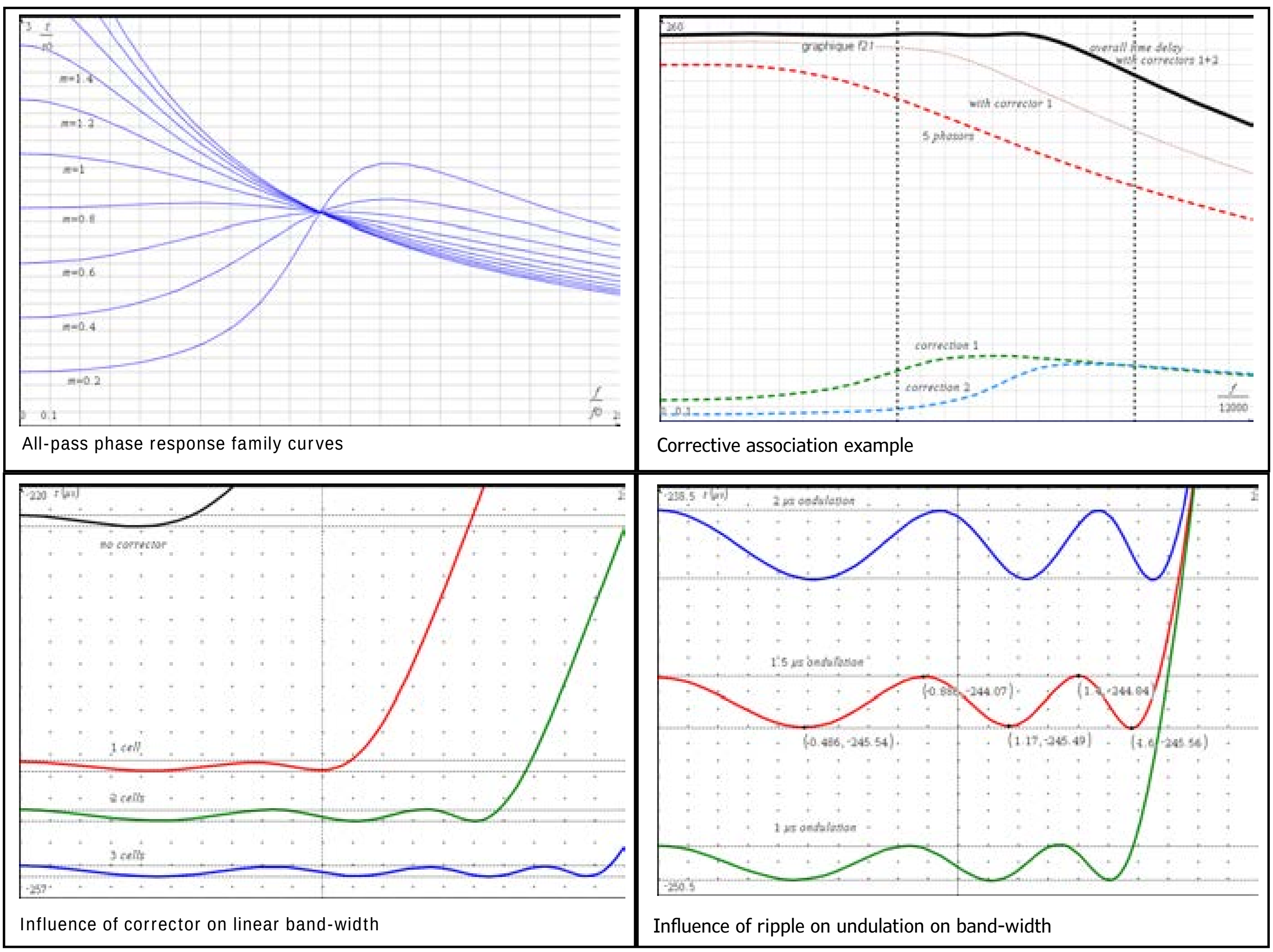
For my project, I don't use a fourth-order Bessel low-pass filter, but I could. As it is fourth order, the phase rotation is about 180° at the cut-off frequency. So, the pure delay approximation will have to provide about 180° phase rotation at 20 kHz. Such a phasor would be at least a 14th order system with exotic component values.

I did a simulation of the circuit described in the *Elektor* article (mentioned earlier) to show how sensitive a simple subtractive filter is to the capacitor's precision (see **Figure 1**). The possible consequences of a 2% deviation of each capacitor are apparent. However, the overall phase (i.e., of the sum of the low-pass and high-pass) is not as greatly affected as the quality of the high-pass.

This is an important issue because the high-pass selectivity must be better than +40 dB/decade. Over its band-pass, a high-frequency driver works with constant acceleration amplitude. Indeed, the signal recorded on a CD is the speaker acceleration equivalence. This is required to link with the acoustic damping of high-frequency drivers, which is both representative of electrical/acoustical transduction and proportional with the squared frequency as long as the wave length is higher than the high-pass radius.

If you want the tweeter to be protected from excessive excursion below its dedicated band-pass—which is necessary to keep it safe and working well—you'll have to cut more than +40 dB/decade. To place the frequency cutoff at a frequency higher than the tweeter's resonance frequency constitutes the beginning of its acoustic band-pass. The 40 dB/decade will only ensure that it will work at constant movement amplitude through the low frequencies.

This means you'll need high-precision capacitors or extensive experimentation to achieve a satisfying result. And we are not even dealing with the linear



All-pass phase response family curves

Corrective association example

Influence of corrector on linear band-width

Influence of ripple on undulation on band-width

Table 1: Phase correction and near-linear phase bandwidth shows the influences of different corrective actions.

phase crossovers yet, just with a "simple" subtractive crossover.

Digital Methods

This is probably the reason why digital filters offer an attractive alternative to what promises to be a time-consuming effort with a less than average probability of success.

So the digital solution could be ideal as long as you are patient enough to wait for the technology to provide a high enough performance at a low cost. While waiting for that to happen, you could try to enhance the quality of your amplifier's power supply, using large capacitors and/or a big transformer, but such audiophile "miracles" seem to have dissuaded many "musicoholics" from throwing more gold coins at their music systems.

Moreover, the digital solution is not as simple as it seems, even in 2015, because you need a linear phase anti-aliasing filter as well as a linear phase smoothing filter in case the source is something other than a CD player and involves harmonics greater than 20 kHz.

Now assume that this is the solution you choose. Basically, the theory how to proceed is widely known. You can synthesize a finite impulse response (FIR) filter whose coefficients are close to a piece of windowed cardinal-sine for the Low-Pass. Here is a generic impulse response (IR) formula generating the frequency response shown in **Table 1** where k is a parameter you have to tune to achieve acceptable rejection:

$$h(z)=\frac{1}{6}\times\left(\frac{z^{-5}}{2}+\sum_{n=0}^{10}\left(\frac{\sin\left(n\frac{\pi}{6}\right)}{\left(n\frac{\pi}{6}\right)}\times e^{-\frac{n^2}{k}}\times z^{-n}\right)\right)$$

You'll recognize a windowed sampled piece of cardinal sine. This IR is shown in **Figure 2**. The green Gaussian curve only shows similarity. This is the IR we'll try to simulate later, hoping for a naturally smoother shape. It comprises only the central part of a cardinal sine and the chosen window makes it very close to a Gaussian IR. The reasons of that choice will be briefly discussed.

You can get a high level of selectivity using a wider part of cardinal sine but it would be a mistake to think it is limitless because it would require a huge number of calculations. What is not more appreciated than an abundance of op-amps. And it would be another mistake to think the final quality would be better because high selectivity involves transient oscillations (Gibbs theorem).

Note that, to be a realistic digital filter, this filter would need more coefficients to get the crossover-frequency as low as 2 kHz. I have previously attempted to realize it with 10 TDA1022 analog delays, but the 70 dB $\left(\frac{S}{N}\right)_{dB}$ and the 0.5% distortion ratio were definitely prohibitive. Despite that, I will show you how such a filter is a good base for a realistic "pseudo-digital" crossover.

Pseudo-Digital Method

For the past 10 years, I have been contemplating and experimenting with a solution that I call the pseudo-digital method. I conducted the experiment like I use to (i.e., like someone living far from a community with limited access to academic literature).

I was not surprised to learn that similar literature existed from around the same time, as I am not the only one who had this idea. Nevertheless, it is often used with low-pass networks, instead of all-pass, as pure delay approximations because they

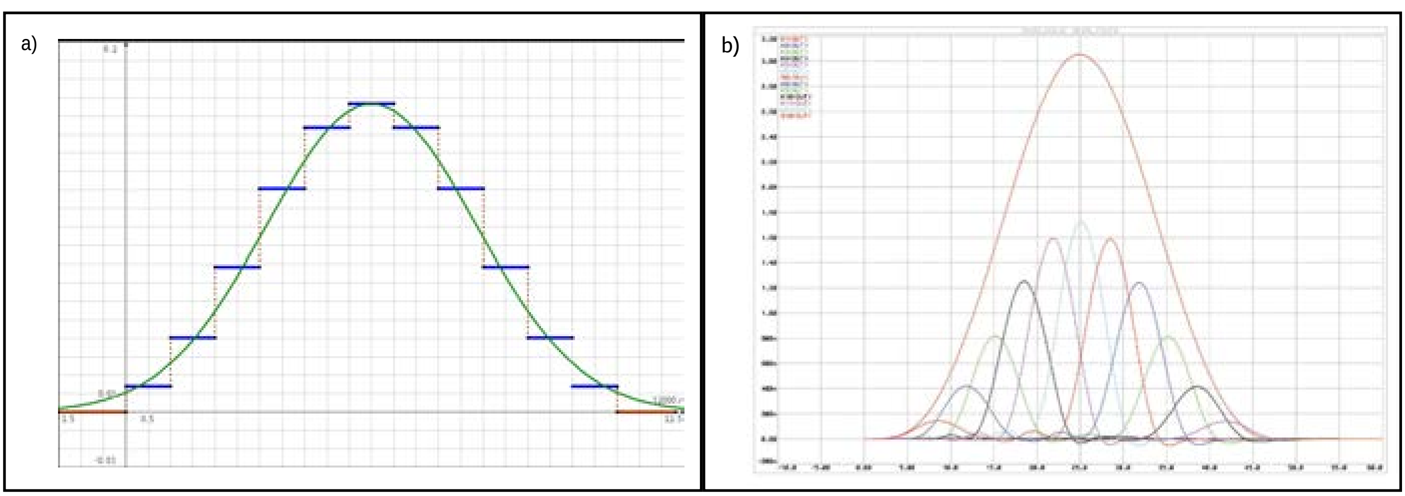


Figure 2: The near Gaussian low-pass impulse response (IR) is shown using a digital IR (a) and a psudeo-digital IR (b).

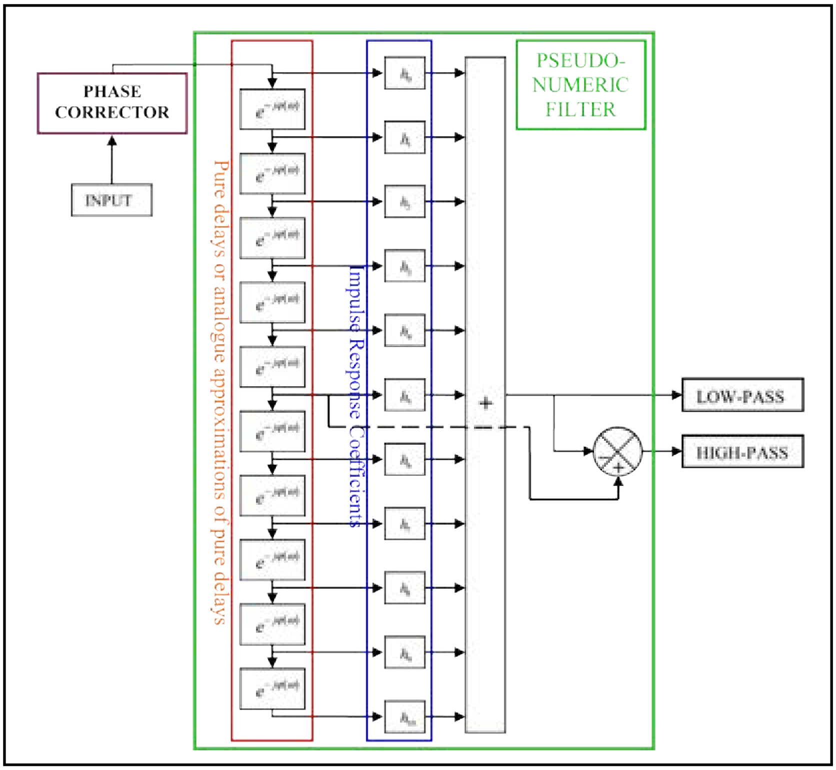


Figure 3: The structure of the pseudo-digital filter is shown in detail.

often only address low-pass filtering.

My first goal was to design easy-to-do high-order audio filters with a potentially tunable crossover frequency. I tried to replace the pure delays provided by temporal sampling using unity gain phasor cells (see **Figure 3**).

I was interested in Padé's approximations of the isochronous transmittance of a pure delay as they can be used with all-pass electronic structures. Pure delay Padé's approximations show:

$$e^{-j\omega T_{delay}} \simeq \frac{1+\sum_{n=1}^{m}\left(\left(-j\omega T_{delay}\right)^{n} \big/ \left(n\times 2^{n}\right)\right)}{1+\sum_{n=1}^{m}\left(\left(j\omega T_{delay}\right)^{n} \big/ \left(n\times 2^{n}\right)\right)}$$

My initial projects used first-order approximations of pure delays and I published three articles in the French magazine *Electronique Pratique.* The first and second articles describe my methodology. The third article describes a "switched capacitor's bass tunable filter" where only the low pass is really affected by switching artifacts.

This methodology is interesting due to its theoretical clarity and the fact that it has low sensitivity to component tolerance. The result is easy, inexpensive, and close to the theory. This robustness, with respect to component tolerance, is due the fact that you don't need perfect phase matching between two physically isolated systems.

I received good feedback about these filters, with the exception of the high number of op-amps and the phase distortion, although noise and distortion were quite acceptable despite the number of op-amps. However, I never discovered if any audio specialists were curious enough to reproduce it and listen to it.

For this project, I have tried to be more reasonable in terms of selectivity to reduce the number of op-amps and use second-order approximations of pure delays to improve phase linearity and better quotient:

$$\frac{selectivity}{Nb\_OAs}$$

Transmittance is:

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

Instead of:

$$\underline{T}(j\omega)=\frac{1-j\frac{\omega}{\omega_0}}{1+j\frac{\omega}{\omega_0}}$$

One op-amp is required in both cases.

Tuning the damping factor leads to 1.5-µs time delay flatness until approximately 0.6 $\times f_0$. This is not enough for this application because the op-amps are too important when it comes to achieving acceptable selectivity.

Note that sampling N times each half-wave of a cardinal sine for the IR results in the crossover frequency being the frequency where phase rotation is:

$$\frac{\pi}{N}$$

per pure delays approximation. Using about m = 0.85 damping factor results in:

$$f_{cross}\simeq\frac{0.93}{6}\times f_0\simeq\frac{f_0}{6.45}$$

So, if phase is linear until 0.6 $\times f_0$ (see **Table 1**), I need:

$$\frac{20}{2\times 0.6}$$

samples of each lobe for a 2-kHz cutoff. Restricting it to the central part, which is twice as wide, I needed at least 30 phasor cells. This is definitely too much.

So, I had to design a phase corrector placed before the series of identical phasors. To do so, I took my graphing calculator and tried to adjust the coefficients. After only a few hours, I obtained a satisfying corrector design by following the example of Chebychev filters with an extrapolation to phase domain (see **Table 1**).

I was close to a theoretical formalization. So, I placed the phase corrector before the filter for it to be effective on both low-pass and high-pass. The overall time-delay equation, assuming that:

$$f_0 \simeq 12000 \simeq \frac{2000}{1.6} \quad \leftrightarrow \quad f_{cross} = 1860Hz$$

$T_{delay} = T_{delay\_phasors} + T_{delay\_corrector\_11} + T_{delay\_corrector\_12}$

with:

$$\begin{cases} T_{delay\_phasors}(\omega) = \dfrac{1}{12000\times\omega}\times\left(10\times\arctan_2\left(1+2\times j\times 0.85865\times\dfrac{\omega}{\omega_0}-\left(\dfrac{\omega}{\omega_0}\right)^2\right)\right) \\ T_{delay\_corrector\_11}(\omega) = \dfrac{1}{12000\times\omega}\times\left(2\times\arctan_2\left(1+2\times j\times 0.265\times\left(\dfrac{\omega}{1.14884\times\omega_0}\right)-\left(\dfrac{\omega}{1.14884\times\omega_0}\right)^2\right)\right) \\ T_{delay\_corrector\_12}(\omega) = \dfrac{1}{12000\times\omega}\times\left(2\times\arctan_2\left(1+2\times j\times 0.1188\times\left(\dfrac{\omega}{1.53987\times\omega_0}\right)-\left(\dfrac{\omega}{1.53987\times\omega_0}\right)^2\right)\right) \end{cases}$$

We get T_{delay} ≈ 240 µs over the linear phase bandwidth.

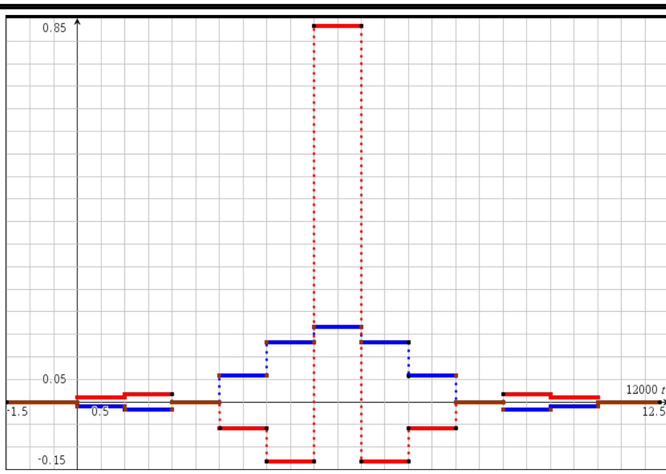
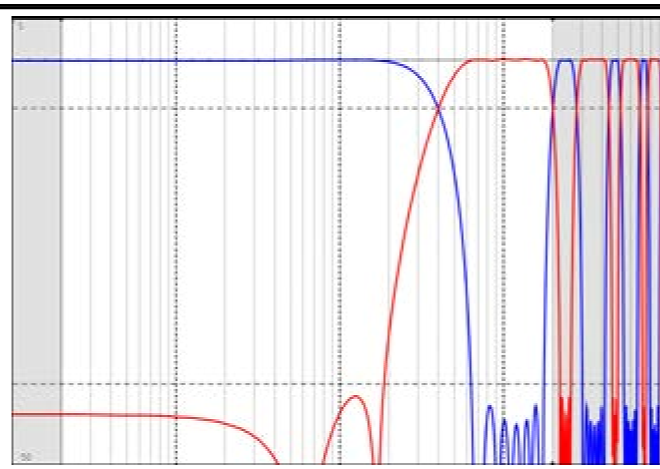
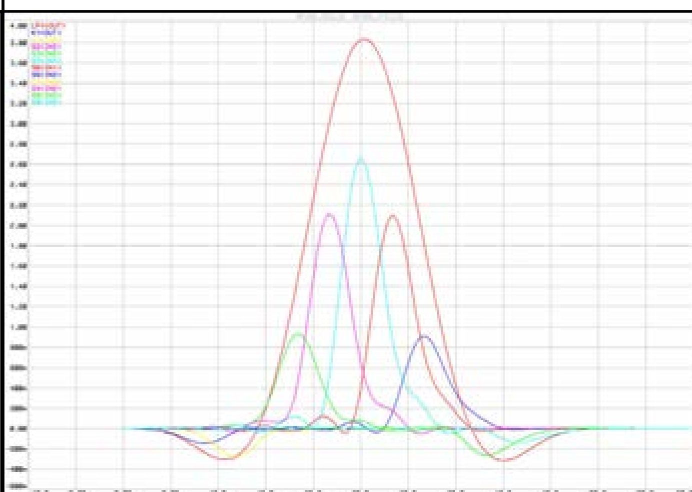
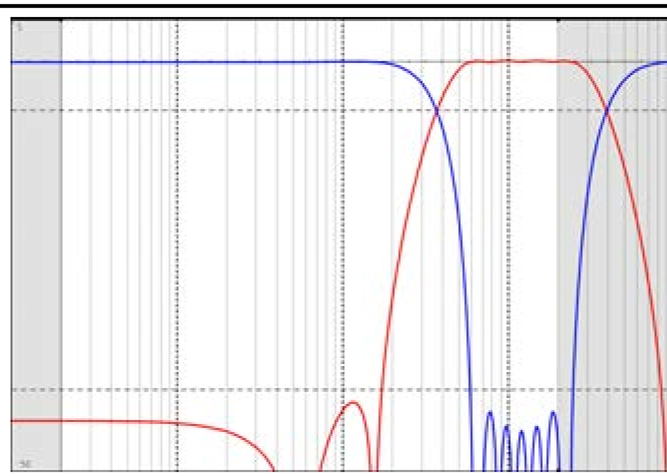
	Impulse Response	Frequency Response
Digital filter		
LP	$20\times\log_{10}\left(\frac{1}{3}\left\lvert 0.5+\sum_{n=1}^{6}\frac{\sin\left(n\times\frac{\pi}{3}\right)}{\left(n\times\frac{\pi}{3}\right)}\times e^{-\frac{n^2}{23}}\times\cos\left(\frac{n\times\pi\times f}{12000}\right)\right\rvert\right)$	$20\times\log_{10}\left(\frac{1}{3}\left\lvert 0.5\times\sum_{n=1}^{6}\frac{\sin\left(n\frac{\pi}{3}\right)}{\left(n\frac{\pi}{3}\right)}\times e^{-\frac{n^2}{23}}\times\cos\left(2n\times angle\left(1+2\times0.85\times i\times\frac{f}{12000}-\frac{f^2}{12000^2}\right)\right)\right\rvert\right)$
Pseudo-digital filter with our approximation		

Table 2: This details the impulse response and frequency response for both the digital filter and the psuedo-digital filter with the upper active octave set at 4 to 8 kHz.

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.

Conclusion

Now that we can use an analog system to simulate the behavior of the digital filter previously described, we can see the effect of the phasor cells phase distortion beyond $0.6 \times f_0$ (while the phase corrector extends overall phase linearity to $1.6 \times f_0$) as a blessing because we get only one aliasing of the frequency response, and this aliasing is rejected far beyond 20 kHz (see **Table 2** and **Table 3**). Note that the response of the digital filter repeats itself every 2 π phasing of pure delays and we have restricted ourselves to 2 π phasing approximations of pure delays.

Next month, I will describe the rest of the

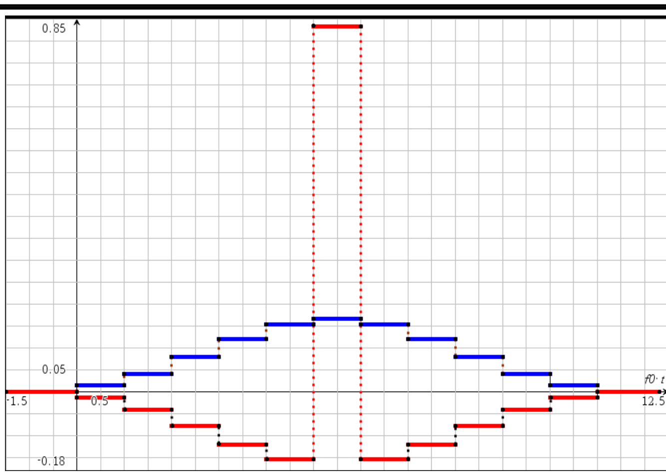
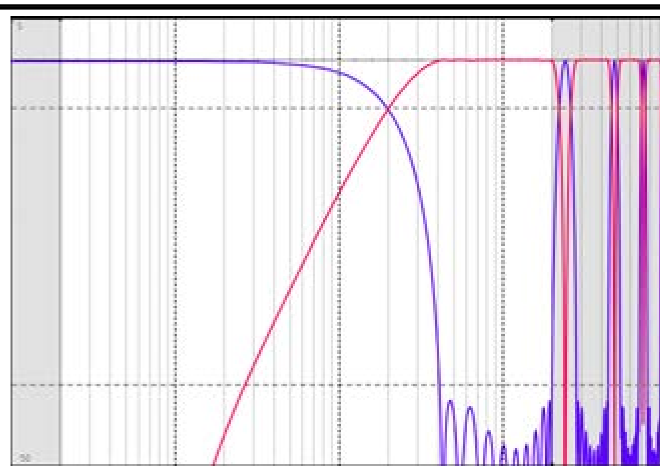
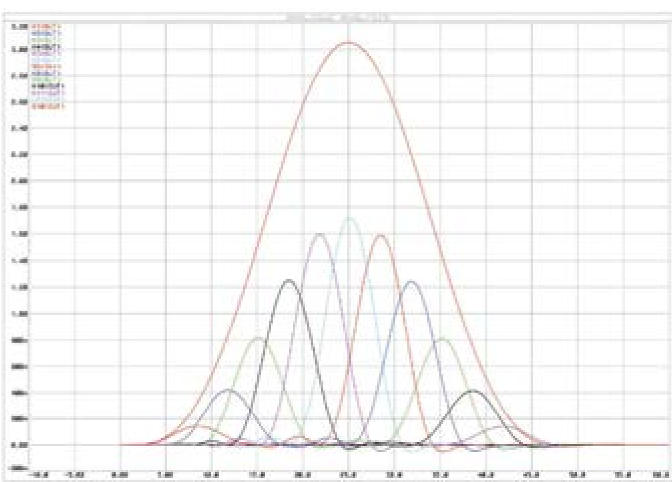
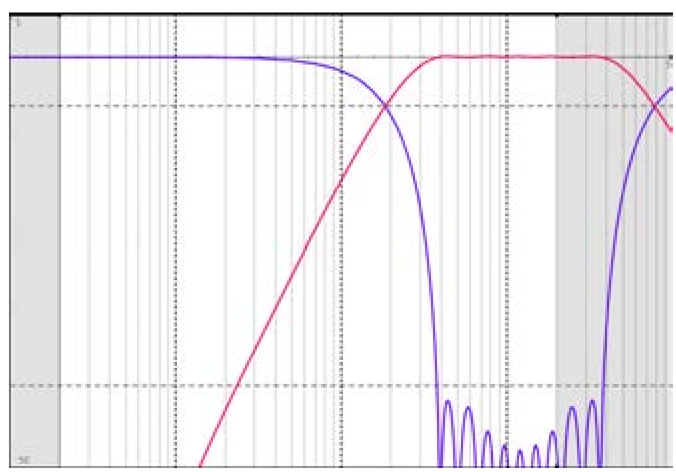
	Impulse Response	Frequency Response
Digital filter		
LP	$20\times\log_{10}\left(\frac{1}{3}\left\|0.5+\sum_{n=1}^{6}\frac{\sin\left(n\times\frac{\pi}{6}\right)}{\left(n\times\frac{\pi}{6}\right)}\times e^{-\frac{n^2}{30}}\times\cos\left(\frac{n\times\pi\times f}{12000}\right)\right\|\right)$	$20\times log_{10}\left(\frac{1}{3}\left\|0.5\times\sum_{n=1}^{6}\frac{\sin\left(n\frac{\pi}{6}\right)}{\left(n\frac{\pi}{6}\right)}\times e^{-\frac{n^2}{30}}\times\cos\left(2n\times angle\left(1+2\times0.85\times i\times\frac{f}{12000}-\frac{f^2}{12000^2}\right)\right)\right\|\right)$
Pseudo-digital filter with our approximation		

Table 3: This shows the impulse response and frequency response for both the digital filter and the psuedo-digital filter with the lower octave set at 2 to 4 kHz.

project, focusing on crossover-frequency tuning, and measurements. I will also detail corrector 2, which is simultaneously performing the phase correction and the 20-kHz anti-phase-distortion filtering. In the meantime, enjoy the filter. Because of the limited article size, I did not provide detailed explanations about the implementation. 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 will also describe the rest of the project with justifications for my technical choices.

My future efforts include enhancing the output voltage swing, limited to more than ±500 mV with a crossover-frequency tuning board, depending on the crossover-frequency value. I will provide precise measurements of noise and distortion next month as well. For now, I am just saying that 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 0.1% but I need more precise measurements to give an exact value.

Author's Note: I would like to offer a special thanks to Stéphane Cnockaert who contacted me to discuss my first filters and gave me probably the best advice when working on my personal designs. He said to use as few op-amps as possible; pseudo-order of the high-pass is better than two; use a near Gaussian impulse response for the low-pass; have optimal overall filtering to avoid transient artifacts.

Also, special thanks to the late André Bustico, who did a simulation of my first filters as soon as I designed the circuit and encouraged me to consider forward-phase linearity.

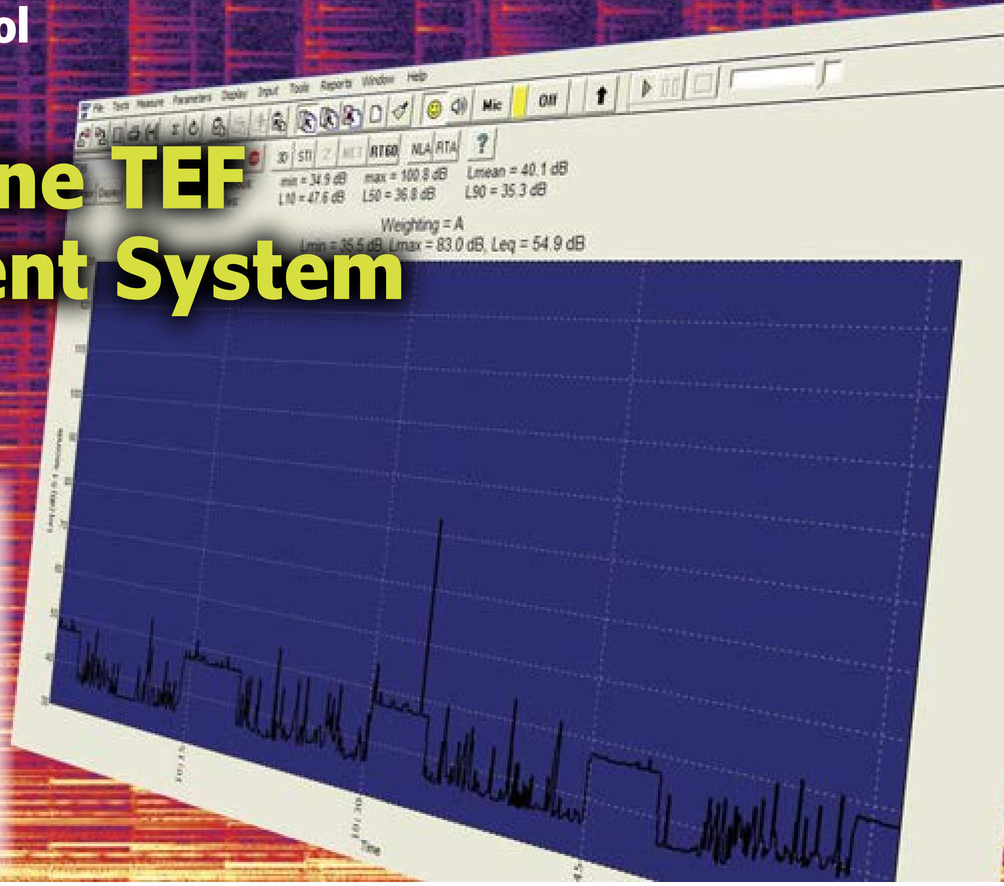
Project Files

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

Resource

"Active Phase Linear Cross-Over Network," *Elektor*, September 1987.

The Gold Line TEF Measurement System



Learn the history of the time delay spectrometry (TDS) measurement concept and the time energy frequency (TEF) analyzer in this sound control article. Then, explore one of its modern-day descendants, the Gold Line TEF measurement system.

By
Richard Honeycutt
(United States)

The Invention of TDS and TEF

The first computerized acoustical measurement systems had their inception in the 1970s, based on the work of the late Richard C. Heyser, who was then employed at the Jet Propulsion Laboratory (JPL). Heyser invented a measurement concept called "Time Delay Spectrometry," or TDS, which enables quasi-anechoic measurements to be made in echoic environments, with high noise immunity. Heyser's original TDS system consisted of numerous pieces of standard test equipment (e.g., oscilloscopes, signal generators, etc.) plus a specialized box of his design, which was dubbed the Time-Energy-Frequency (TEF) Analyzer. This original system is now housed in the Acoustics Department of Columbia College in Chicago, IL, where it fills three equipment racks.

Photo 1: This current TEF 25 kit replaces Richard Heyser's original three racks of equipment.

In 1979, Gerrald Stanley of Crown International led a team of engineers and technical people who developed the first dedicated TDS instrument—the TEF 10. This advanced "measurement computer" included three Z80 microprocessors, math coprocessors, two 5.25" disc drives, a video display, a CP/M operating system, and all the necessary filters, oscillators, preamplifiers, and support systems. The system was housed in one approximately 50-lb self-contained unit.

TEF's Evolution

In 1990, the TEF-20 was introduced. It combined all of the display and data storage functions onto a standard PC. The TEF line has since been acquired by Gold Line, and the current model is the TEF 25. Heyser's original three racks of equipment have been replaced by a small USB preamp, a test microphone, a handful of cables and adapters, and a set of software modules (see **Photo 1**).

The TEF software is collectively called Sound Lab, and is available in separate modules. The software modules that are of the most interest for acoustical measurement are the Noise Level Analysis (NLA) and TDS modules. The RTA module provides real-time analysis functionality for sound-system measurements.

Rather than depend upon a computer sound card of unknown quality, the current TEF system includes the TEF 25 USB I/O device with known premium-quality performance. The TEF 25 is a true dual-channel device with balanced TRS and XLR inputs and XLR outputs, excellent phase matching between channels, 48-V phantom power and a noise floor of –130 dBV. The front panel includes signal-present, clip, and 48-V LEDs for each input, and signal LEDs for each output. The TEF 25 is powered by the USB port. An included wall wart is used to provide power for certain 48-V test microphones. USB drivers are saved in the computer with the installation of the Sound Lab software. The user installs the drivers, following a step-by-step guide provided with the TEF system.

TEF with Sound Lab

A TEF system equipped with Sound Lab plus the TDS, NLA, and RTA modules provides the following measurements:

- Time Response
  - Basic energy-time curve
  - RT60 broadband, full octave, 1.3 octave
  - %ALcons
- Frequency response
  - RTA
  - FFT using TDS to create an anechoic time window for measurements
  - 3-D waterfall plot
- Noise, STI, and RASTI

TEF's Features

To make measurements using the TEF system, the user opens the Sound Lab software to the first screen (see **Photo 2**). The three toolbars across the top are designated as the Main Menu, Direct Access Toolbar, and Measurement Toolbar. In general, any test function can be accessed in more than one way, using these toolbars.

The Test menu (Main Menu toolbar) enables the creation and the execution of custom tests, using the FORTH scripting language. The Measure menu provides one way to execute any measurement for which the necessary software modules are present. The Parameters menu allows the user to define test parameters for different types of tests.

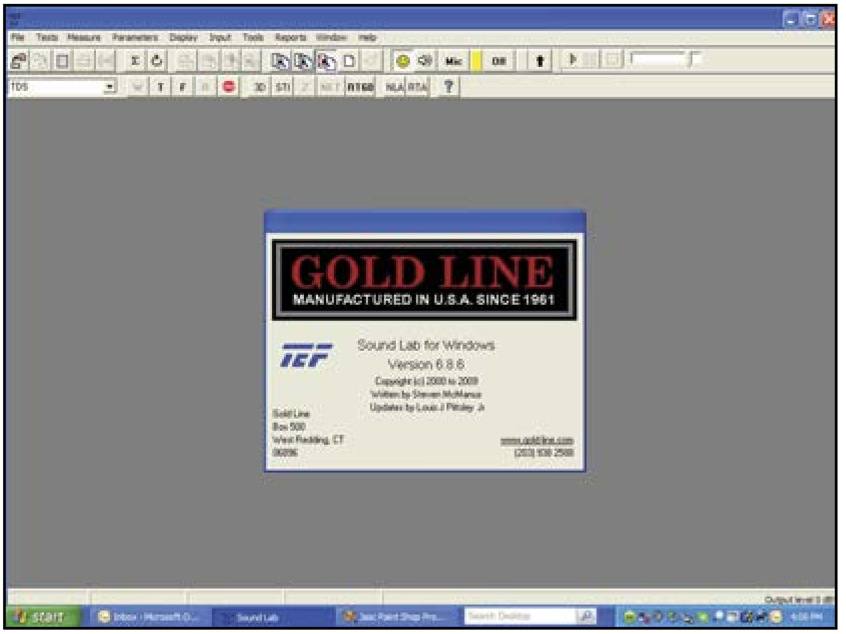


Photo 2: This is the TEF opening screen.

Photo 3 shows the parameters dialog for time-response tests. The Display menu (Main Menu toolbar) provides control of screen configurations. The Tools menu is populated by options depending on the software modules that are loaded. The Reports window allows the user to add a custom image to printed report pages. The Help button accesses an html help file that is loaded to the user's computer during the installation process.

Located to the right of center in the Direct Access Toolbar is a smiley-face indicating that good USB

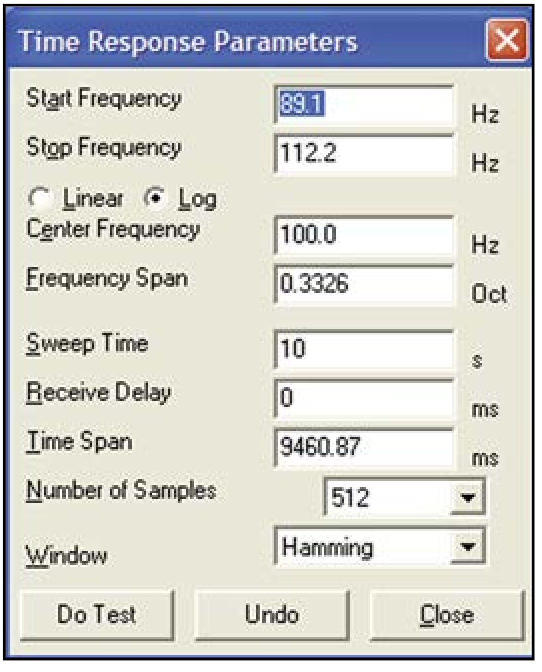


Photo 3: The parameters for time response tests are set via this window, accessible through the Parameters menu.

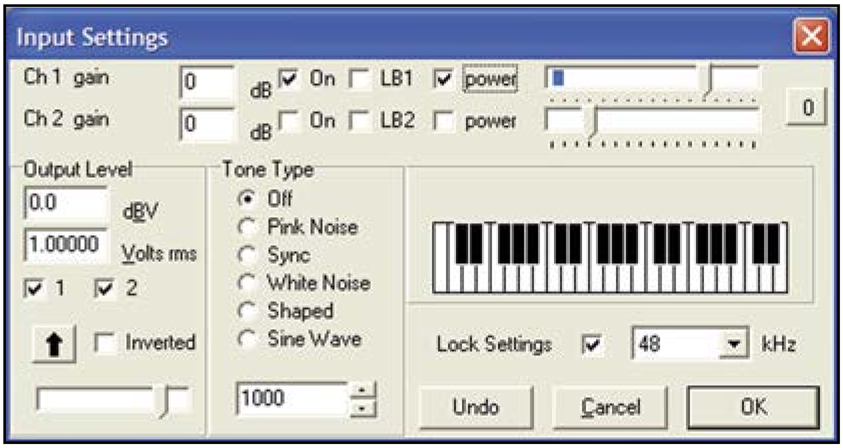


Photo 4: Input selection and output wave parameters are chosen via the input settings dialog.

communication with the TEF box exists. To the right of the smiley-face is a button with a loudspeaker logo, which brings up the input settings dialog box (see **Photo 4**). Via this dialog, the channel 1 or the channel 2 input can be turned on or off, or put in loopback mode. Output level and tone type can be set, and a piano keyboard can be used to select the frequency, using the standard musical pitch scale.

The next four buttons on the Main Menu Toolbar enable selection of inputs for each channel: microphone, internal, or off. To the right of each selection button is a rectangular indicator that turns yellow to indicate that phantom power for that channel is on.

The Measurement Toolbar provides one-click operation of specific tests. These include:

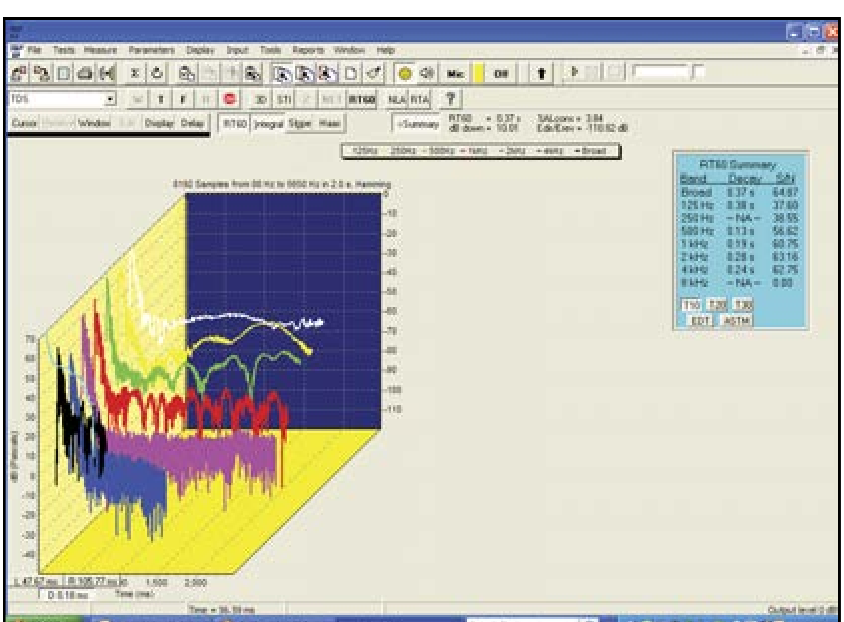


Photo 5: The octave-band RT60 measurement results include much information.

- Time response
- Frequency response
- 3 D (waterfall)
- STI speech intelligibility
- RT60 in full- or 1/3-octave bands
- NLA noise analysis
- RTA functions,
- Additional tests, according to the software modules that are loaded.

To find the RT60 in octave bands with TEF, the user first connects a measurement microphone to the TEF 25's channel 1 input, and a power amplifier and test loudspeaker to an output XLR connector. Then, the user selects Mic by pressing the second button to the right of the smiley-face on the Direct Access Toolbar and presses the loudspeaker button on that same toolbar to bring up the Input Settings dialog. From that dialog, the Ch 1 On and power check-boxes are checked, and the Ch 2 On button is left unchecked. The measurement and post-processing automatically proceed under the control of a predefined script. **Photo 5** shows the results.

The 3-D display at left shows the time response for each octave band, as well as the broadband response. Once the Summary button in the measurement toolbar has been pressed, a data box appears, as shown at the upper right. This data box lists the broadband RT60 and the RT60 for each octave band from 125 Hz to 8 kHz. Also shown is the SNR observed during the measurement. If the SNR is too low to permit a valid RT60 measurement, the software will display N/A rather than state an erroneous value.

In the example measurement shown, no RT60 was found for the 250-Hz and 8-kHz bands, either because of excessive noise in the room or due to insufficient test signal output from the test loudspeaker. The RT60 determined by the T10, T20, T30, or ASTM C-423 method can be displayed, as can the Early Decay Time (EDT). The broadband RT60, %ALcons, and direct-to-reverberant ratio are displayed at the right-hand end of the Measurement Toolbar.

Measuring STI uses a similar procedure, except the STI button on the Measurement Toolbar is used instead of the RT60 button. **Photo 6** shows the results of an STI test. The Transmission Index (TI), early RT (the earliest data point that is –10 dB with respect to the maximum level, used for %ALcons calculations), SNR, and noise are displayed for each octave band. From this data, the STI and the broadband SNR are calculated and displayed, and the STI classification (Poor-Fair-Good-Excellent) is shown.

NLA Measurements

The NLA module of Sound Lab provides the

NEW RELEASES

satori® TW29RN

(Also available with all black faceplate)

Nominal Impedance	: 4 Ω
DC-resistance, R_e	: 3.2 Ω
Resonance frequency, f_s	: 700 Hz
Piston area, S_d	: 9.6 cm²
Sensitivity (2.83 V/1 m)	: 97 dB
V.C. diameter	: 29 mm
Rated power handling	: 80 W

- Non-resonant diaphragm design for minimum high frequency break-up
- Two part aluminum faceplate with integrated mechanical decoupling
- Dual balanced compression chambers for improveddynamics
- Dual copper caps for absolute minimum voice coil inductance and minimum phase shift
- High saturation neodymium motor system with T-shapedpole piece for lower distortion
- Non-reflective cast aluminum chamber with optimized damping for improved dynamics
- Shallow flow optimized magnet structure for optimum coupling to rear chamber
- CCAW voice coil for low moving mass
- Long life silver lead wires
- Low resonance frequency for extended range

SB ACOUSTICS SB29SDC

Nominal Impedance	: 4 Ω
DC-resistance, R_e	: 3.2 Ω
Resonance frequency, f_s	: 580 Hz
Piston area, S_d	: 9.6 cm²
Sensitivity (2.83 V/1 m)	: 93 dB
V.C. diameter	: 29 mm
Rated power handling	: 100 W

- Fine weave soft fabric dome for smooth frequency response
- Large surround area for better high frequency dispersion
- Dual balanced compression chamber for improved dynamics
- Copper cap for reduced voice coil inductance and minimum phase shift
- Saturation controlled motor system for low distortion
- Non-reflective rear chamber with optimized damping for improved dynamics
- Flow optimized vented pole piece for optimum coupling to rear chamber
- CCAW voice coil for low moving mass
- Long life silver lead wires
- Low resonance frequency

SB ACOUSTICS 6" SB17CRC35-8

(Also available in 4 Ω)

Nominal Impedance	: 8 Ω
DC-resistance, R_e	: 5.7 Ω
Resonance frequency, f_s	: 29 Hz
Total Q-factor, Q_{ts}	: 0.35
Equivalent volume, V_{as}	: 40 liters
Piston area, S_d	: 118 cm²
Sensitivity (2.83 V/1 m)	: 87 dB
V.C. diameter	: 35.5 mm
Linear excursion, X_{max}	: ± 5.5 mm
Rated power handling	: 60 W

- Rohacell®/Carbon fibre sandwich cone for optimized stiffness/damping ratio
- Vented cast aluminum chassis for optimum strength and low compression
- Low damping rubber surround for improved transient response
- Non-conducting fibre glass voice coil former for minimum damping
- Extended copper sleeve on pole piece for low inductance and low distortion
- CCAW voice coil for reduced moving mass
- Long life silver lead wires
- Vented pole piece for reduced compression

SB ACOUSTICS 8" SB23MFCL45-4

(Also available in 8 Ω)

Nominal Impedance	: 4 Ω
DC-resistance, R_e	: 3.3 Ω
Resonance frequency, f_s	: 27 Hz
Total Q-factor, Q_{ts}	: 0.36
Equivalent volume, V_{as}	: 39.2 liters
Piston area, S_d	: 208 cm²
Sensitivity (2.83 V/1 m)	: 87.5 dB
V.C. diameter	: 45.5 mm
Linear excursion, X_{max}	: ± 11.7 mm
Rated power handling	: 120 W

- Long stroke design
- Linear high stability suspension
- Low damping rubber surround
- Long copper voice coil
- Non-conducting reinforced fibre glass coil former
- Very stiff mineral filled Polypropylene cone
- Vented cast aluminium chassis
- Large vented motor system
- Long life silver lead wires
- Optimized for smaller enclosures

further information :

- www.sbacoustics.com
- www.sinarbajaelectric.com
- OEM contact : mkt@sinarbajaelectric.com

www.facebook.com/sbacoustics

Photo 6: TEF displays the results of a speech transmission index (STI) measurement in these panels.

Octave	TI	Early RT60	S/N ratio	Noise
125	0.87	0.3	11.1	0
250	0.8	0.38	8.9	0
500	0.86	0.27	10.7	0
1000	0.82	0.32	9.6	0
2000	0.85	0.28	10.4	0
4000	0.83	0.31	9.9	0
8000	0.91	0.22	12.4	0

STI	0.84
Early RT60	0.3
S/N ratio	10.44
Evaluation	Excellent

ability to analyze noise over a specified time period. NLA can be set to run for an interval from one minute to 24 hours. Often this type of measurement is used for community noise measurements or to document compliance with Occupational Safety and Health Administration (OSHA) noise requirements in the work place. NLA is also useful when designing and installing sound systems in places where noise is the primary inhibitor to intelligibility (e.g., gymnasia, railway stations, etc.).

The latest NLA release (version 6.8.14 or greater) is capable of two-channel measurements. Also, a new ability has been added to merge several consecutive NLA files into a summary file that contains all the information of the individual files

An enhanced version of NLA is available that permits import of noise files in the .wav format.

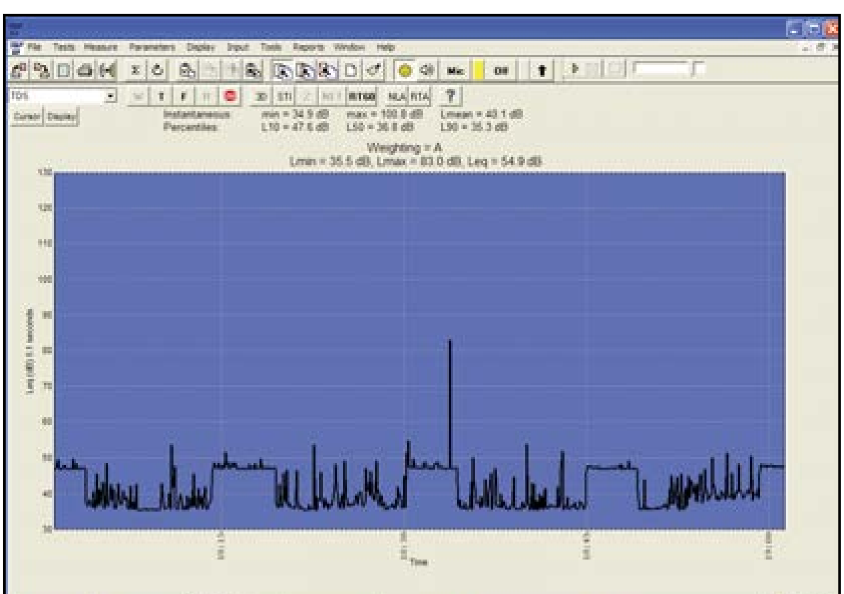


Photo 7: A noise level analysis (NLA) measurement shows the noise level vs. time, plus statistical data resulting from analysis of the noise.

The post-process Edit function gives the user the ability to edit an NLA measurement after it has been made by selecting points where extraneous noise has occurred and muting those points so that they do not interfere with the measurement. The muted data remains but it is skipped when calculating any statistical information about the measurement.

The Import feature enables you to load a .wav file into Sound Lab and convert it to an NLA measurement. The Output feature enables you to save an audio recording from the microphone at the time of the measurement, which is synced to the NLA measurement. The audio streaming function makes it easier to select a time span of a finished NLA measurement and play back the audio associated with that selected time span through the computer speakers or a headset. The Listen function extends this capability to allow you to hear the audio from the microphone during a live NLA measurement session.

Photo 7 shows a completed NLA measurement. At the bottom is a trace showing the noise level in an office over a one-hour period, with the actual clock time shown on the horizontal axis. The intervals during which the HVAC system operated clearly shows, as does one momentary spike during which an object was dropped onto the floor.

The information at the top provides the instantaneous minimum, maximum, and mean noise levels, as well as the levels that are exceeded 10%, 50%, and 90% of the time, respectively. The frequency weighting is also displayed. The data analysis can be specified by use of the NLA parameters dialog (see **Photo 8**).

Selection of the Integration time controls how the NLA data is calculated. This can be changed at any time before, during, or after the measurement has completed. The following choices are provided:

- Leq is the equivalent continuous sound level. The steady continuous level yields the same total sound energy over a given period of time as the varying measured levels.
- Lden is a 24-hour Leq, except 5 dB is added to all levels measured between 7:00 PM and 10:00 PM and 10 dB is added to all levels measured between 10:00 PM and 7:00 AM to account for the need for more quiet during sleep hours.
- Ldn is a 24-hour Leq, except 10 dB is added to all levels measured between 10:00 PM and 7:00 AM.
- Dose is a rating of the time-weighted acoustic exposure level that a person receives over a period of time, usually 8 hours.
- Linear/Time displays data in time vs. mPascals instead of time vs. decibels.

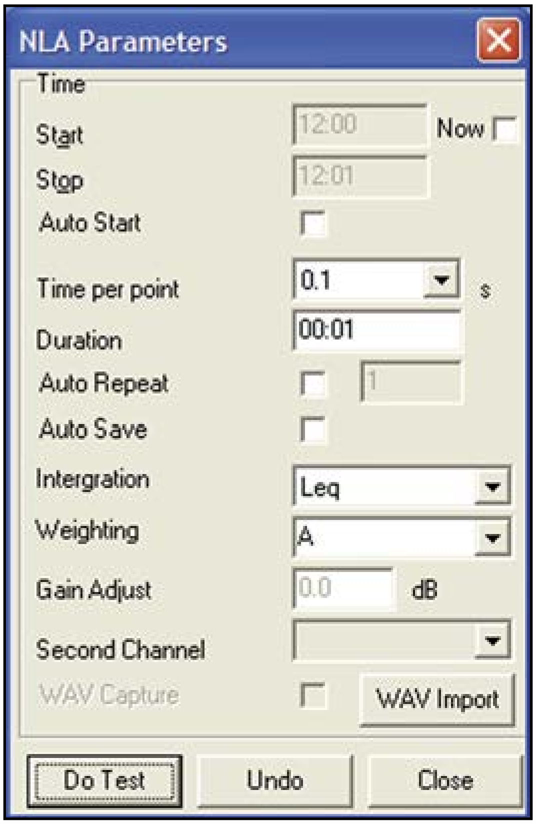


Photo 8: Parameters for noise analysis are set using this dialog box.

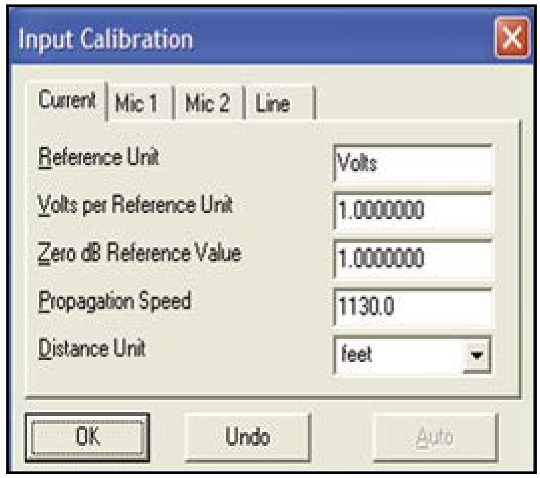


Photo 9: This screen enables you to enter microphone calibration information into Sound Lab.

- Register is a chart of sound pressure levels (SPL) encountered during a measurement vs. the percentage time that SPL was seen.
- Histogram is a count of the number of times an SPL level was seen during a measurement.

Microphone Calibration Parameters

To make accurate measurements using any system, microphone calibration parameters must be used during the calculations. These are entered in TEF Sound Lab by means of the Input Calibration dialog box (see **Photo 9**). An auto-calibration function is also provided, requiring the use of an external acoustical calibrator to excite the microphone. TEF04 systems are available as individual components or as kits. The standard TEF test microphone is manufactured in the US by Gold Line. For more information, visit www.gold-line.com/tef/t-tefpre.htm.

Epiphany Leads to Innovative Audio Startup

An Interview with Patrick Donohue, CEO and Founder of HearNotes

Patrick Donohue founded HearNotes in an effort to create truly WireFree earbuds.

This is the current HearNotes company and product logo.

By
Shannon Becker
(United States)

SHANNON BECKER: Tell us about yourself.

PATRICK DONOHUE: I'm originally from the Midwest, and although my professional background has primarily revolved around banking and finance, my interest in audio technology stems from my experience as a musician and a classically trained pianist and composer, as well as from my experience as a national collegiate athlete. [Patrick is also a two-time Kentucky state tennis champion.]

I hold a bachelor's degree from Miami University in Oxford, OH, and began my career in real estate finance, ultimately making my way to JPMorgan Chase, where I worked for about five years. Then, in 2012, I was promoted to San Francisco, CA, to run one of the firm's most prominent banking centers in the Financial District, leading a team of 14 employees and overseeing a $50 million balance sheet.

SHANNON: How did you get interested in audio technology?

PATRICK: A little-known fact about my background as a musician is that I've also been writing and recording music for nearly two decades. My interest in audio technology initially piqued around age 14 when I produced my first album with a 16-track MIDI recorder. Although archaic looking back, it was quite leading-edge at the time, and I suppose that innate curiosity has just stuck with me ever since.

SHANNON: Tell us about your company HearNotes (www.hearnotes.com).

PATRICK: HearNotes is a San Francisco-based startup on a mission to pioneer the evolution of portable audio technology and advance the mobile music listening experience. The vision was first conceptualized on December 14, 2013 in burst of profound intuition, triggered as a result of a great deal of research I'd already been conducting. It was a defining moment of clarity around what seemed my inevitable path towards driving a quantum leap in audio technology innovation. I walked away from my post at JPMorgan Chase and founded HearNotes on January 7, 2014.

SHANNON: How did you learn of Kleer technology and why did you decide to use it in your headphones?

PATRICK: The industry research that followed my original "epiphany" admittedly felt a bit like trying to find a needle in a haystack, without the guarantee that the needle even existed in the first place. But that exhaustive research led to the "discovery" upon which HearNotes is founded—Kleer technology.

Kleer is quite simply one of the industry's best-kept secrets for permanently solving the well-known limitations of Bluetooth. It is the missing evolutionary link toward building a truly WireFree earbud solution, and it represents a quintessential opportunity to advance an overly commoditized industry that is in dire need of innovation.

SHANNON: Tell us about your WireFree Earbuds concept. What makes the concept so unique?

The HearNotes product family includes two WireFree earbuds, a transmitter, and an inductive wireless charging pad that doubles as a carrying case.

PATRICK: HearNotes is a truly wirefree mobile music solution, which means there is no wire from ear to device or from ear to ear, as there is with other earbuds available today. These other so-called

HearNotes premium WireFree earbuds are powered by Kleer wireless transmission technology from Microchip.

"wireless" earbuds and headphones are based on Bluetooth technology and, therefore, have inherent limitations. Specifically, Bluetooth was never designed for music listening, so utilizing Bluetooth on our mobile devices is akin to surfing the Internet with a dial-up modem. And Bluetooth cannot support stereo audio output unless there's a wire (not wireless) that connects the two earbuds!

HearNotes is unique in its ability to deliver quality and convenience—pristine, lossless, uncompressed, high-fidelity audio via the Kleer technology. It is a final departure from those silly, antiquated wires that had us tangled up for so long. HearNotes is the really truly wireless audio solution the market has been awaiting.

SHANNON: Were there any challenges involved with the product design?

PATRICK: Yes, and I would say that product design represented the fundamental challenge. We first needed to recruit engineers that truly understood the nuances of Kleer technology and how to properly integrate it. Second, given the commoditized nature of the industry, it wasn't like we had a playbook to reference, and previous design attempts were minimal to non-existent.

The question was never if we could build HearNotes, but rather how we would design them to be sleek and cool enough to facilitate rapid consumer adoption. Thankfully, we have an amazing engineering team and early feedback on design continues to be very positive.

SHANNON: You left your product's fate in the hands of the public by posting it on Fundable (www.fundable.com). Why did you choose the crowdfunding route and what kind of response did anticipate?

PATRICK: Fundable, and the crowdfunding route in general, continues as one element to our strategy. Late last year, HearNotes was approached by Fundable to participate in the Staples Crowd2Shelf Contest because it was suggested that our product would not only be a strong contender, but it would also be considered for a rare opportunity to partner with Staples. We were named a finalist, and shortly thereafter officially debuted at the 2015 International Consumer Electronics Show (CES) with tremendous public reception and named Best of Show.

Thankfully, our unveiling was blessed with tremendously positive industry, retail, and consumer reception and anticipation, one measure of which is the more than 70 editorials worldwide and an estimated 500 million digital impressions. People are talking about HearNotes, and we can't wait to give people the opportunity to experience the true joy and freedom of The WireFree Solution that wireless music should be.

SHANNON: Was this your company's first product? If not, what other products do you offer?

PATRICK: HearNotes: The Universal Edition represents the company's debut generation. There are plans in the works to develop an array of WireFree product offerings. Stay tuned.

SHANNON: What's next for HearNotes?

PATRICK: We launched the "We Hear You" Kickstarter campaign on May 4, 2015 to support the HearNotes effort to redefine mobile music listening and put the customer's experience right at the center of our mission, exactly where it belongs. We are super excited about Kickstarter's vast worldwide community of early adopters who are very eager to support the type of innovation present in HearNotes, and be among the first who get in on the ground floor. We anticipate a continued positive response!

Any of your readers who want to be an early adopter for HearNotes, should please check us out on Kickstarter. For more information about HearNotes, visit www.hearnotes.com, Kickstarter, Gethearnotes Facebook, Gethearnotes Twitter, and HearNotes on YouTube.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• Over 300 Exhibitors from Across the Globe
• Over 150 private LIVE DEMO studios (You can't experience this in big box stores!)
• Affordable to Extravagant: Headphones ~ Speakers ~ DAC's ~ Amplifiers ~ Accessories
• LIVE Entertainment Daily! Featuring: Antonia Bennett (Tony Bennett's Daughter), Mike Garson, Fiona Joy, AND MORE!
WINE TASTING ~ CIGARS ~ GOURMET FOOD TRUCKS
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*Background photo courtesy of Hotel Irvine

Construct a PIC-Based Audio DDS Device

I have previously built two direct digital synthesis (DDS) systems using a Rabbit microprocessor programmed in C. I built a PIC-based frequency counter quite a few years ago but have not looked at PICs for about 15 years. So, this time I wanted to see whether or not I could build one using a PIC and still include several useful features.

Photo 1: The main menu selections are located on the front of the unit.

By
Larry Cicchinelli
(United States)

This PC-based audio DDS device contains several convenient features (see the Features List). **Photo 1** shows the front of the unit displaying the main menu selections. The three controls are, from left to right: the amplitude, the Quadrature Encoder, and the four-position joystick. Only one of the LEDs is currently in use and it indicates that the program is running. The remaining two LEDs are for future expansion, although one of them may be set to indicate the marker frequency is being output.

Features List

- Sine wave only—frequency selectable in 1-Hz steps from 1 Hz to 100 kHz with a Direct Digital Synthesis (DDS) resolution of about 0.1 Hz
- Four line LCD
- Quadrature encoder for changing the frequency
- Four-position joystick for navigating the menu and changing the frequency
- Three variations of sine wave output
  - Simple sine wave – one frequency, continuous output
  - Sine wave sweep
    - Linear sweep: user selects step size
    - Octave sweep: user selects full octave or fractional values of 1/2, 1/3, or 1/4
    - User selects lower and upper frequencies, step duration, one marker frequency may be set
  - Frequency list of up to 32 discrete frequencies—the user enters the frequencies and duration
- User can store all entries and select the mode in which the system starts
  - Menu
  - Single frequency
  - Sweep
  - Frequency list
- The amplitude is adjustable via an analog potentiometer and can be calibrated for 0 dBm (into 500 or 600 Ω) full scale
- Fits in a 5″ × 3.15″ × 1.4″ box, making it easily portable (this size box does not have room for batteries)
- Can be powered by 3 AA cells—alkaline battery life should be more than 40 hours if neither the LEDs nor the backlight are used
- Uses a PIC18F25K20 microcontroller and an AD9834 DDS device

The Menus

Navigating the menus is fairly straightforward. The joystick is used to move the cursor to the desired selection. To select the item, simply press in on the Quadrature Encoder (QE). Pressing in on the QE is handled as Enter. The current program does not make use of the joystick's Enter switch. I found this a little bit difficult to operate without also activating one of the direction switches.

The only unusual operation is that if the cursor is on the first entry of the menu, as it is shown in **Photo 1**, and the joystick is moved to the left, you will exit the current menu and the "parent" menu will be displayed. The cursor will be placed on the menu item that had been previously selected.

On any of the menu items that enable you to modify the frequency, the unit's output frequency will follow the changes as you make them with the QE. You can also use the joystick to modify the frequency but the change will not occur until you press Enter. The joystick can be used to move the cursor to different digits of the frequency. Both the QE and the joystick will update the frequency based on the digit selected. For instance, if the joystick is used to move the cursor to the hundreds digit, the frequency will be modified in 100-Hz steps.

The menu system is hierarchical but most entries do not have any sub menus other than entering the frequency value (see the Menu Selections list for the options).

Both the Sweep (fixed step size only) and List selections have additional manual stepping features. If at any time during the Sweep or List output you press Enter, the output frequency will stop advancing and the program will wait for another switch input. Pressing Enter again will cause the output to continue from where it was paused. Moving the joystick to the right will cause the next higher frequency to be output. Moving the joystick to the left will cause the previous (lower) frequency to be output. The Sweep/List operation may be terminated by activating any joystick position while the normal frequency advancing is in progress.

Hardware

Figure 1 shows the connections to the PIC. One of the most important design criteria, which must be decided at the beginning of any microprocessor project, is to assign each processor pin to its required function. It is also a good idea to keep the signals grouped by function, as much as possible, utilizing the features of PIC's different I/O pins. I put the QE outputs on RB0 and RB1 to use the RB0's external interrupt capability. I had to use RC3–RC6, although RC4 is not needed, as the serial peripheral interface (SPI) port for communicating with the DDS.

I also wanted to keep the debug pins (RB6 and RB7) dedicated to that function. That left only the Port A pins for use as the LCD interface. I decided to use the upper four bits for two reasons: The LCD uses bits 4–7 for its 4-bit interface and the sequence for sending a byte to the LCD via its 4-bit interface sends the upper four bits first. You can see from **Figure 1** that RC4 is the processor's only unused pin.

The PIC I selected, the 18F25K20, is a 3.3-V part, which I chose because I happened to have a 3.3-V LCD in my parts bin. It is also in the same family as the processor in the PICkit3. If you would rather use 5-V parts, there is an equivalent PIC, the 18F25K22, which should work fine on the PCB. The code should not have to change based on what I have seen in the 18F25K22's datasheet.

The parts required for a 5-V system are listed as options on the bill of materials (BOM) found in the Supplementary Materials section of the *audioXpress* website. The remaining circuits will work with a 5-V supply. I used a 5-V oscillator, which works just fine with 3.3 V. However, the PCB is designed to handle surface-mount device (SMD) oscillators as well as the four-pin DIP oscillator I used. There are two SMD oscillators listed as options in the BOM, one is 5 V the other is 3.3 V.

Menu Selection List

- **Single frequency**
  - Frequency
- **Save**
  - Main menu
  - Single frequency
  - Sweep
  - List
- **Sweep**
  - Low frequency
  - High frequency
  - Step size
    - Fixed frequency
    - Octave
    - Octave/2
    - Octave/3
    - Octave/4
  - Step duration
  - Marker frequency
  - Go
- **List**
  - Count
  - Duration
  - Frequencies
  - Go

Initially, I did not have a reset circuit for the master clear reset (MCLR) pin of the PIC. It worked fine when using the debugger. However, when I attempted to use it stand-alone, it would not always properly start. I then implemented the MCLR circuit as recommended in the processor manual, which then enabled the correct operation. When operating the generator without the debugger, a jumper plug must be installed between pins 1 and 2 of H105 to connect the reset circuit to the MCLR.

If you want to customize the program, you will need to remove the jumper to connect the debugger cable to pins 2–7. The pinout of H105 (pins 2–7) match that of the PICkit-3. I made an extension adapter for my PICkit-3 that enabled me to plug into H105 as well as into a solderless breadboard. You may want to make your adapter with a seven-pin connector, with pin 1 blank, to help ensure you do not plug it in wrong!

H101 is a right angle header. It should be mounted such that its pins face the board's center and be over the regulator IC. The reason for this is to keep the unit as thin as possible. If the header was vertically mounted, its mating connector would add about 0.5" to the unit's height.

Looking carefully at **Figure 1**, you will notice that

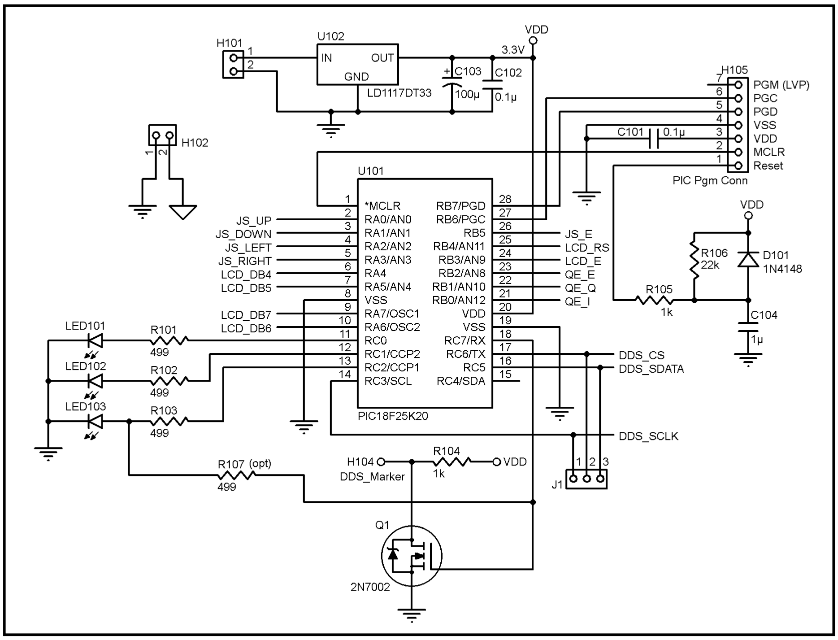


Figure 1: All the connections to PIC are detailed.

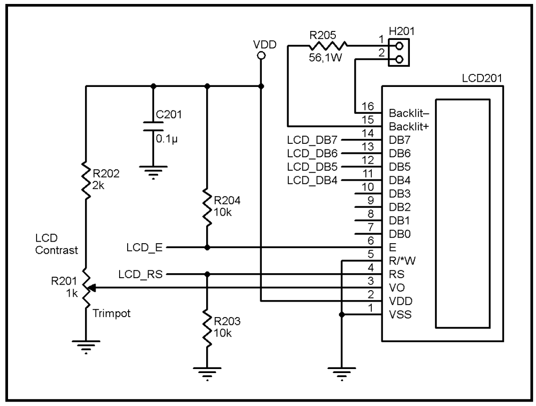


Figure 2: The LCD circuit is pretty straightforward.

About the Author

J. Lawrence Cicchinelli, of Camdenton, MO, received his BSEE from Drexel University Institute of Technology in 1969 and his MSES from Pennsylvania State University in 1981. He worked for Ford Motor Company from 1967 to 2000, including 30 years designing and building test equipment for automotive electronics. He was the technical support manager for Digi International, Rabbit Brand, from 2000 to 2012 and wrote the book *Assembly Language Essentials* published by Circuit Cellar in 2011.

there are two grounds—an analog and a digital. It is probably not necessary for this circuit but it should be considered when designing mixed signal circuits. H102 is not a physical header—it is there just for its pads, which are used to connect the two grounds together. You must install a jumper wire between the two or the circuit will not properly work.

Neither H103 nor H104 need to be installed. The pads for H103 may be used to view the SPI signals for U302—the DDS chip. The H104 location is the only provision on the board for the Marker output. There is also an optional resistor that enables LED103 to turn on whenever the Marker frequency is being output. If you want this option, you should install R107 instead of R103.

If you plan to primarily use battery power, you might want to eliminate the LEDs, including the LCD's LED backlight to save the batteries. If you do want the LEDs, you might want to search for more efficient ones that will not require a lot of current. There are some high-efficiency T-1 LEDs available that are reasonably bright with only 2 mA. If you do use lower current LEDs, you will want to change the values for R101–R103. They currently supply about 6 mA to each LED. The circuit board is designed for LEDs with 0.1" (2.54 mm) lead spacing.

The LCD circuit is quite straightforward (see **Figure 2**). R201 is used to adjust the contrast. It can usually be eliminated by simply connecting its center to ground. H201 is used to connect the LED backlight to the power source. I implemented separate connections for the backlight since my unit draws about 150 mA and I did not want that current flowing through the PCB ground. However, the newer units, specified in the BOM, draw considerably less: 15 mA.

Since the current is less, you may want to simply install a jumper from the LCD end of R205 directly to the output of the regulator. If you still want an external resistor then its value may be different, depending on the main power supply you use. The LCD has a built-in resistor to limit the current to the LEDs based on the normal LCD power supply.

You can calculate the value for R205 as follows:

$$R205 = (V_{POWER} - V_{DD})/I_{LED},$$

if you want full brightness. On my system, I decided to use 100 mA for the LEDs and a 9-V wall wart to power the unit so my resistor is 56 Ω: (9-3.3)/0.1.

The DDS circuit is based on the AD9834 (see **Figure 3**). I have used this circuit two previous times so I already knew how to program it. It uses a 28-bit accumulator so I would liked to have used an oscillator frequency of 26.8435456 MHz ($2^{28}/10$) to achieve 0.1-Hz resolution simply by multiplying the desired frequency in hertz by 10. That would have given me the value to directly program into the frequency registers. However, getting an oscillator of that frequency requires a custom order, which is quite a bit more expensive than I wanted. I settled on a standard 25-MHz oscillator, which is very inexpensive, but it forced me to do some creative

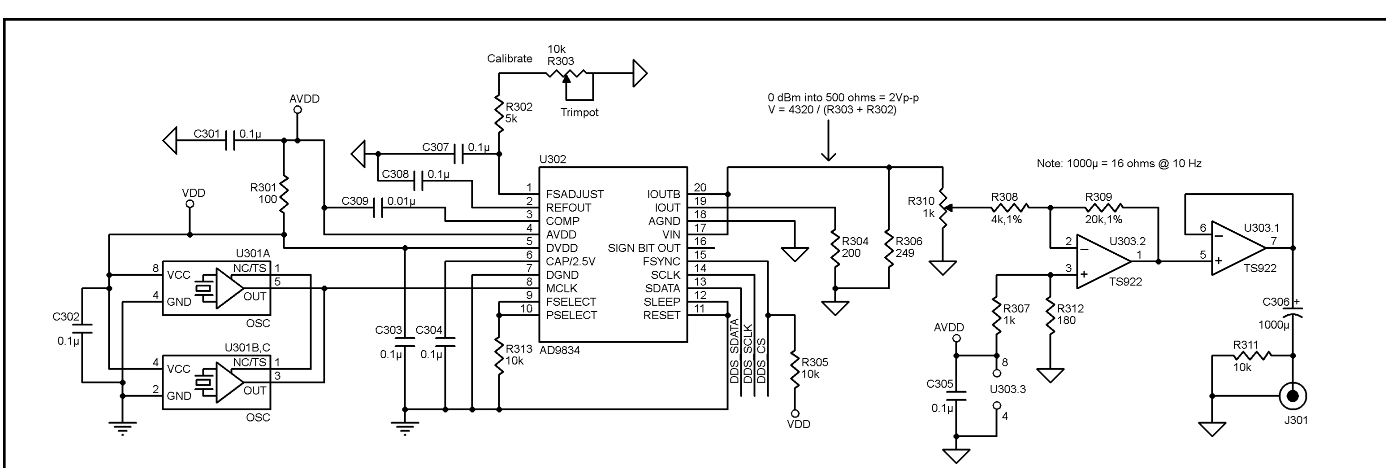


Figure 3: The DDS circuit is based on the AD9834, which is a low-power DDS device.

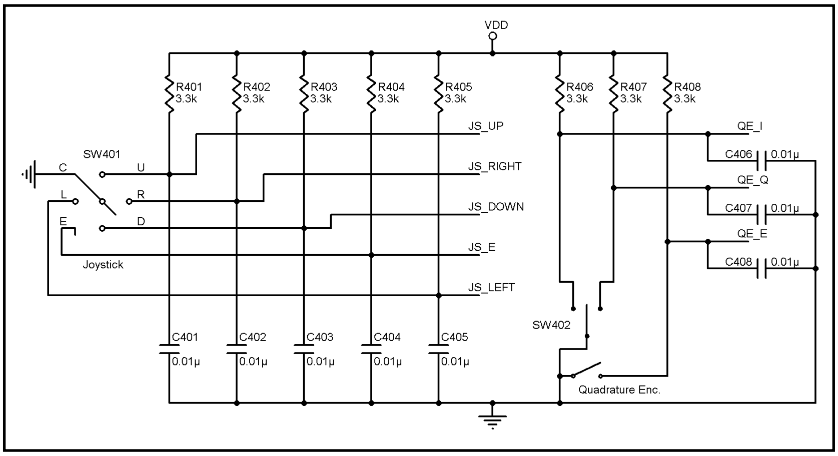


Figure 4: The tone control uses a single potentiometer, which enables adjustment of the frequency response from base boost/treble cut through to treble boost/base cut.

math to get the correct value to program.

R303 is used to calibrate the full-scale output current from the DDS IC so that with R310 fully clockwise, the output voltage is the desired maximum value: 2 V_{PP} is 0 dBm into 500 Ω and 2.19 V_{PP} is 0 dBm into 600 Ω. The DDS's output DAC delivers a current with the formula:

$$I_{OUT}(\text{full scale}) = 18 \times V_{REF}/R_{SET}$$

where V_{REF} is fixed at 1.2 V. Since the output resistor is 200 Ω, the formula becomes:

$$V_{OUT} = 200 \times I_{OUT} = 200 \times 18 \times 1.2/R_{SET} = 4{,}320/R_{SET}.$$

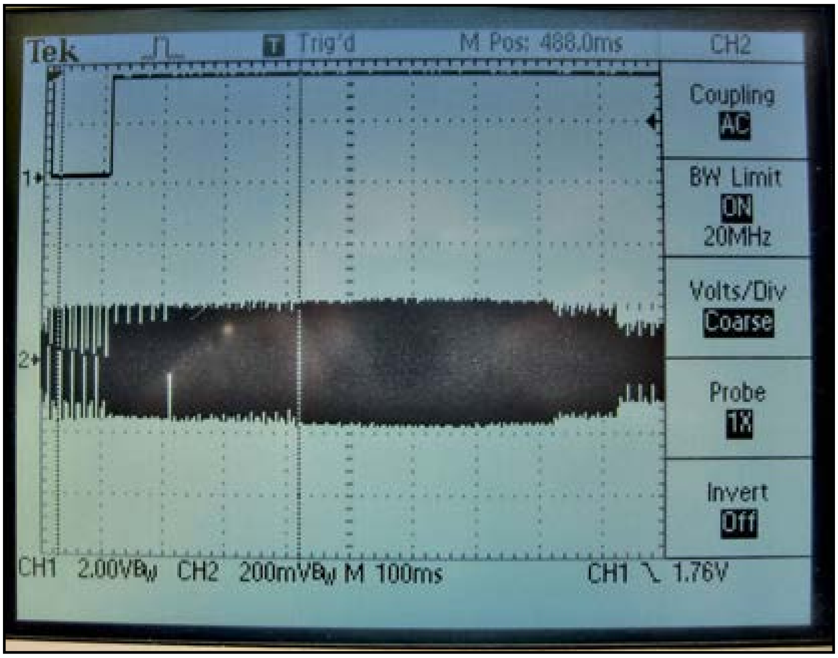


Photo 2: The frequency response is shown with the control roughly centered.

With the resistor values as shown on the schematic, the theoretical peak-to-peak output voltage range is 0.288 to 0.864 V_{PP}, although the compliance of the output for DAC is specified as 0.8 V.

The system is somewhat limited as to its output voltage since it uses a 3.3-V supply. The offset circuit on U302.2 is needed for two reasons: (1) the circuit is an inverter configuration but cannot have the output attempt to go negative and (2) even a rail-to-rail op-amp does not always include 0 V in its output range.

The equation for the output voltage of U303.2 is:

$$V_{OUT} = V_{OFFSET} \times (1 + R309/R308) - V_{IN} \times R309/R308$$

where: V_{OFFSET} = 0.503 V and V_{IN} is the voltage at the arm of R310. The equation becomes V_{OUT} = 3.02 – 5× V_{IN} after inserting all the values.

Keep in mind that this is the DC equation and that the AC output is 5 × V_{IN} as long as V_{IN} does not exceed 0.6 V. If R303 is adjusted for 2 V_{PP} output from U303.1 with R310 set for maximum, then the output of U302 will be 0.4 V_{PP}.

The output circuit is unbalanced and uses capacitive coupling. I would have preferred using a transformer but could not find a low-cost one with the desired bandwidth.

Software

The program is entirely written in Assembly Language. I purchased a PICkit-3 specifically for this project and used MPLab X for the development environment. To help me remember what functions I wrote and what parameters they require, I created a text file that basically contains the headers for each function. Since I am a firm believer in documenting code, all the lines of code are commented as are the functions. The complete MPLab project is available on my website as well as from *audioXpress* (see Resources).

Most of the arithmetic functions are specifically written for this project. Since the design upper limit frequency is 100 kHz with 0.1-Hz resolution, the functions only need to handle a maximum of 20 bits to program the DDS chip. This enables me to limit the functions to handling only three-byte values.

The DDS IC's output frequency is:

$$(F_{Clock}/2^{28}) \times N$$

where N is the value programmed into its 28-bit accumulator. Since I am using a 25-MHz oscillator, the equation becomes: Frequency = 0.0931322 × N. Since I know what frequency I want, solving the equation for N results in: N = 10.737418 × Frequency.

The first thing the algorithm does is multiply the desired frequency by 10. That result is then multiplied by 1.073730, which you can see is close to the desired value. The method I use to generate the fractional multiplier is to add values of the frequency × 10 divided by appropriate powers of two as follows, let Frequency × 10 be represented by F × 10 so, N = F × 10 + F × 10/16 + F × 10/128 + F× 10/512 + F × 10/1,024 + F × 10/2,048).

To get closer than this to the "real" value, the program would have to execute six more shifts and an addition. I did not think the extra processing was worth the minor improvement.

I wrote two functions to help me with the calculation: 3-byte shift right and 3-byte add. To conserve memory, the 3-byte shift is done "in place" while the 3-byte add updates an accumulator but does not modify the shifted value. This enables me to use the shifted value for both the add operation as well as the succeeding shift operations. There are other math functions that handle 3-byte values: multiply by 10, shift left, and subtract.

I also needed a divide function that would handle 3-byte values. I was all set to write one myself but I found a very fast function on the Internet that handles a 24-bit dividend and a 16-bit divisor—just what I needed! The source for the function is noted in the Project Files found in the Supplementary Materials section of the *audioXpress* website. I only had to make a few minor changes to accommodate my specific memory usage.

The Microchip website contains all the documentation on the PIC processors as well as MPLab X, which is a free download. There is a link on the homepage under Design Support for the official Forum which has lots of help for all users. To find the divide function previously mentioned, I entered "PIC integer divide" into my search engine and I was presented with several pages of possibilities.

Construction Notes

I have been using DipTrace for all my schematics and PCB designs for several years. There is a free version available from www.diptrace.com, which is fully functional. All of the design files are available from my website (www.qsl.net/k3pto) as well as from the Supplementary Materials section of *audioXpress* (www.audioxpress.com).

The BOM for the system includes all the parts for the circuit board—except for some mounting hardware—and the vendor part numbers. The power connector and the switch are parts I found in my stock and can be whatever you want to use.

I purchase most of my parts from Digi-Key Electronics since it has free shipping if a check is

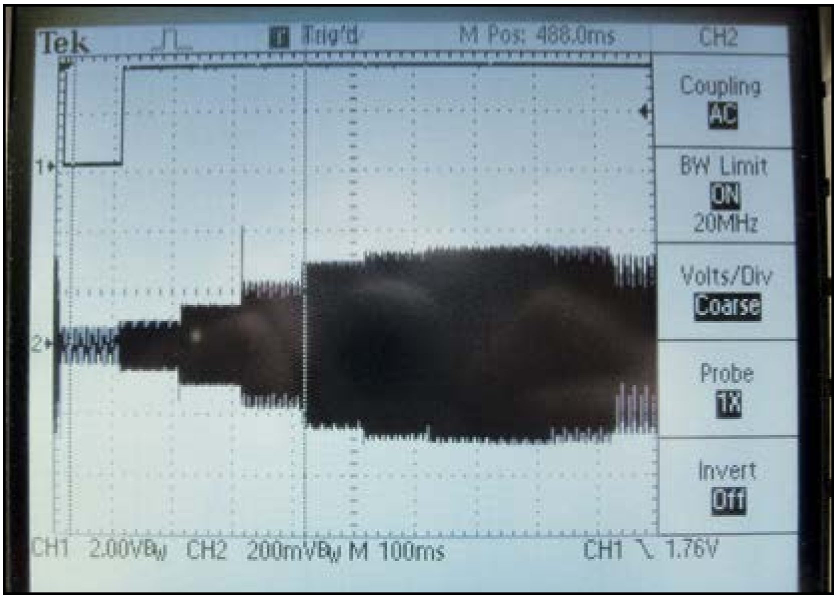


Photo 3: Here the frequency response is shown with the control in the treble boost/base cut position.

included with the order. If you use this service, be sure to include whatever sales taxes are applicable. There are only two parts in the BOM that are not from Digi-Key—the 5-V LCD and the knobs. You might notice that I have specified mostly 1% resistors. I have found that 1% and 5% SMD resistors are the same price so I always get the 1% units.

I did have a little difficulty implementing the joystick due to its shaft, which is small. I found an aluminum standoff in my stock that fit nicely over the shaft. I filled one end of the standoff with glue and put the joystick shaft into it. It works nicely as a shaft extension. The part listed on the BOM is similar. It should work, but I have not tried it.

I obtained the knobs from a friend. Each of the three controls has a 6-mm shaft, including the extension on the joystick. However, I did have to modify the joystick knob. I ground some of the bottom off so it would not hit the top surface of the box as the switch is moved around. The knob listed on the BOM is similar—at least its dimensions are what is needed.

The socket (BOM1) and header (BOM2) listed for the LCD had to be cut to the correct size (8 pins each). You will need two of each. I mounted the headers on

Project Files

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

Resources

DipTrace software, www.diptrace.com.

K3PTO Published Articles, www.qsl.net/k3pto.

Source

MPLAB PICkit 3 and MPLab X
Microchip Technology, Inc. | www.microchip.com

the LCD and the socket on the PCB, although it would work just as well if you reverse them.

Each time you cut a socket to size you will lose one pin. Cutting the header does not sacrifice any pins. One thing I like to do with all my connectors is to "paint" them with a permanent marker so I can easily see pin 1. I have a silver marker that shows up quite nicely on the black body of the headers.

I created a template for drilling the top of the case, based on the component centers on the PCB. The file should be printed using DipTrace since it will print 1:1. There is also a .jpg file, which can be used as a reference. I put the dimensions on it so the size can be verified after printing. The cutout for the LCD is also indicated using dimensions. Although there are two mounting holes in the PCB for the LCD, I did not use them. The two 8-pin sockets hold it in place. The hole for the joystick might seem a bit large but it must enable the shaft to move from side to side and top to bottom. I generally drill my holes a little bit small and then enlarge them using a reamer until I get them to the right size.

I soldered the LEDs onto the PCB last. To get the correct height, I mounted the PCB into the box with the LED leads in their holes. I pushed the LEDs up from the bottom of the PCB, until they protruded through the box by the amount I wanted. Then, I soldered them into place.

Usage

One simple use of the generator is to characterize audio filters. I built a simple tone control circuit for demonstration purposes (see **Figure 4**). The tone control uses a single potentiometer, which allows adjustment of the frequency response from base

Direct Digital Synthesis Primer

Direct Digital Synthesis (DDS) is a method of developing a programmable frequency using primarily digital techniques. The only analog part of a DDS system is a digital-to-analog convertor (DAC) and whatever filters might follow it. One of a DDS's primary features is that it can develop frequencies over a large range with a small value of resolution. The DDS device used in this article could develop frequencies from 0.1 Hz to more than 10 MHz with 0.1-Hz resolution.

Figure 1 is a simplified block diagram of a DDS system, where F_C is the main oscillator, or reference clock, which must be at least twice the highest desired output frequency. This is due to the Nyquist Criteria, which basically states that a signal must be sampled at least at twice its frequency in order to reproduce its frequency.

M is the tuning word, which is the mechanism for determining/controlling the output frequency. F_{OUT} is the output frequency and is equal to $M \times F_C/2^N$.

You can easily see from the formula that if $F_C = 2^N$, the output frequency will be the value of M in hertz, which is why I wanted an F_C of 26.843545 MHz for the 28-bit Phase Register (2^{28} = 268,435,456). This would have given the system a resolution of exactly 0.1 Hz and made the calculations much easier.

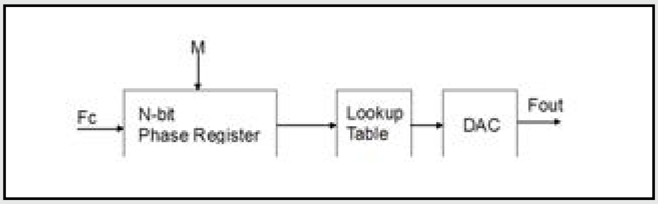


Figure 1: This is a simplified block diagram of a Direct Digital Synthesis (DDS) system.

The operation of the Phase Register is such that the value of M is added to its content with every F_C clock. When it overflows its range, it is reset to 0. The value of N for most DDS ICs ranges from 24 to 48 bits.

Also, and quite importantly, the data to the Lookup Table is truncated to the number of bits required by the DAC. Also, since a sine wave is symmetrical in its four quarters, the most significant two bits are used to control some mapping logic to create the full sine wave from a Lookup Table containing the sine values for 0° to 90°. With a 10 bit DAC, the Lookup Table would be driven by bits 16-25 for a DDS IC with a 28-bit Phase Register.

Another method of generating a programmable frequency is the Phase-Locked Loop (PLL), which has been around for quite a few years and is still a viable method for very high frequencies. One of the main differences between a PLL and a DDS is that the DDS will generate frequencies below its main oscillator while a PLL generates higher frequencies. Also, a PLL utilizes an analog oscillator that is usually controlled by an analog voltage whereas the DDS is completely digital except for the DAC, which does not enter into the control loop.

An advantage of the PLL is that its output can be very "clean" (i.e., there should be few harmonics or other noise/distortion components to the signal in a properly designed system). However, the DDS will have noise components that must be filtered out. The main advantage of a DDS is that it enables large frequency changes with high speed and excellent resolution. It is also possible to control a DDS signal's phase so that you can have multiple outputs with known phase relationships among them.

boost/treble cut through to treble boost/base cut. The generator is set to its sweep mode with the following parameters:

- Start frequency = 100 Hz
- Stop frequency = 55 kHz
- Step size = 1 octave
- Duration = 100 ms
- Marker = 100 Hz

With the above settings, the output frequencies, in hertz, will be: 100, 200, 400, 800, 1,600, 3,200, 6,400, 12,800, 25,600, and 51,200 Hz. The only reason I set the upper frequency so high is that I wanted 10 frequencies to have a "nice" oscilloscope display. With the oscilloscope set to 100 msec/div, each division represents a single frequency.

Photo 2 shows the frequency response with the control roughly centered. Trace 1 is the Marker, which is set to trigger when the frequency is 100 Hz. The slightly reduced signal at the high end is probably due to the circuit's simplicity.

I did verify that the output of the generator is flat across the control's entire range. **Photo 3** shows the frequency response with the control in the treble boost/base cut position. **Photo 4** shows it in the base boost/treble cut position.

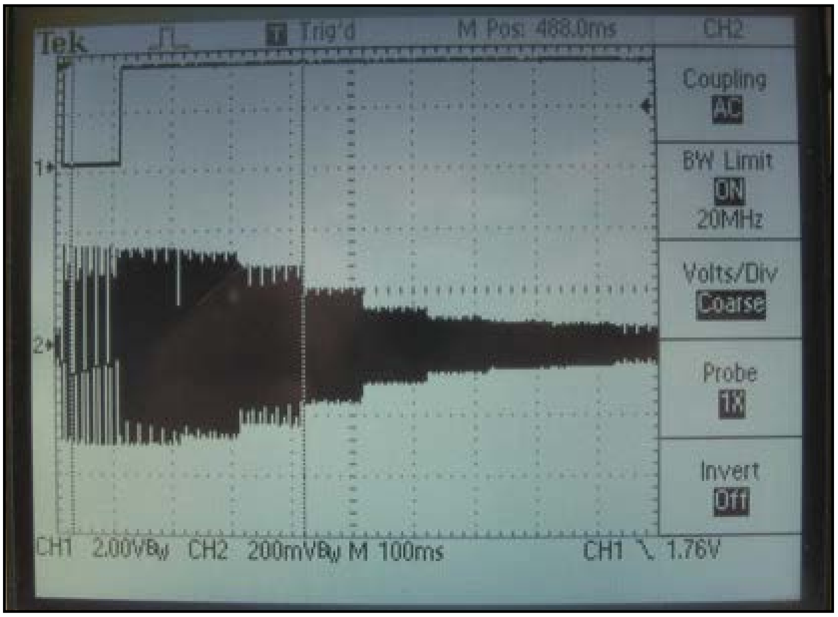


Photo 4: The frequency response is shown in the base boost/treble cut position.

Build a Single-Ended Guitar Tube Amplifier

Photo 1: This small, single-ended guitar tube amplifier is great for studio use.

Music, especially rock, jazz, blues, and pop owes much to the electric guitar and to vacuum tube amplifiers. I enjoy building simple tube amplifier designs because I believe simple circuits work better. Simple circuits are more effective, especially when the electric signals carry music information, because the signal distortion is easily handled.

By
Costas Sarris
(Greece)

When a guitarist friend urged me to build a guitar tube amplifier, we agreed that the amplifier design should be a simple, low-wattage single-ended amplifier—built much like a Swiss army knife—for everyday use. Inspired by late 1950s Fender "Princeton" and "Champ" amplifier designs, I decided to construct a small but efficient single-ended head amplifier, that any DIYer could easily build (see **Photo 1**).

This small amplifier is a medium/high gained head, meant to be played at room or studio levels. Although it can get pretty loud, it can also be played with smooth bass and "quiet drums" on small stages.

Because of its warm sound, the amplifier isn't made for any kind of hard metal music. On the other hand it is ideal for blues, rock, jazz, or funk music. The essence of this amplifier is simplicity. That's what makes it different (see **Photo 2**).

The Circuit

The music information generated as an electric signal by an electric guitar pickup is weak and needs amplification. Guitar pickups have different output voltages depending on the model. Low-output models tend to produce a "clean" sound, while high-output models tend to overdrive amplifiers and produce a "dirty and aggressive sound." The output voltage of most pickups varies between 100 mV and 1 VRMS.

This amplifier uses the high gain 12AX7/ECC83 double triode vacuum tube for the preamp stage, and the 6L6GC beam power tube for the output stage in a single-ended Class-A design.

Class-A amps are voltage amplifiers. In a Class-A amplifier the output voltage's wave shape is the same with the shape of the signal voltage applied to the grid. In other words, the power tube runs at full power all the time and that affects the way the output tube distorts. A Class-A power amplifier typically sounds warm and natural. When it tends to clip, it sounds dynamic and aggressive. That's the reason these amplifiers are desirable for many guitarists.

Preamplifier Circuit Design

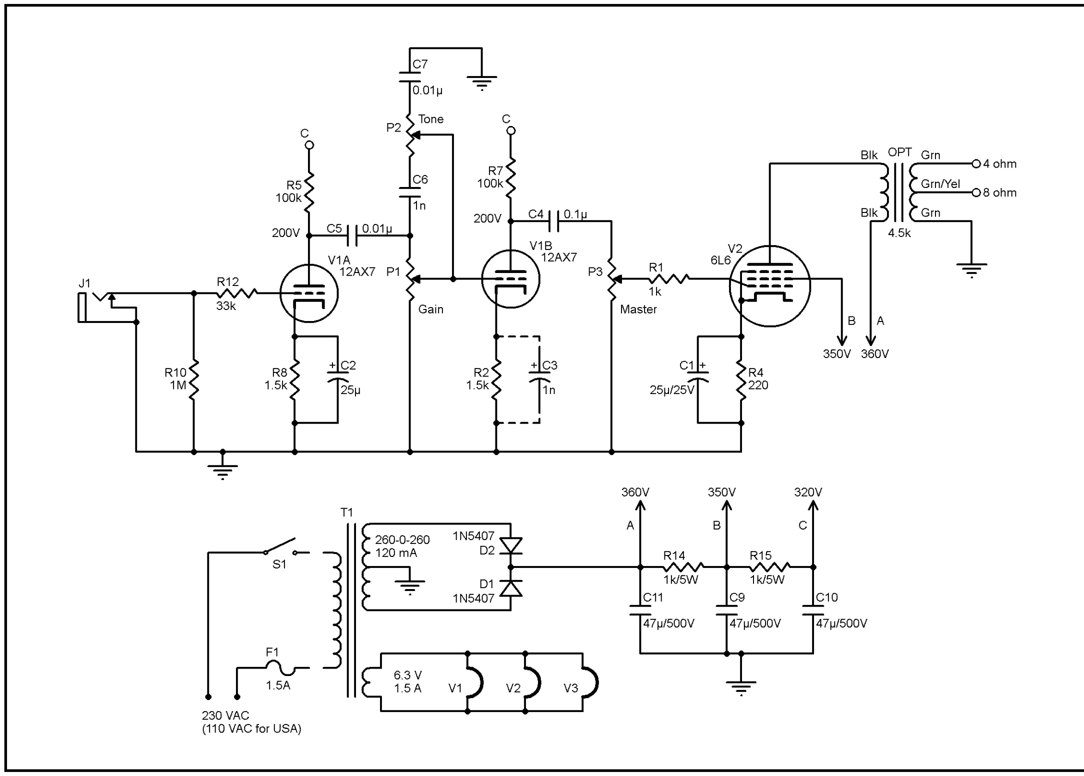
The preamp stage uses a high gain 12AX7/ECC83 dual triode vacuum tube, and operates as a grounded cathode amplifier. The output is taken from the second half of the 12AX7 triode plate to the power tube's grid, using a high-quality coupling capacitor 0.1 µF. The driver tube is cathode biased with a 1.5-kΩ resistor (R8, R2 shown in **Figure 1**), and a 200-V plate voltage (both stages).

The preamplifier's first half stage is a fully bypassed cathode bias circuit. The input grid-stopper resistor R12 is 33 kΩ. To bypass the cathode resistor in the first stage (the first half triode), I used a 25-µF/25-V audio electrolytic capacitor (C2 shown in **Figure 1**).

The gain potentiometer (1 MΩ) acts as a variable grid resistor to the grid of the second half of the 12AX7 preamp tube. The tone control section consists of a simple "tone boost control" circuit. C6 is the treble-shunt capacitor, so P2 (the tone control pot) affects both the "tone cut" and the boost. The amount of boost

Photo 2: This vacuum tube amplifier generates a warm sound.

Figure 1: The amplifier's design uses 12AX7/ECC83 dual triode vacuum tubes.



About the Author

Costas Sarris earned his BS in Business Computers from North College Thessaloniki in 1990. Today, he works as a Medical Systems Engineer for BioLab Fostieris, a leading medical diagnostics company in Greece. He has been constructing handmade tube amplifiers since 1992 as a hobby and decided to share his audio tube amplifier passion with the DIY audio community. He lives in Thessaloniki, Greece, with his wife and daughter. When he is not constructing tube amps or listening to music, he mountain bikes.

depends on the gain pot's setting. Tweaking the gain, you can adjust the amount of distortion you want.

The second half of the triode stage can be with or without a cathode bypass capacitor. Decreasing the cathode bypass capacitor's size, improves the amplifier's transient response.

While the power tube cathode resistor is fully bypassed (C1, 25 µF/25 V), the power stage needs about a 20-V signal amplitude to be driven to full power. In this circuit, the grid resistor is a variable resistor (1-MΩ master volume pot). So if we use a small size (e.g., 1,000 pF) cathode bypass capacitor in the

Photo 3: For this amplifier, I chose a custom-made, E-shaped, high-quality single-ended output transformer.

Photo 4: I used a filtered IEC power input connector for the AC input in the power supply.

Photo 5: The wood casing is made from handcrafted 16-mm thick birch plywood.

preamplifier's second stage, the amplifier's input sensitivity—with the master volume pot fully open (1 MΩ)—is only 15 mV. That means we only need a 15-mV input to overdrive the power stage! Using a high-output guitar pickup, we can easily overdrive the amplifier to produce the desired "aggressive sound." Increasing or decreasing the cathode bypass capacitance in the second half of the driver stage, we can improve the tone response, according to our individual taste.

Power Amplifier

In this project, the 6L6GC power tube is running at a 58-mA plate current when no signal is applied, and a 350-V plate voltage. The output transformer's primary impedance 4.5 kΩ. The power tube cathode resistor is fully bypassed by a 25-µf/25-V capacitor (C1). The input grid-stopper resistor is 1 kΩ (R1). The voltage across the cathode resistor is measured at 19.5 V.

Output Transformer

One of the most critical components in the sound path is the output transformer. Therefore, you should never compromise on the output transformer's quality.

For this amplifier, I chose a custom-made, "E" shaped, high-quality single-ended output transformer (see **Photo 3**). The resonance-free frequency range for this transformer is much better than 80 Hz to 20 kHz (–3 dB). The transformer has two secondary windings, 4 Ω and 8 Ω. I also recommend using Hammond Manufacturing's 125ESE or 125FSE output transformers.

The Power Supply

When designing a single-ended tube amplifier circuit, the power supply section must be carefully considered. For this project, my custom-made power transformer used the following windings:

Photo 6: I soldered all the parts to the soldering tag board prior to the assembly.

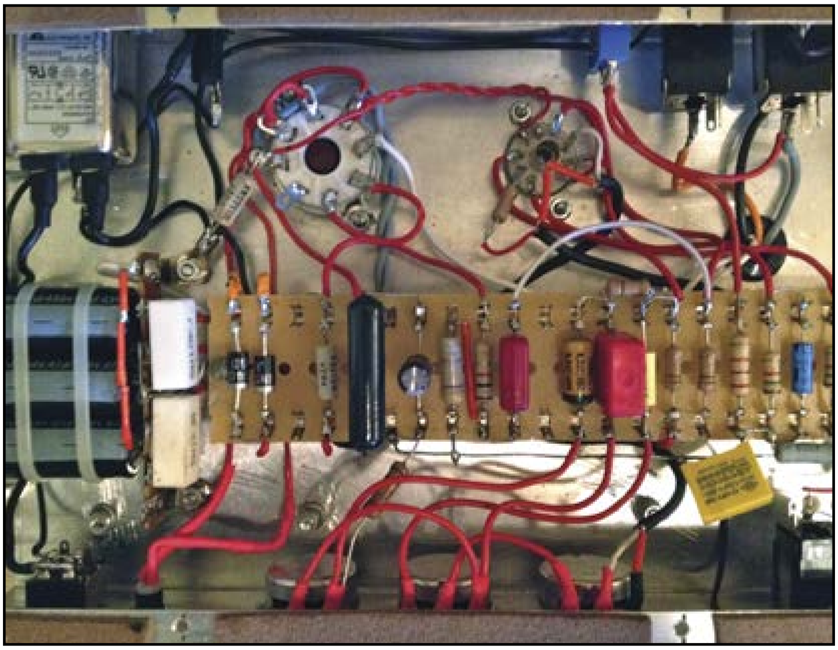


Photo 7: To complete the wiring, I connected all the parts directly from tube sockets and input/output jacks to the tag board.

- The primary winding was a 230 VAC (110 VAC in the US)
- The secondary windings were 6.3 V/1.5 A filaments for 12AX7 and 6L6 power tube and 260 V-0-260 V/120 mA

Hammond Manufacturing's 269AX power transformer is a good substitute and can be easily found. The rectifier circuit uses a single section "pi" filter or a "capacitor-input filter." Both the filter capacitor C11 and the smoothing capacitors C9 and C10 are 47-µF/500-V electrolytics. The rectifier bridge circuit in the power supply section consists of two 1N5407 general purpose rectifiers. Filaments are powered with 6.3 VAC for driving and power tubes. For the AC input in the power supply, I used a filtered IEC power input connector (see **Photo 4**).

Assembling the Amplifier

For the amplifier's construction, I used a ready-made 1.2-mm thick aluminum chassis. The dimensions were 25 cm × 15.5 cm × 4.5 cm (W × D × H). I made the wood casing from handcrafted 16-mm thick birch plywood (see **Photo 5**). Then, I covered the wood casing in leather. Tolex or textured vinyl can also be used.

Before the final assembly, I soldered all the parts to the soldering tag board (see **Photo 6**). All the pots are Alpha 24-mm full-shaft 1-M linear.

I installed all the basic components on the chassis (the tube sockets, the output transformer, the power transformer, the pots, the input, and the speaker jacks, etc.). Then, I finished the basic wiring starting

Parts List

Capacitors

C1 . . . . 25 µF/25 V
C2 . . . . 25 µF/25 V
C3 . . . . 1 nF/25 V
C4 . . . . 0.1 µF/500 V
C5 . . . . 0.01 µF/500 V
C6 . . . . 1 nF/500 V
C7 . . . . 0.01 µF/500 V
C9, C10, C11 . . . 47 µF/500 V

Fuse

F1 . . . . 1.5 A/ 250 V

Potentiometers

P1, P2, P3 . . . . . 1 M/alpha 24-mm linear potentiometer

Rectifiers

D1, D2 . . . . 1N5407 rectifiers

Tubes

6L6GC. . . . . power vacuum tube
12AX7/ECC83. . . preamp driving vacuum tube

Photo 8: All the tube sockets were made of high-quality porcelain.

with the filaments. I used 1.2-mm wire for the ground plane line, and for all the critical paths. All the soldering underneath the chassis, is hand-done, point-to-point wiring using a soldering tag board to place the parts.

All the power supply parts soldered in a separately tag board, installed in the left side of the chassis underneath the power transformer. To complete the wiring, I connected all the parts directly from the tube sockets and input/output jacks to the tag board (see **Photo 7**). All the tube sockets were made of high-quality porcelain (see **Photo 8**). For the final step, I mounted the chassis to the wood casing.

Test Impressions

Two guitarists tested the amplifier. One is Elias Zaikos, a legendary guitarist and founder of Blues Wire (www.blueswire.gr), and the architect of blues in Greek blues scene. The other guitarist and a good friend is George Liagas, owner and designer of Warlord custom pedals (www.guitarpedals.eu). He is also the one who urged me to build this amplifier.

Both guitarists agreed that with 10 W of output power, this amplifier rocks with amazing tone, and sounds superb when combined with a good-quality speaker. For our tests, we used a P12R 12” Alnico speaker.

In Practical Application

“Costas Sarris was kind enough to let me try one of his little wonders, and I’m so happy he did that!” said Zaikos. “I’m a working musician, I play tough and hard, electric blues. Guitar sound needs harmonics, tone, bite, and character. His amp gave me the chance to travel from sweet and bold single notes to cutting and exploring double stops and all points between, really nice! To make it short, when you play the blues on the road, all you need is an amp to “hear” you, to feel your touch and express your deepest inner self...and that ain’t easy. Costas’ little amp made me feel like I was playing through an old trusty sonic box. I couldn’t ask for more. Clear and gently or nasty and growling, this was a fantastic experience, Costas, thank you for sharing your work!”

So plug in and rock! Cheers!

Author’s Note: I would like to acknowledge George Liagas, founder and designer of Warlord custom pedals for his infinite help in testing and fine tuning the amp and Elias Zaikos for his effort testing and tweaking the amplifier.

Resources

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RCA Receiving Tube Manual, Technical series RC-19, RCA, 1959.

Finding Information on Vacuum Tubes

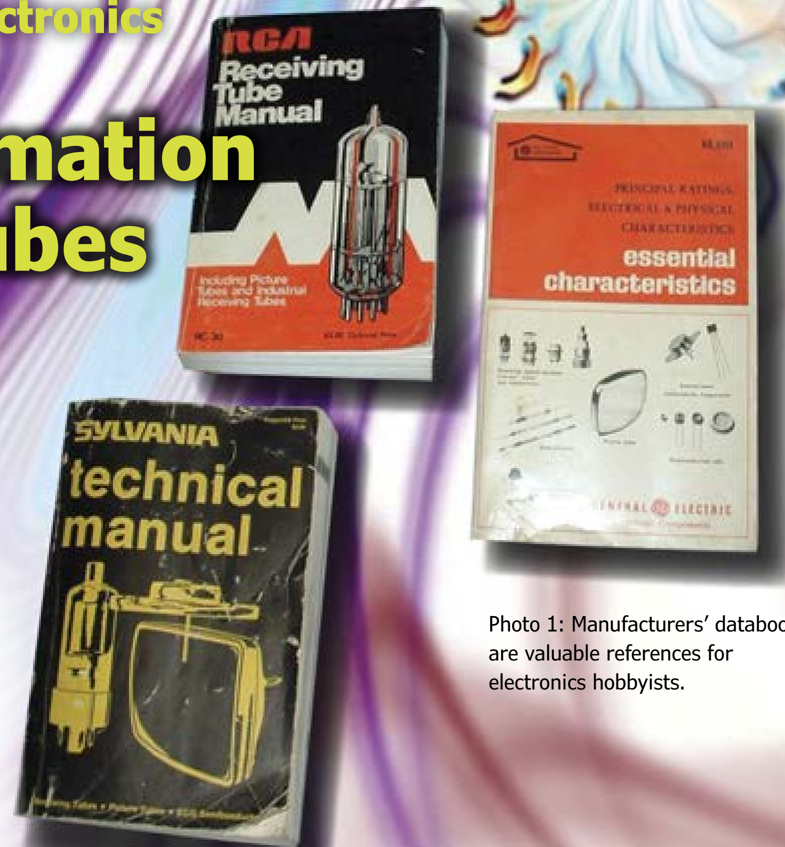

Photo 1: Manufacturers' databooks are valuable references for electronics hobbyists.

This article attempts to unravel some of the challenges associated with finding information on hollow-state components and how to determine what the data actually means. Even with the vast resources of the Internet, finding useful data can still be difficult.

By
Richard Honeycutt
(United States)

The late 1950s and early 1960s was a challenging time to be an electronics hobbyist. The availability of information on hollow-state components was increasingly limited to older books and a few magazines, in anticipation of the widely-heralded death of vacuum-tube technology. Most of the books on solid-state components were textbooks: manufacturers' data was hard to come by, and understanding what the component data meant—when you could actually find data—was not a straightforward process.

Fast-forwarding to today, we find that in some respects, the hobbyist has an easier time. The Internet provides a way to access data on most components, be they solid- or hollow-state. (To borrow from the popular Mark Twain misquote, reports of the death of hollow-state technology turned out to be greatly exaggerated!) Yet, often it is more convenient to have a databook at one's elbow than to have to shuffle among multiple webpages for specifications on several components. And, the problem of understanding the data remains.

Interpreting the Data

The second part of a hobbyist's problem — understanding the data—often comes down to recognizing the necessary and achievable degree of accuracy in the circuit-design process. For example, we often give the equation for the gain of a triode circuit as:

$$A_V = \frac{\mu R_l}{r_P + R_l}$$

where μ = the amplification factor of the triode, r_P = the value of the triode's plate resistance, and R_l = the AC load resistance connected to the triode plate (i.e., the plate resistor's value paralleled with the impedance of whatever the amplifier stage's plate is feeding).

The typical assumption most of us would make upon seeing this equation is that just as the resistance of a carbon-film resistor is a constant value, so are the μ and r_P of the triode. Is this true? No. Does it matter that these values depend upon

ELECTRONICS

12AX7-A
TWIN TRIODE

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DESCRIPTION AND RATING

The 12AX7-A is a miniature high-mu twin triode each section of which has an individual cathode connection. The 12AX7-A is especially suited for use in resistance-coupled voltage amplifiers, phase inverters, multivibrators, and numerous industrial-control circuits where high voltage gain is desired. A center-tapped heater permits operation of the tube from either a 6.3-volt or a 12.6-volt heater supply.

Use of the 12AX7-A is advantageous in applications that require low hum and microphonic output.

GENERAL

ELECTRICAL

Cathode—Coated Unipotential

	Series	Parallel	
Heater Voltage, AC or DC	12.6	6.3	Volts
Heater Current	0.15	0.3	Amperes

Direct Interelectrode Capacitances

	With Shield*	Without Shield	
Grid to Plate, Each Section	1.7	1.7	pf
Input, Each Section	1.8	1.6	pf
Output, Section 1	1.9	0.46	pf
Output, Section 2	1.9	0.34	pf

MECHANICAL

Mounting Position—Any
Envelope—T-6½, Glass
Base—E9-1, Small Button 9-Pin

MAXIMUM RATINGS

DESIGN-MAXIMUM VALUES, EACH SECTION

Plate Voltage	330	Volts
Positive DC Grid Voltage	0	Volts
Negative DC Grid Voltage	55	Volts
Plate Dissipation	1.2	Watts

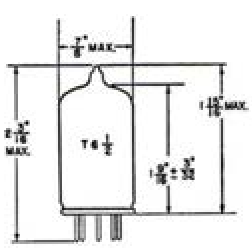
Heater-Cathode Voltage
Heater Positive with Respect to Cathode

DC Component	100	Volts
Total DC and Peak	200	Volts

Heater Negative with Respect to Cathode

Total DC and Peak	200	Volts

PHYSICAL DIMENSIONS



EIA 6-2

TERMINAL CONNECTIONS

Pin 1—Plate (Section 2)
Pin 2—Grid (Section 2)
Pin 3—Cathode (Section 2)
Pin 4—Heater
Pin 5—Heater
Pin 6—Plate (Section 1)
Pin 7—Grid (Section 1)
Pin 8—Cathode (Section 1)
Pin 9—Heater Center-Tap

BASING DIAGRAM

EIA 9A

GENERAL ELECTRIC

Supersedes ET-T5098 dated 6-53

Figure 1: This is the first page of the General Electric (GE) datasheet for the 12AX7A.

the tube's operating conditions? Sometimes. Without getting into the physics of a tube's construction and what elements affect the μ and r_P, let's see what information we can find, and then figure out what it means.

Understanding the Datasheet

The most common information source about a particular vacuum tube's operating characteristics is a datasheet. For many tubes, these datasheets have been compiled into "tube manuals," as RCA called them, or "technical manuals," as Sylvania called them (see **Photo 1**).

These books are now available from eBay, Amazon, and perhaps other Internet sellers, and in .pdf form from a several websites. Individual tube datasheets can also be downloaded from frank@pocnet.net and 11 mirror sites. As we will see, different manufacturers include different forms of tube data in their datasheets, and tube manuals may contain more than just datasheets.

Figure 1 shows the first page of the General Electric (GE) datasheet for the 12AX7A dual triode. At the top is the tube's general description and its intended applications. Below this description are electrical and mechanical characteristics.

The electrical characteristics include the heater (filament) voltage and current both for the full heater—pin 4 and pin 5—connected across a 12.6-V supply ("Series"), and for the two halves of the heater connected in parallel across a 6.3-V supply (one side of the heater supply connected to pin 4 and pin 5, and the other connected to pin 9). The heater's electrical data is important for specifying the power transformer's filament winding ratings. The other class of electrical characteristics is the interelectrode ("parasitic") capacitances. These capacitances affect the frequency response of the amplifiers built using these tubes.

The mechanical characteristics tell the designer what kind of socket the tube needs (the "Base"), and how much physical space to allow for each tube (the "Envelope"). The envelope number, T-6 ½, may be keyed to a dimensioned illustration in another part of the tube manual. In this case, the dimensioned outline appears at the lower left of page 1 of the datasheet.

The data section entitled "Maximum Ratings" is pretty much self-explanatory. These are the ratings that should not be exceeded when the tube is operating. Unlike solid-state devices, vacuum tubes will survive short periods during which the maximum ratings are exceeded, although the life of the tubes may be diminished. Some datasheets use the term "design-maximum values" to indicate these specifications. They may even specify maximum instantaneous values, which, if exceeded, may result in the tube's catastrophic failure.

The only maximum value that may need clarification is the plate dissipation. In a Class-A amplifier stage, the average value of plate current remains constant (often called "I_{PQ}," for quiescent plate current). The product of $I_{PQ} \times V_P$ equals the tube's plate dissipation. This is not the same as the power output of the tube stage. In fact, for a Class-A stage, the sine-wave power output cannot be greater than one-third of the plate dissipation. Plate dissipation causes heating of the tube's plate. If the plate dissipation becomes too great, the tube can "red-plate," meaning that the plate can be heated red-hot, with the possible result of the plate and/or its supporting structures sagging and damaging or destroying the tube.

The information found at the bottom section of **Figure 1** is self-explanatory. **Figure 2** shows the second page of the GE 12AX7A datasheet. The information shown on this page applies to the tube's typical operation in a circuit. At the top, two Class-A amplifier examples are given: one with a 100-V plate voltage and another with a 250-V plate voltage. Notice that even though the control grid is biased more negatively for the 250-V case, the I_{PQ} is twice as large as for the 100-V case. The μ is the same in both instances, but the plate resistance is lower with the higher plate voltage. The transconductance

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CHARACTERISTICS AND TYPICAL OPERATION

CLASS A₁ AMPLIFIER, EACH SECTION

Plate Voltage	100	250	Volts
Grid Voltage	−1	−2	Volts
Amplification Factor	100	100	
Plate Resistance, approximate	80000	62500	Ohms
Transconductance	1250	1600	Micromhos
Plate Current	0.5	1.2	Milliamperes
Equivalent Noise and Hum Voltage, Each Section, Average, True RMS†		1.8	Microvolts

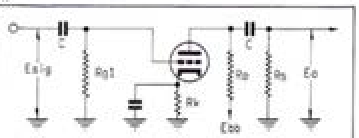
* With external shield (EIA 315) connected to cathode of section under test.
† Referred to grid and measured under the following conditions: Ef = 6.3 volts AC (parallel connection), center tap of heater transformer grounded; Ebb = 250 volts, Rb = 100,000 ohms; Rk = 2700 ohms, bypassed with 100 uf; Rg = 0 ohms; amplifier range = 25 to 10,000 cps.

Design-Maximum ratings are limiting values of operating and environmental conditions applicable to a bogey electron tube of a specified type as defined by its published data and should not be exceeded under the worst probable conditions.
The tube manufacturer chooses these values to provide acceptable serviceability of the tube, making allowance for the effects of changes in operating conditions due to variations in the characteristics of the tube under consideration.
The equipment manufacturer should design so that initially and throughout life no design-maximum value for the intended service is exceeded with a bogey tube under the worst probable operating conditions with respect to supply-voltage variation, equipment component variation, equipment control adjustment, load variation, signal variation, environmental conditions, and variations in the characteristics of all other electron devices in the equipment.

The tubes and arrangements disclosed herein may be covered by patents of General Electric Company or others. Neither the disclosure of any information herein nor the sale of tubes by General Electric Company conveys any license under patent claims covering combinations of tubes with other devices or elements. In the absence of an express written agreement to the contrary, General Electric Company assumes no liability for patent infringement arising out of any use of the tubes with other devices or elements by any purchaser of tubes or others.

CLASS A RESISTANCE-COUPLED AMPLIFIER
EACH SECTION

Rp Meg.	Rg Meg.	Rg1 Meg.	Ebb = 90 Volts			Ebb = 180 Volts			Ebb = 300 Volts		
			Rk	Gain	Eo	Rk	Gain	Eo	Rk	Gain	Eo
0.10	0.10	0.1	1700	31	5.0	1000	40	15	760	43	30
0.10	0.24	0.1	2000	38	6.9	1100	46	20	900	50	40
0.24	0.24	0.1	3500	43	6.5	2000	54	18	1600	58	37
0.24	0.51	0.1	3900	49	8.6	2300	59	24	1800	64	47
0.51	0.51	0.1	7100	50	7.4	4300	62	19	3100	66	39
0.51	1.0	0.1	7800	53	9.1	4800	64	24	3600	69	46
0.24	0.24	10	0	37	3.9	0	53	15	0	62	32
0.24	0.51	10	0	44	5.4	0	60	19	0	67	41
0.51	0.51	10	0	44	5.0	0	61	17	0	69	35
0.51	1.0	10	0	48	6.4	0	66	21	0	71	41



Note: Coupling capacitors (C) should be selected to give desired frequency response. Rk should be adequately by-passed.

Notes: 1. Eo is maximum RMS voltage output for five percent (5%) total harmonic distortion. 2. Gain measured at 2.0 volts RMS output. 3. For zero-bias data, generator impedance is negligible.

Figure 2: This is the second page of the General Electric (GE) 12AX7A datasheet.

Resources

"General Electric 12AX7-A Twin Triode" datasheet, www.mif.pg.gda.pl/homepages/frank/sheets/093/1/12AX7A.pdf.

"RCA 6AV6 Twin Triode—High mu Triode," Electron Tube Division Radio Corporation of America (RCA), 1959.

Sylvania Radio Tube Technical Manual, Sylvania Electrical Products, 1951, www.tubebooks.org/tubedata/Sylvania/1951/sylvania_1951.pdf.

"Tung-Sol 12AX7 Specifications," Tung-Sol Electron Tubes, www.tungsol.com/tungsol/specs/12ax7-tung-sol.pdf.

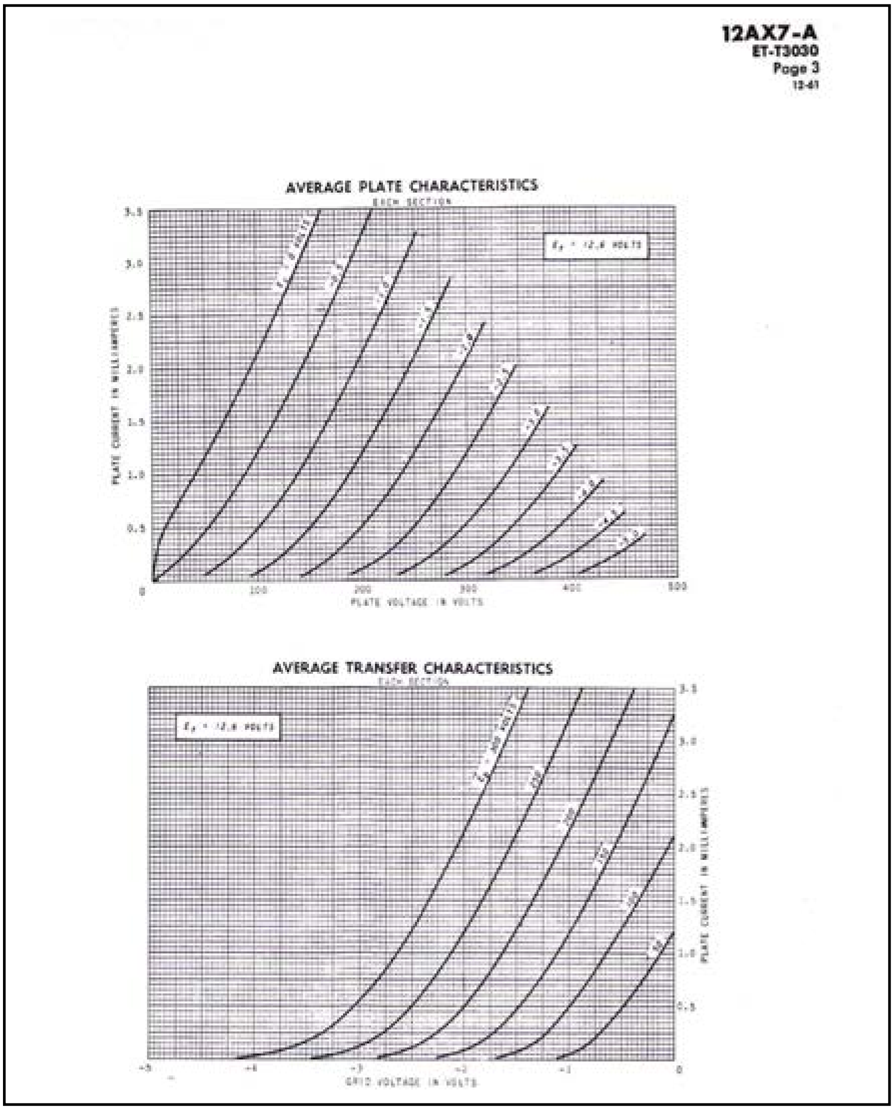


Figure 3: This is the third page of the General Electric (GE) 12AX7A datasheet.

is also higher.

A tube's plate resistance (r_P) is the ratio of the change in the voltage across the tube (V_{AK}) to the plate current. It is also sometimes called the "dynamic resistance" of the tube. The transconductance (g_m) is a sort of gain. It is the ratio of the change in plate current produced in response to a given change in control-grid-to-cathode voltage. Thus, whereas the μ is the gain when the triode is viewed as a voltage-controlled voltage source (VCVS), the g_m is the gain when the triode is viewed as a voltage-controlled current source (VCCS).

A true voltage source has zero output resistance, but a triode amplifier stage has an output resistance approximately equal to r_P. A true current source has an infinite output resistance, since the output current does not depend upon the load resistance connected to it. Thus, a triode amplifier stage is really neither a VCVS nor a VCCS. But just as the stage voltage gain can be expressed as:

$$A_V = \frac{\mu R_l}{r_P + R_l}$$

it can also be expressed as:

$$A_V = g_m r_p$$

The table shown at the bottom of **Figure 2** shows the component values, the gain, and the maximum output voltage of one triode of a 12AX7A with 10 different combinations of B+ voltages and component values. The upper six cases represent cathode-biased circuits, probably biased for minimum distortion (although the datasheet does not say so).

The lower four cases are grid-leak biased. Some tube manuals (e.g., the *RCA Receiving Tube Manual*) include a section after the datasheets in which similar design tables are included, each table applying to a variety of tube types,

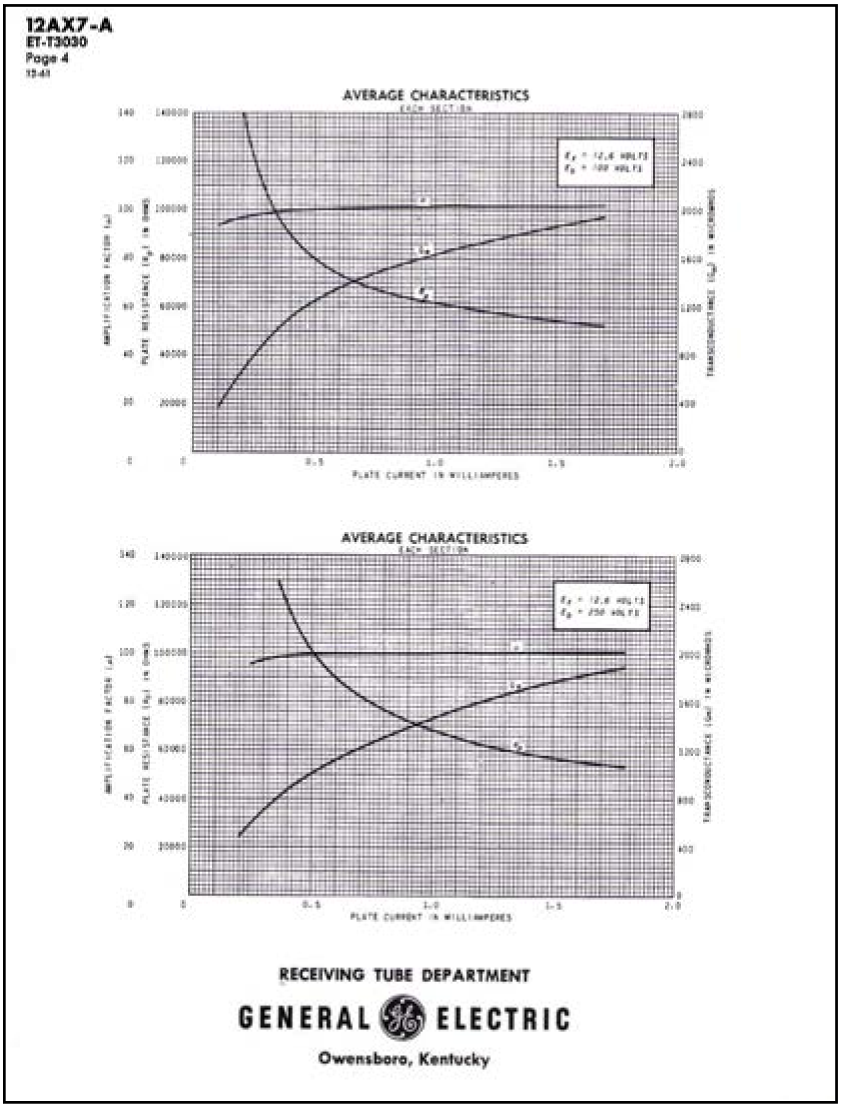


Figure 4: This is the fourth page of the General Electric (GE) 12AX7A datasheet.

in lieu of including the table in the datasheet.

Since a triode of a 12AX7A is virtually identical to a triode of many other tube types, the design data for the 12AX7A also applies to the 3AV6, the 4AV6, the 6AV6, the 6EU7, the 12AV6, and the 20EZ7, as well as the ECC83 (the European number for 12AX7A), the 7025 (a low-hum/noise version of a 12AX7A), the 6681, and the 7729 (which are industrial versions).

If you calculate the gain using the component values given in the chart, and a μ of 100, you will not obtain the voltage gain predicted by the chart. Part of the reason is that μ varies with the triode's operating conditions. There may be other reasons as well, but in my experience, the voltage gain and maximum output voltage predicted by these design charts is only approximate.

Figure 3 shows the plate characteristic curve (plate current plotted vs. plate voltage) and the transfer characteristic curve (plate current plotted vs. control-grid voltage). The parameter that is held constant for each curve of the plate characteristic is the control-grid voltage. Thus, the value of r_P can be determined by inverting the slope of the curve at the desired grid-bias voltage. This is true because:

$$r_P = \frac{v_P}{i_P} = \frac{\Delta V_P}{\Delta I_P}$$

and the slope of the curve is "rise/run," or ΔIP/ΔVP. The slope is taken at the Q-point, which is the point representing zero-signal plate voltage and current.

The tube's transconductance at any operating point can be found as the slope of the transfer characteristic curve at that operating point.

Figure 4 shows the GE datasheet's fourth page for the 12AX7A, which contains two graphs that enable the designer to find the value of μ, r_P, and g_m for any chosen value of plate current.

Datasheets Differ Based on the Manufacturer

Different manufacturers include different selections of data in the datasheet for each tube type. For example, the RCA datasheet for the 12AX7A refers the reader to the 6AV6 datasheet, which does not include the transfer characteristic or the graphs of μ, r_P, and g_m vs. plate current. The *Sylvania Technical Manual* shows no curves at all for either the 12AX7A or the 6AV6. The Tung-Sol datasheet does not include the transfer characteristic, and includes the graphs of μ, r_P, and g_m vs. plate current in a different format than that used by GE.

At least half-a-dozen other manufacturers' datasheets can be found online, each with its own individualities. Thus, in seeking vacuum-tube data, designers have many options from which to choose.

Author's Note: In Hollow-State Electronics article "Classic Tube Power Amplifier Circuits," which ran in the April 2015 issue of audioXpress, *there were a few errors. First, the phase splitters in the Heathkit and Fisher amplifiers are of the split-load variety, not long-tailed pairs. In both cases, the phase splitter is the second stage, and it is followed by a balanced (Class-A push-pull) driver stage. Second, the cathode circuit of the Fisher has cathode feedback provided by a tapped winding of the output transformer, not a tapped inductor. Third, the transformer winding feeding the screens of the McIntosh actually provides positive feedback to bootstrap the final driver and output screens. My thanks to reader Roger Modjeski of The Berkeley Hi Fi School (berkeleyhifischool.com) for alerting me to these corrections.*

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

May 26–28, 2015 — 19th AES Brazil Expo

Convention: May 25–28
Exhibition: May 26–28
Expo Center Norte Pavilhão Amarelo, Vila Guilheme, São Paul, Brazil
www.brasilexpo.com.br/en

The largest event of its kind in Latin America, AES Brazil is not exactly the usual Audio Engineering Society (AES) convention you find in the US or Europe. The Brazilian event is much more business oriented and actually combines several exhibitors displaying video, lighting, and rigging solutions, apart from professional audio, with a dominant focus on live sound and professional audio. The convention itself is a well-attended event, attracting audio engineers and high-level professionals mainly from Brazil. Most of the sessions are only in Portuguese. Organized in cooperation with Francal and the local AES section, the expected 75 exhibitors, occupying a combined area of 5,000 m^2 will be the focus for international visitors, even though most of the large exhibits are from the largest local distributors or local manufacturers. Harman, Yamaha, Meyer Sound, and Proshows will promote workshops and practical training sessions.

May 29–31, 2015 — The Home Entertainment (T.H.E.) Show Newport

The Hotel Irvine, Irvine, CA, US
http://theshownewport.com

The Home Entertainment (T.H.E.) Show has announced sweeping changes for T.H.E. Show Newport Beach. T.H.E. Show Newport Beach will now be held at The Hotel Irvine in Irvine, CA. T.H.E. Show Las Vegas, usually scheduled for January will be put on hiatus after 16 years. The promoters explain "all of T.H.E. Show's resources and time will be spent making 2015 a game-changer and a spectacular event."

Visitors to T.H.E. Show Newport Beach can visit the Headphonium exhibitors, the Marketplace, vinyl, CD, and accessories vendors all in one location—a giant 15,000-ft^2 Grand Ballroom in the center of The Hotel Irvine. Returning to the event in 2015 will be the gourmet food trucks, a giant high-end auto show, live entertainment day and night, wine tastings, and many popular educational seminars orchestrated by the Los Angeles and Orange County Audio Society (LAOCAS).

May 31–June 5, 2015 — Euronoise 2015

MECC Congress Center, Maastricht, The Netherlands
www.euronoise2015.eu | Euroacoustics.org

Euronoise 2015, the 10th European Congress and Exposition on Noise Control Engineering, will be held in Maastricht, in the heart of Europe where the first treaties leading to the creation of the European Union were signed. Acousticians and noise experts from all over the Europe will gather for the event on noise control, and soundscape in Europe, organized by the European Acoustics Association (EAA).

Promoted this year by the Belgian and Dutch acoustical societies, ABAV and NAG, the Euronoise 2015 conference comprises keynote lectures, tutorials, parallel technical sessions, an exhibition, and short courses. The program focuses on environmental noise control, acoustics, and improvement of people's quality of living. The organization confirms it has received more than 600 abstracts.

Euronoise 2015 will host an extended European Acoustics Exhibition, addressing all European providers of acoustic products, services, and information. In this exhibition, companies, manufacturers, engineering service providers, research and administrative units, technical and scientific organizations, and publishers will discuss the newest results in research and development, about equipment and instrumentation, methods and software solutions and materials, standards, guidelines, and publications.

June 13–19, 2015 — InfoComm 2015

Orange County Convention Center, Orlando, FL, US
www.infocommshow.org

InfoComm International is the trade association representing the professional audiovisual (AV) and information communications industries worldwide. InfoComm 2015, the largest audiovisual show in North America, will be held in Orlando, FL, with the convention taking place from June 13–10, and the exhibits open from June 17–19. InfoComm 2015 has more than 500,000 net ft^2 of floor exhibits and special events space. More than 37,000 professionals are expected to attend the show, with more than a third of attendees coming from technology managers, specifiers, and end-user communities.

More than 980 companies are confirmed exhibitors. In addition to the show floor and hundreds of InfoComm University sessions, Manufacturers' Training, and specialized conferences, InfoComm has numerous networking events. The 2015 opening session at the Chapin Theater, on June 16, will feature a panel of industry thought leaders on The Internet of Everything, moderated by Nick Bilton, Lead Technology Writer/Reporter for *The New York Times*. The traditional Awards Dinner is also confirmed for the same day, recognizing honorees of Adele De Berri Pioneers of AV Award, Distinguished Achievement Award, Educator of the Year Award, Sustainable Technology Award, Women in AV Award and Young AV Professionals Award.

July 9–11, 2015 — Summer NAMM 2015

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

With exhibits already surpassing last year's totals, the Summer National Association of Music Merchants (NAMM) 2015 will 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 is a business-friendly environment mixed with great entertainment and networking opportunities, benefiting from an amplified presence of pro audio brands. The Summer NAMM event provides a dynamic platform for emerging and established brands to create lasting impressions as buyers plan for the second half of the year.

Nearly 25 companies, including Marshall Amplification, Washburn Guitars, Santa Cruz Guitar Company, Zildjian, DigiTech, Seymour Duncan, and Amati's Fine Instruments are coming back to the show floor for the first time in several years. New in 2015, TEC Tracks offers educational sessions focused on recording and live sound production in a dedicated area adjacent to the NAMM Idea Center.