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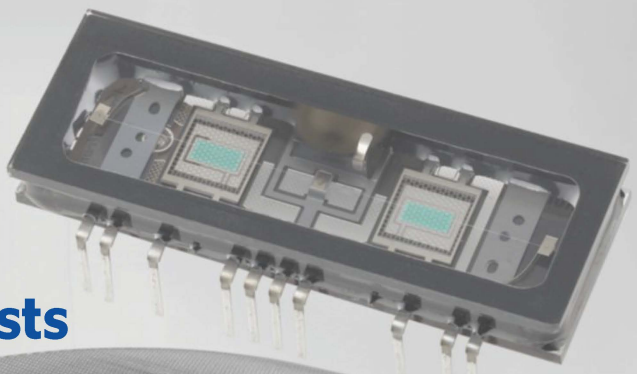
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By Bruce Brown





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By Pete Millett

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The Positive Energy of Tubes

This is our Glass Audio special edition. For those less familiar with the magazine's history and first-time/occasional readers, *Glass Audio* was a periodical for tube lovers with a penchant for building their own equipment. In 2000, Edward T. Dell—who also published other popular periodicals including *The Audio Amateur*, *Audio Electronics*, and *Speaker Builder*—decided to merge all the titles into a single monthly publication, named *audioXpress*.

audioXpress carries on the Glass Audio tradition to the present day. When *Glass Audio* was founded in 1989, solid-state equipment was a dominant focus and it was thought the interest in tubes would diminish. Quite the opposite has happened. Despite the difficulties of decreasing supplies of tubes, rising prices for many of them, and the slow development of other manufacturing sources in Eastern Europe, Russia, and China, the community of enthusiastic tube project constructors continues to thrive.

Quoting Dell, in *audioXpress*: "Readers continued to discover the wonderful world of tubes, and many started seeing possibilities in hybrid circuits, which combine the advantages of solid-state power and voltage control serving the operation of tube circuitry in ways not possible in the earlier history of electronics.

Almost every ancillary part in tube designs is better and cheaper today than it was when tubes began to be regarded as obsolete. In short, tubes are sounding better than they ever did. Tube enthusiasts also know that building tube equipment is, in many ways, easier and more fun than solid state. The feeling is much more 'hands on' with tubes and the resulting units are easier to modify or repair."

So, we thought it would be extremely motivating to feature an article, written by Dan Fraser, detailing the thinking and challenges of reviving the legendary Dynaco ST-70, which is now under ownership of Radial Engineering and counting on the personal enthusiasm and commitment of its President and founder, Peter Janis.

During our exchanges with Radial in preparation for the article, I contacted our tube experts (among the best in the world) and asked them to send me relevant questions to submit to Dan Fraser. It was then that I truly understood the challenge. Virtually all our authors had extensive previous knowledge and experience with Dynaco tube amps and kits. Virtually everyone had their own opinions. No one wished to bear the responsibility of designing a successor Dynaco tube amp today and many even questioned the reasons to do it. Whether agreeing or strongly opposing Radial's options for the new ST-70 Series 3, the truth is, there was no consensus on what to do, just an agreement on the huge responsibility of carrying on David Hafler's tradition.

The other reality is, as Peter Janis found out, that in tandem with the surprising vinyl revival, there's increased interest on tube electronics, mostly from people who don't want to deal with reliability issues and know nothing or care about electronics. The market is there and firm orders have been placed, justifying the challenge.

And no one understands the importance of tubes better than Fabio Camorani, who details his fifth-generation iteration of his Kismet tube design. Describing his 30 years' experience with tube designs and how those provide the right amount of energy to music reproduction, he writes, "When I listen to music at home, my way is to look for the same emotion I feel during live music." And tube designs provide exactly that right level of energy and emotion.

And if you read Pete Millet's article about the Korg Nutube 6P1, a new, miniature, low-power dual directly-heated triode device, it suddenly becomes clear that the sound of "tubes," in a broader sense, is here to stay.

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.

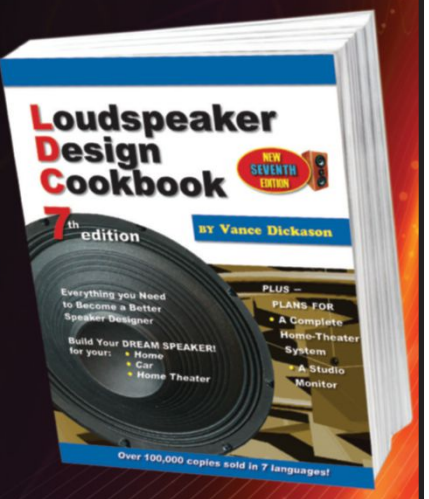
Dr. Richard Honeycutt fell in love with acoustics when his father brought home a copy of Leo Beranek's landmark text on the subject while Richard was in the ninth grade. Richard is a member of the North Carolina chapter of the Acoustical Society of America. Richard has his own business involving musical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

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

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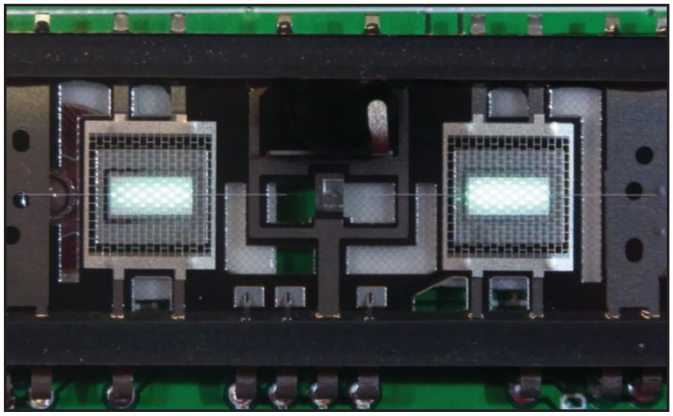
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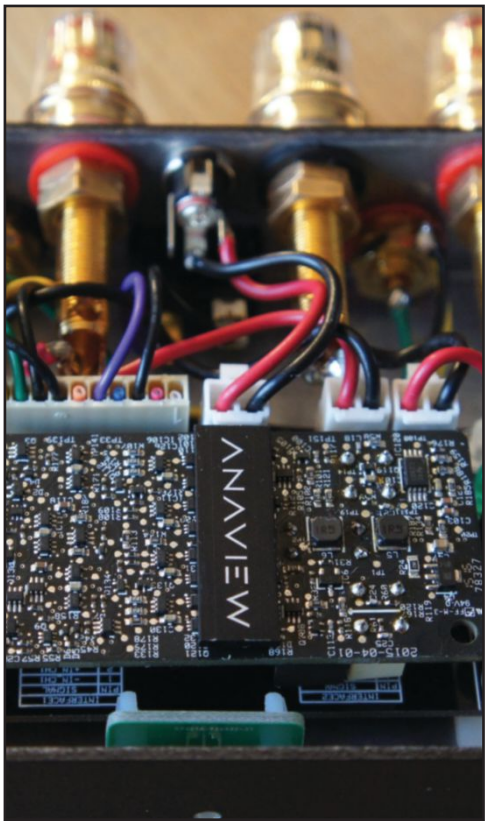
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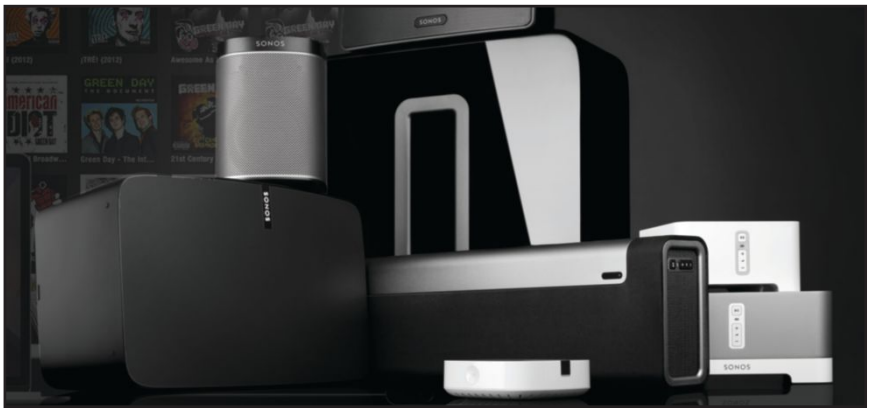
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Streaming Music Experiences (Part 5)

Streaming from Sonos and Deezer



By
Ron Tipton
(United States)

In this series, we continue to evaluate the quality of streaming music services. This month we explore Sonos and Deezer.

Sonos began as a wireless "whole-house" system for streaming digital stereo music from several providers. The 45-name provider list, as of October 2016, is included as an Adobe .pdf file in the Supplementary Material. To my knowledge, Deezer and TIDAL are the only providers streaming CD-quality FLAC files.



Photo 1: My Sonos-Connect box sits in my audio equipment rack. The white indicator is illuminated during normal operation and the bar below the indicator is an up-down volume control.

But not everyone wants a wireless system so Sonos introduced two hardware modules, the Sonos-Connect and the Sonos-Amp, which connect to your router through an Ethernet cable. The Sonos-Amp includes a stereo power amplifier, while the Connect model has its analog output on a pair of RCA connectors and a digital TOSLINK (optical) connector. I didn't need the Amp model so I purchased a Connect. The price seems a bit high for a stereo DAC but it does come in an attractive white enclosure and it works well (see **Photo 1** and **Photo 2**).

Sonos has free control software for iOS and Android mobile devices, and Mac and Windows (7 and up) computers. Installation is straightforward, both your computer and Connect box must be connected to your router. After installation is complete, click the Sonos desktop icon. **Photo 3** shows the opening screen.

As part of the installation, you are asked where the Connect box will be located. I don't know why this matters, but I selected Media Room and this is displayed in the left side column. The next step is choosing your streaming service by clicking the last selection in the right-side column. I could have chosen TIDAL but I decided to give Deezer a try. During the Deezer setup, I choose Elite to

be able to stream CD-quality FLAC. You are also asked for your billing information. However, the first 30 days are free and you can cancel at any time. Deezer has a useful feature that TIDAL doesn't have—Track Information displays FLAC when it is being streamed. It seems most of Deezer's music is available in that format. The third icon in the right column shows Deezer as an added provider and Search Deezer appears in the upper right corner Search box.

You can choose more than one service and an icon for each will be added to the right-side list. Radio service is available through TuneIn. According to Deezer's online information, more than 100,000 worldwide radio stations are accessible. I have not tried this because I suspect none stream very high-quality audio.

It's also easy to stream music from your computer or a network-attached storage (NAS) device. This is the Windows setup, but the Mac setup is similar. Click Manage in the top menu bar and the select Music Library Settings. Then click Add. If your music library is not in your My Music folder, select the correct folder. Your music library will be scanned and indexed so you can easily find what you want. Clicking the My Music Library icon in the right-side list will let you make your selection from a variety of options.

When you add or remove library music, rescan it by doing the following: Click Manage again and then Update Music Library Now. Click Yes. Your library will be rescanned and indexed.

Listening and Measuring

Next up was a listening test and I was impressed. But, as usual, I wanted to take some measurements of the streaming quality.

The Connect box has a TOSLINK optical digital output and I have a sound card with an optical input. It seemed better to use a digital link to avoid the DAC in the Connect and the ADC in the sound card. But I got rather mixed results—some of my recorded file comparisons using the DiffMaker software were much better than others.

So, I switched to an analog link. I used signal splitters at the Connect analog RCA output connectors and a stereo cable to the sound card's line input. However, I got the same mixed results. Some of the results are tabulated in **Table 1**, but the actual variation was much higher.

My theory was: The streaming channel is sometimes quiet and sometimes noisy. I needed some measurement evidence to support this hypothesis, but it is logical, given the unpredictability of the Internet and the worldwide web. I don't know



Photo 2: These are the Sonos-Connect's rear panel connectors. (Photo courtesy of Sonos)

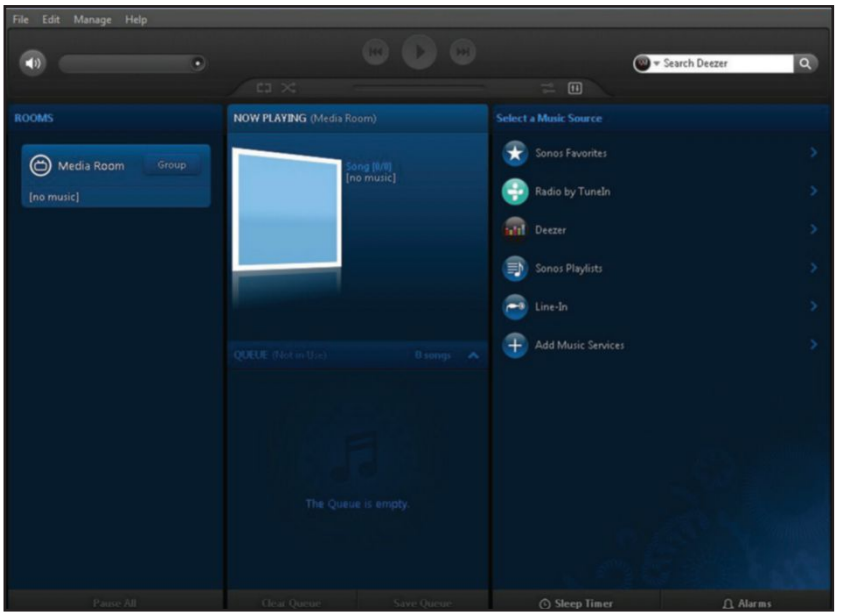


Photo 3: The opening screen shows that Deezer has been selected as the music provider by the Deezer icon in the right-side list and the words Search Deezer in the upper right Search box.

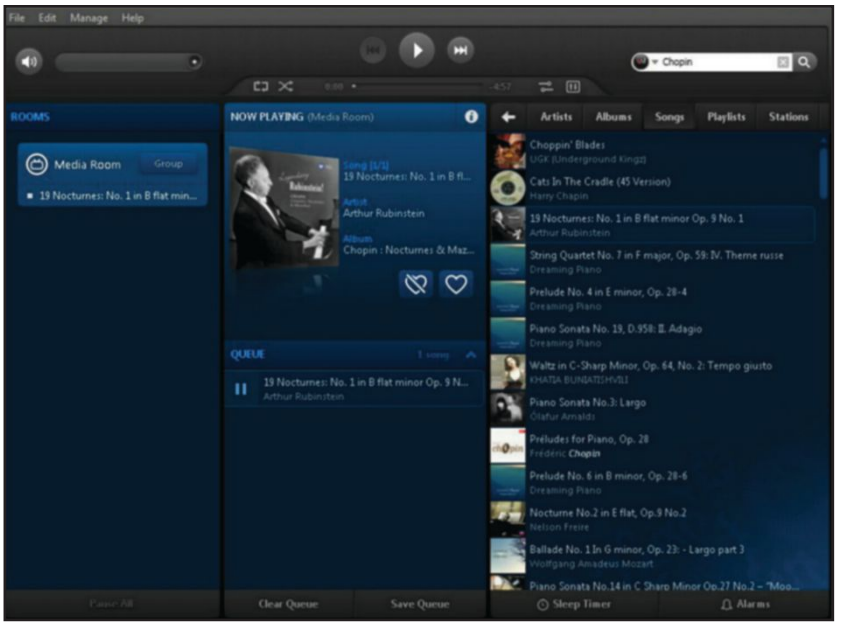


Photo 4: Playing a music selection is easy. I clicked Songs in the right-side menu bar and then entered Chopin in the Search box. I chose one of the songs, a nocturne, and clicked Play Now. The album cover and information is displayed in the center panel and the song is added to the center panel list. A Playlist can be built by selecting additional song titles.

Media	Sample Rate Error	Left Null Depth	Right Null Depth
Frédéric Chopin, "19 Nocturnes: No. 1 in B flat minor," Arthur Rubinstein, *Chopin Nocturnes & Mazurkas*	0.016 ppm	49 dB	48.8 dB
Miles Davis, "Stella by Starlight," *Kind of Blue* (Legacy Edition)	1 ppm	37.2 dB	37.1 dB
Willie Nelson, "Always on My Mind," *Always on My Mind*	2.61 ppm	37.3 dB	37.3 dB

Table 1: A comparison of the average sample rate errors and channel null depths for the three measured music files shows non-ideal numbers, but my listening tests were not affected. Apparently, the quality of the web transmission does not remain constant, sometimes introducing bit rate jitter and adding clicks. My listening tests were not affected, but this doesn't mean a poorer quality could not occur.

where the Deezer servers are located, but it makes no difference because the routing can be switched from moment to moment.

A streaming session is shown in **Photo 4**. Finding music is easy in Deezer. For example, click Songs in the right-side menu bar and then enter an artist or composer's name in the Search box. I entered Chopin and picked one of his nocturnes from the right-side list.

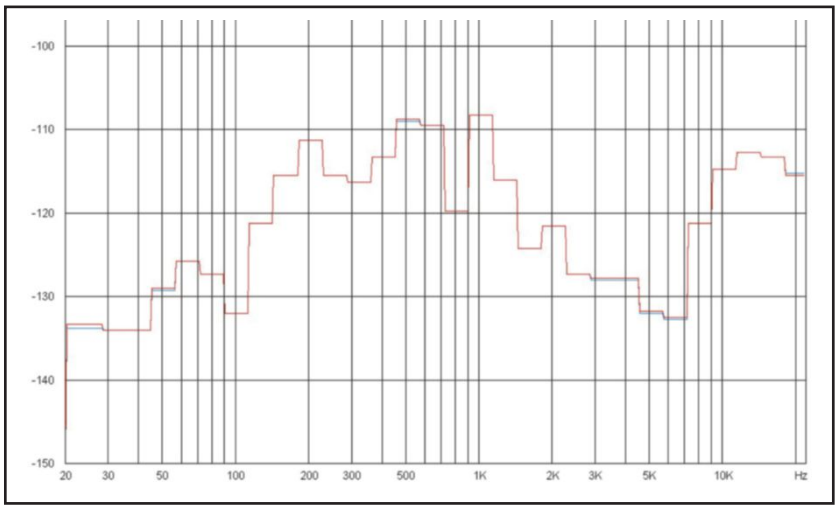


Photo 5: This third-octave frequency spectrum of a Chopin difference .wav file shows good correlation (cancellation) between the two recorded .wav files. All frequency components are below -100 dB.

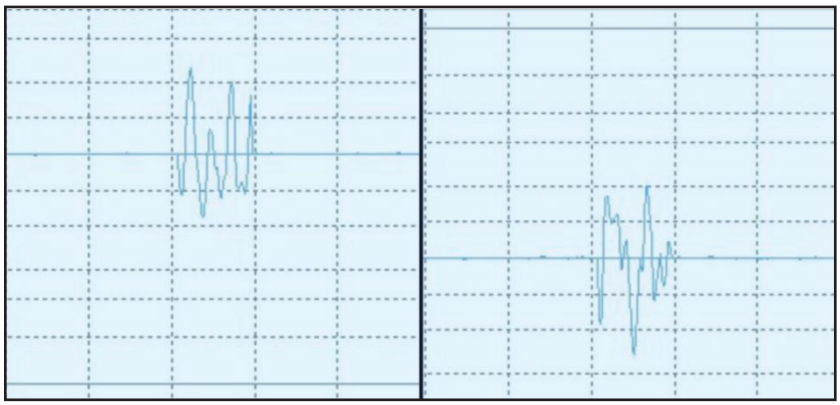


Photo 6: This screenshot is a click in a Willie Nelson difference .wav file. The click was expanded in time and amplified by 20 dB. Although prominent in the difference file, it can't be seen or heard in either of the recorded .wav files that produced this difference.

For each of the music selections shown in **Table 1**, I used the free Audacity software to record several playings over a couple of day's time and compared them with Diffmaker. A higher null depth number means the compared tracks are more alike.

Diffmaker also generates a track difference .wav file that can be examined in an audio editor. The Frédéric Chopin tracks with their 49 dB null depths were just flat lines in the editor. So I scanned the difference file in Spectrogram and got the spectrum shown in **Photo 5**—it's very quiet.

The Willie Nelson tracks were noisier than Chopin with several large "spikes" in the difference file.

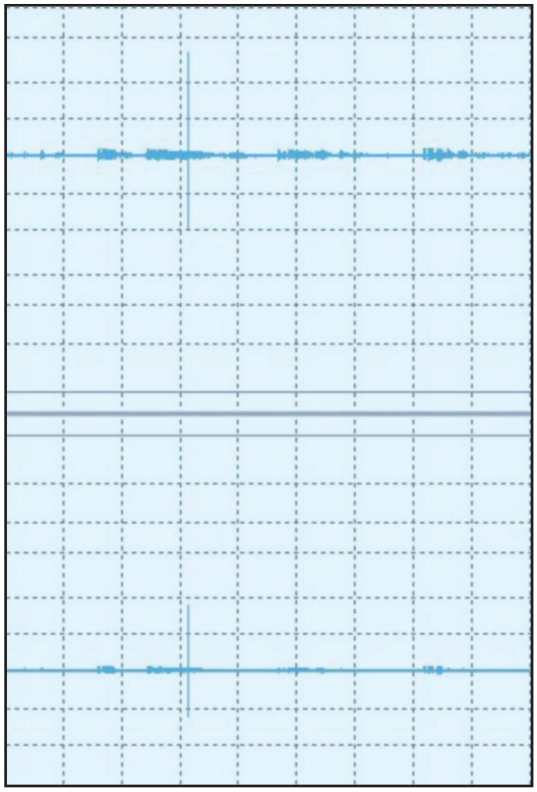


Photo 7: A click and some feed-through "fuzz" in a Miles Davis difference .wav file is shown here. Time expansion was not done, but the file was amplified by 6 dB. I suspect the fuzz was caused by intermittent bit rate jitter in one of the recorded .wav files, which caused incomplete cancellation in the difference file. None of this was evident in my listening tests.

Sound thinking in coaxial design



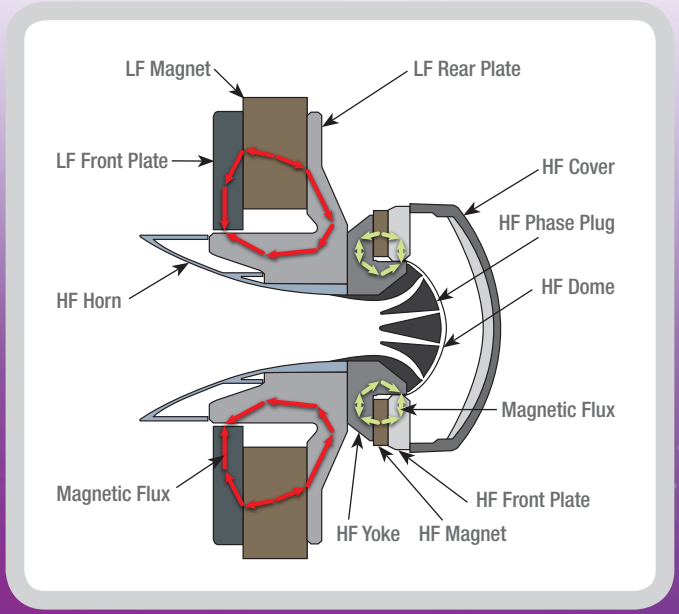
How a common magnet motor delivers a better sounding, lightweight, full-range speaker

Coaxial drivers have long been chosen as an alternative to two-way speaker systems, delivering benefits in size, weight and off-axis response. By concentrically aligning the low and high frequency components, coaxial loudspeakers act as a single source.

Moving the concept forward, Celestion's FTX coaxial range features fully combined LF and HF components that are powered by a Common Magnet Motor Assembly - i.e. using the same magnet for both elements. This enables the voice coils and hence the acoustic centres of the two drivers to be brought much closer together. The end result is a significant improvement in signal coherence and time alignment, and the most natural sounding audio reproduction.

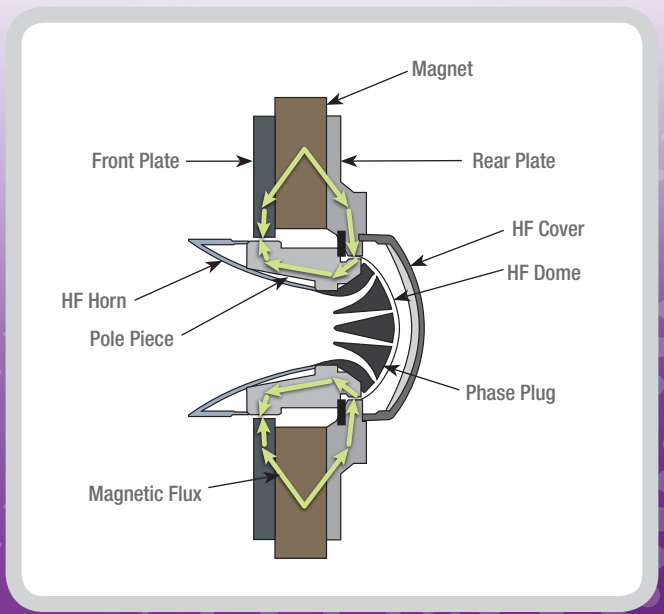
The magnetic flux is optimally distributed between the two elements (calculated using FEA), to achieve the best balance between HF and LF response (see diagrams below). Additionally, the use of a single magnet assembly also means lighter weight and a more compact profile, compared to more conventional dual motor designs.

Specify the innovative choice - Celestion Pro Audio Drivers. See our full range at celestion.com



Traditional Coaxial Driver

With a magnet for each driver, the traditional coaxial design is heavier and the voice coils are further apart.



Celestion Common Magnet Motor Design

Celestion's single magnet design is a more compact, lighter-weight alternative with significant improvements in signal coherence and time alignment.

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CELESTION

MusicScope WAV File	"M"	"S"	"I"	True Peak	PLR
Frédéric Chopin	-16.1 LU	-20.9 LU	-26.4 LU	-8.4 dB	5.2
Miles Davis	-15.2 LU	-16.4 LU	-22.6 LU	-7.9 dB	7.2
Willie Nelson	-16 LU	-17.1 LU	-22.6 LU	-8.4 dB	7

Table 2: The three measured music files were surprisingly close to equal loudness as indicated by the "S" column, the 3 second sliding average. The Chopin track was a little less loud but not enough that I had to adjust the volume control. The Peak-to-Loudness-Ratio (PLR) numbers are in the right range for good dynamics.

One of the spikes is shown in **Photo 6**, expanded in time and amplified by 20 dB. Although seemingly large in this photo, it was inaudible when listening to the music track.

The Miles Davis tracks were also noisy, with spikes and "bleed through" as shown in **Photo 7**. This track was amplified by 6 dB to make the "fuzz" more noticeable. I suspect the fuzz was caused by intermittent bit rate jitter, which makes the two recorded tracks not quite cancel. Again, the distortion was not audible. So, my next question was: Does Deezer use any loudness normalization?

Loudness

Although Sonos recognizes loudness tagging in FLAC file metadata, Deezer does not use it. FLAC files have a metadata field named ReplayGain that can be set to a value that sets the playback volume. The file must be scanned by software to find a loudness number that can be used to calculate ReplayGain.

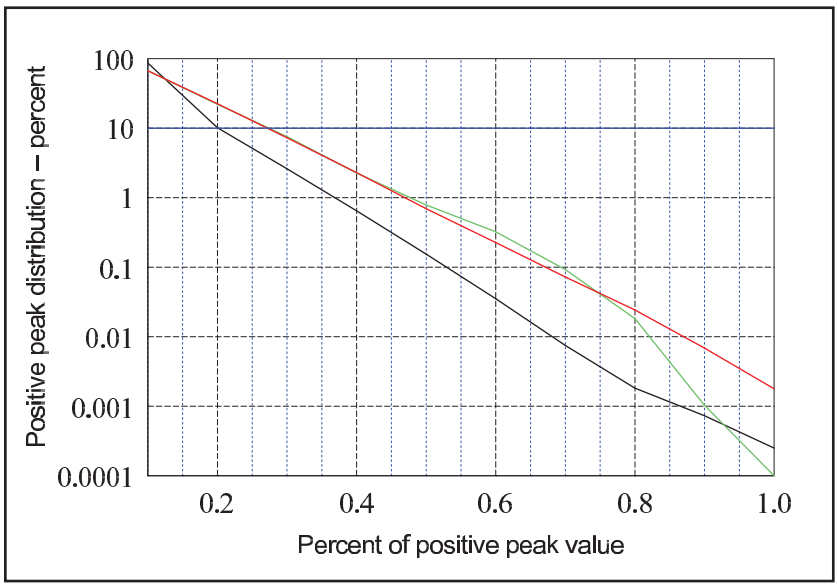


Figure 1: The dynamic range of the three measured music files are very good with little sign of compression. The black line (Frédéric Chopin) is the best, with green line (Miles Davis) and the red line (Willie Nelson) close behind. The blue line white noise is added for comparison, it has zero dynamic range.

To go any farther with this would be off subject, but I intend to write an article on "Loudness Revisited" because many interesting things are happening.

Table 2 presents the loudness measurements for the same tracks used in the music quality measurements. I have found the "S" value (3 second sliding average) to be a fairly good indicator of subjective loudness. This shows the Miles Davis and Willie Nelson tracks to have about the same loudness and it seemed that way to me in my listening tests. The Frédéric Chopin was a bit softer but not enough to need a volume change. Apparently just coincidence because I didn't know it would work out this way.

Dynamic Range

I again used my dynamics.exe program to graph the dynamic range of the three music files (see **Figure 1**). All have good dynamics with little sign of compression. The black line (Chopin) is the best with the others nearly as good. It appears that Deezer is just streaming tracks "as they are" with no processing. The blue line is white noise for comparison – it has no dynamic range. The dynamics program is again included in the Supplementary Material as dynamics.zip. This file also includes the "C" source, library files and documentation.

I am very pleased with the Sonos-Connect, Deezer combination. It's an excellent way to enjoy stereo music. A .pdf copy of the Sonos-Connect User Guide is also included in the Supplementary Material.

Project Files

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

Sources

TIDAL
Aspiro AB | tidal.com

Audacity
Audacity | www.audacityteam.org

Deezer
Deezer | www.deezer.com

Audio DiffMaker
Liberty Instruments, Inc. | www.libinst.com

SONOS
Sonos, Inc. | www.sonos.com

MusicScope
XiVero GmbH | www.xivero.com
(Windows & Mac)

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Audio Praxis—Amplifier Series

ETAL Anaview ALA0080 Class-D Module

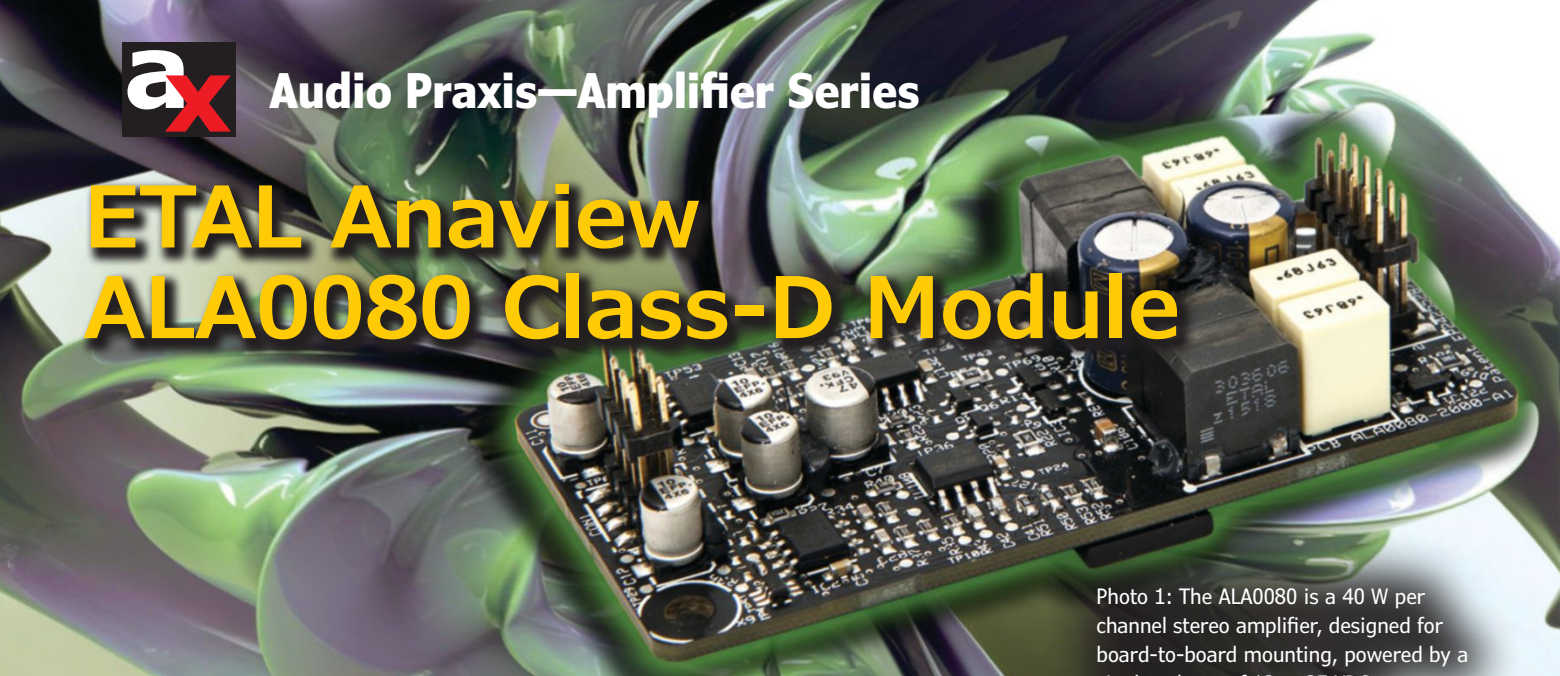


Photo 1: The ALA0080 is a 40 W per channel stereo amplifier, designed for board-to-board mounting, powered by a single voltage of 12 to 25 VDC.

In this article series, *audioXpress* explores the history, recent developments, and new products from some of the most innovative amplifier platform companies serving the audio industry. This month we focus on the Swedish brand Anaview, now part of the ETAL Group, and its latest generation amplifier modules—in particular the surprisingly small ALA0080 Class-D module.

By
Ward Maas
(The Netherlands)

Class-D amplifiers have been on the market for quite some time. Gradually, they started to be regarded as workhorses. Typical environments in which they were used would be pro audio or low-to-mid-end consumer products (e.g., TV sets). You would find them where lots of power is required and where high-end quality was not a primary concern. However in the last couple of years more and more claims were made that, with the ease of power, the audio quality had also improved. In fact, many manufacturers of new-generation Class-D amplifiers claim these amplifiers are now able to easily compete with the best.

We have been following the developments on this front with great interest. An impressive number of important product introductions took place in 2016. On the corporate side of things, 2016 was also a year of changes with mergers, buyouts, and acquisitions.

During the 141st Audio Engineering Society (AES) Convention in Los Angeles, CA, Swedish company ETAL introduced new Class-D products from Anaview, shortly after the company's acquisition in August 2016 (see Sidebar). Among the new solutions was a compact Class-D amplifier module that generated lots of curiosity. Not only was the footprint of the new ALA0080 module just 68 mm × 33 mm and 18 mm in height, but the power density at 2× 45 W at 6 Ω was remarkable. Unlike other Anaview Class-D amplifiers, this new module doesn't have a power supply on the PCB. With a recommended power supply voltage between +12 V and +25 V, this makes the ALA0080 an ideal candidate for systems integration (see **Photo 1**).

And, the tiny ALA0080 serves a variety of applications. It is designed for board-to-board mounting and modern solutions such as Power-Over-Ethernet (PoE) devices for commercial network audio applications where audio and power are provided on a single Cat-5 cable, as well as battery-powered and rechargeable products—which is one the fastest growing categories in the audio industry.

Stefan Ruuth, Sales Manager at ETAL, explains, "In terms of size-performance, the Anaview ALA0080 is unmatched, delivering amazingly clean and room-filling sound from a small module. We've already seen a lot of interest from designers of networked audio devices, portable audio devices, docking stations, audio receivers, powered speakers, and residential audio systems. It's perfect for

Anaview ALA0080
ALA series stereo amplifier
ETAL Group AB
Box 8064
SE-163 08 Spånga, Sweden
+46 (0)8 759 35 00
www.anaview.com
anaview.eu@etalgroup.com
anaview.us@etalgroup.com

applications that need great sound but where space and/or power are limited."

After reviewing the specs, you will find figures such as residual noise on a -120 dB for all frequencies under 24 kHz and a total harmonic distortion (THD) of 0.004% at 1 kHz at 1 W. So yes, all the more reason to have a closer look at this little gem (see **Figure 1**). But first, some more background on Anaview and ETAL.

Scandinavian Power

It's not a coincidence that so many companies working in the field of Class-D amplification originated in Scandinavia. Most of these companies know each other, from start-up projects, joint developments, or simply from the college days of their engineering teams.

The principles of Class-D amplification are not that patentable or even that difficult. Getting a circuit alive, and more importantly, keeping it alive, is less trivial. Hence the patents in the Class-D amplifier world are more of the control loop and housekeeping kind. In professional audio's day-to-day field operations, there are often stress situations where the fault condition tolerance of an amplifier is highly appreciated.

Between 2003 and 2009, Anaview developed custom designs, based on its proprietary circuitry and technology, for audio companies around the world. In 2009, it filed two patent applications: Adaptive Pole Control (APC) and Adaptive Modulation Servo (AMS). Both are circuits that strive to optimize the behavior of basic globally self-modulated Class-D topologies, without significantly increasing the parts count. Of course, Anaview was building on that experience when it designed the ALA0080.

With the Class-D maturing market, Anaview teaming up with the Swedish ETAL Group as an independent brand seems a smart move. Not only will ETAL bring the resources and experience of a large company, it also offers the in-house development of inductive components for high-efficiency power supplies and signal applications. Of course, inductive components are a key factor when it comes to Class-D amplifiers.

ETAL develops, manufactures, and sells magnetic components and audio amplifiers to a worldwide market through offices in Europe, Asia, and the US, and through distribution partners in an additional 20 countries. Production takes place at the company's own facilities in Estonia and Sri Lanka, as well as production partners in Asia.

The Class-D amplification segment needed to step up—not only from an amplification design point of view, but with housekeeping items such as over/under voltage, overcurrent, and over-temperature protections. In the early days of Class-D, products were not always compliant with the electromagnetic compatibility (EMC) regulations. Certainly for systems integration, it is a good starting point if all the components are made according to the standards. And of course, the complete system has to be checked as well. When we look at the specifications of recent Anaview products, that evolution becomes immediately apparent.

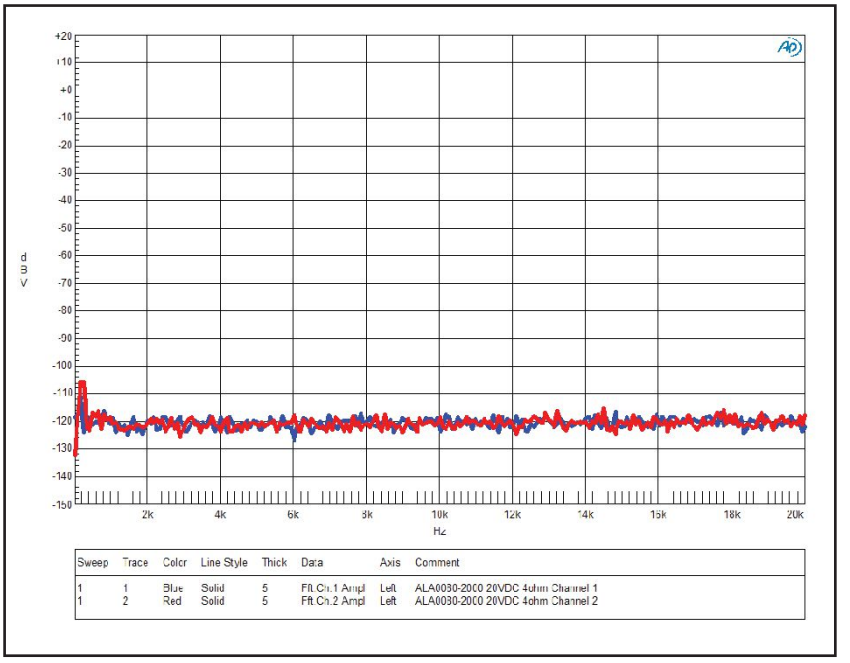


Figure 1: Anaview ALA0080's Fast Fourier Transform (FFT) residual noise at 20 VDC was 4 Ω, for both channels.

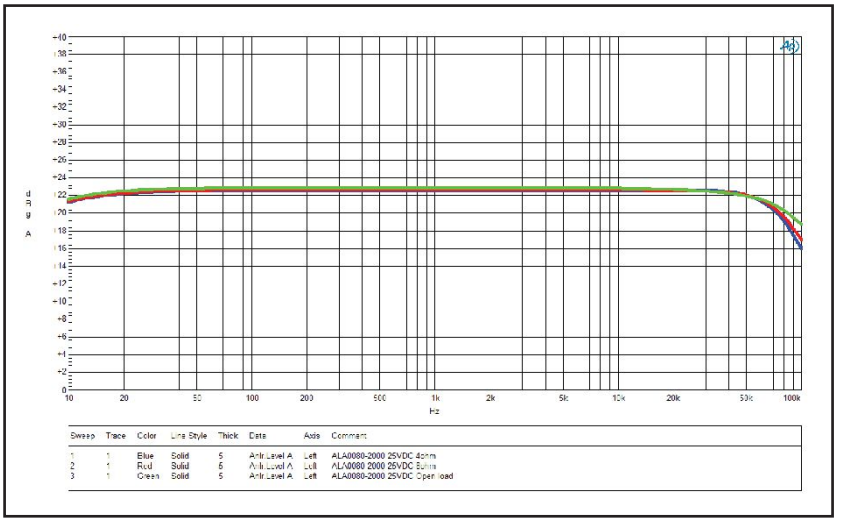


Figure 2: The Anaview ALA0080 frequency response shows it has a -3 dB point at about 80 kHz.

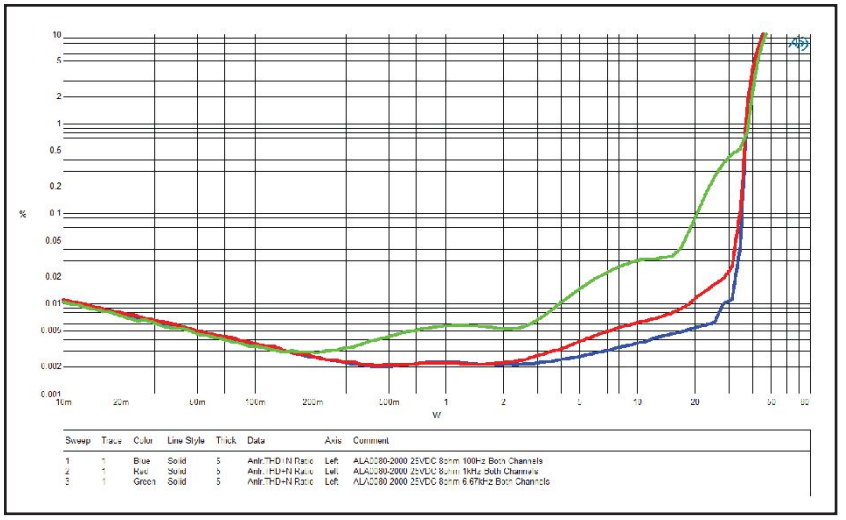


Figure 3: Anaview ALA0080's THD vs. Power was 25 VDC at 8 Ω.

The ALA0080

In the early days of Class-D amplification, the frequency response graph often revealed a serious weakness in amplifying the higher spectrum of the audio bandwidth. A -3 dB reading below 20 kHz was a clear indication that its use was limited to the bass region. In fact, it was a sign that product development was not finished. **Figure 2** shows that the ALA0080 has a -3 dB point at about 80 kHz. No more discussion about usability with respect to this.

Another item on the specs wish list for an amplifier is absolute silence when no music is played. **Figure 1** shows an almost ideal graph of the noise floor, at about -120 dB. The times are gone when the loudness of the residual noise was an indicator for the actual music sound level, which could be expected. In our practical tests, this amplifier proved to be absolutely silent.

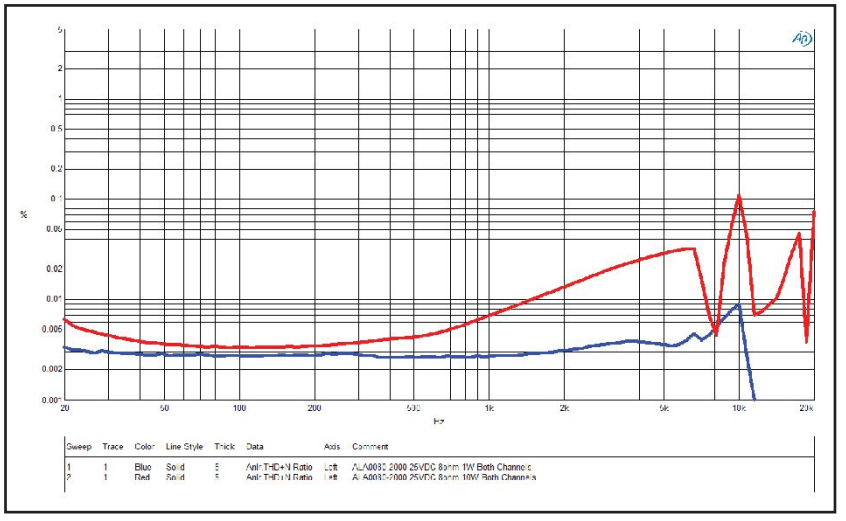


Figure 4: Anaview ALA0080's THD vs. frequency was 25 VDC at 8 Ω, for both channels

Looking at the THD vs. Power graph, we see some artefacts that can be described as a Class-D signature (see **Figure 3**). The left side of the graph is dominated by noise, the right side by clipping. The middle part (certainly for higher test frequencies) depends upon the topology. I have to say that these are nice figures. A THD of 0.004% at 1 kHz is a successful result. In the critical bandwidth from 20 Hz to 20 kHz, everything remains below 0.1%. The THD vs. Frequency graph also shows some effects that can be found in Class-D designs (see **Figure 4**). The artefacts on the higher frequencies are a clear design signature. You do see the filter and the loop design.

We have to remind ourselves that we are looking at these graphs with analog amplifiers in mind. However, a switching amplifier doesn't behave like an analog amplifier. The design challenges are completely different. The effect of the output filter, the load, the effect of dead-time, and switching the output devices on and off provide design questions that need answering. The choices made in that process can be seen in these graphs. For instance, a steep (non-ideal) passive LC filter is between the pulse width modulation (PWM) signal and the (variable) load (loudspeaker) removing the switching frequency and harmonics. So, controlling the audio signal at the loudspeaker requires some non-trivial feedback mechanisms, preferably before and after this filter.

The interesting question is how the end result will sound. Or better yet, not sound. In the end, if these differences from the original signal are low enough, we will not hear them. In that respect, analog and Class-D amplifiers have something in common. The THD figures at 1 W are an accomplishment.

Device Under Test

With this knowledge, it's time for a usability test. For the test, we used a tiny demonstration amplifier with the ALA0080 built in. This amplifier was so small that the four output terminals, the two inputs, and a tiny on/off switch just fit on its back side. **Photos 2–4** provide a good size perspective.

Despite the small size, we could see the PCB is tightly packed with components. To reduce the size, Anaview looked at every function in the design, optimizing the part count and size for each. Sometimes that was done by replacing discrete solutions with small footprint ICs, and sometimes the opposite (i.e., replacing large footprint ICs with compact discrete solutions). Anaview also designed its own custom-made output inductor—optimized in size and thermal performance for this application. The complete unit has a maximum height of 18.5 mm.

One of the advantages of Class-D amplification is its high efficiency. Efficiency values are around 90%. With no heatsinks (weight!), switch mode power supplies provide all the desired power in light small enclosures (e.g., a module integrated on a powered loudspeaker, a 19" rack unit, or a compact instrument amplifier). In fact, Class-D amplification can be used anywhere we need audio amplification. Normally, Anaview products come with a built-in switch mode power supply. However, the ALA0080 has no stringent requirements in terms of power supply for integration in a system. For our tests, we used a simple small laptop-style power supply at Moving Studios in Almere. The studio is a bit hidden in an industrial area of The Netherlands, but we could make all the noise we liked. Moving Studios is run by a group of music aficionados and is rapidly becoming a focal point of the Almere music scene (see **Photo 5**).

Hands-On Experience

First, the power-on. No thumps, bumps, clicks, or noise. And then, there was music. In fact, a bit disappointingly, the ALA0080 did just what it was supposed to do. Nice!



Photo 2: I placed two Lego figures next to the Anaview ALA0080 Class-D amplifier to show its size.



Photo 3: The Anaview ALA0080 is so small that the connectors can barely fit on the back plate.





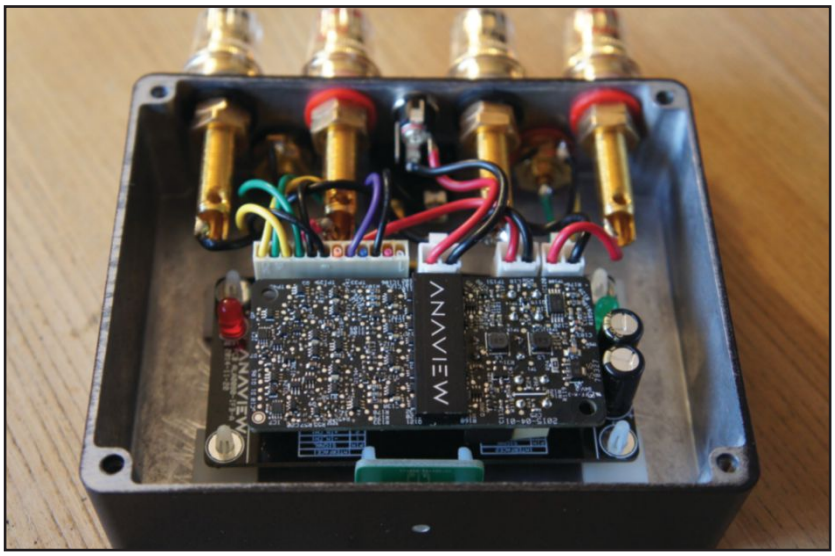


Photo 4: Inside the enclosure, you can see the Anaview ALA0080 is even smaller.



Photo 5: We pushed the Anaview ALA0080 to its limits at Moving Studios, with eight Electo-Voice PA speakers.



Photo 6: We challenge you to find the amplifier.

At first, we just listened to a selection of CDs and homemade live recordings that were available. Very enjoyable. No noise, very clear, and a very tight bass. Just rock solid.

After spending some time listening to music, we decided to get serious. As load, we had eight sets of medium-size Electro-Voice (EV) systems. We started with 8 Ω and added an extra EV set in parallel each time. With a normal louder volume, there was no problem driving all eight speaker sets in stereo mode.

What is always amazing is the fact that the amplifier did not even get warm to the touch. Of course, we wanted to test the limits. While playing Beyonce's *Lemonade* CD, we were finally able to get the amplifier to momentarily stumble. The amount of bass energy in that recording compared to the voice level was remarkable. Without getting into that debate, I can only imagine that for this type of content you would want to have some extra power.

But let's be fair. When will you need this tiny amplifier to power a complete live sound PA system? With a footprint of just 68 mm × 33 mm, this is an ideal stereo (!) amplifier for almost every medium-size amplification need in any application. Definitely a high-quality candidate for a lot of brands that are looking for something this good in such a small package (see **Photo 6**).

With the power supply range, the low noise, the small size, the output power, and most importantly, the fact this amplifier is very efficient, the ALA0080 is also a good candidate for automotive use. With the impressive volume levels we were able to get at low impedances, we still had reasonable headroom remaining. A multichannel set powered with the ALA0080 would be an attractive product, sound wise, and it could be small and living-room friendly.

Overall Opinion

I was very impressed with the high-quality levels of this Class-D amplification solution. There are no more excuses with Class-D amplifiers, as we can attest with this Anaview product. Plus, it also shows

About the Author

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

that high-quality Class-D amplifiers will conquer the audio market in a lasting way.

One more (big) thing. Anaview also sent us another product to test—the AMS1000, which can be better described as a high-end powerhouse: 2 x 500 W, at 2 Ω. We will share our results with that slightly larger (in size) amplifier in a separate article. Let's just say that the Moving Studios' manager was highly interested in this amplifier.

Author's Note: We would like to thank Andree Troost for his support in general, and his music and his ears in particular. I would also like to thank Victor van Zeeland and Moving Studios for generously offering their support.

ETAL Group and Anaview Integration

In mid-August 2016, ETAL Group, based in Stockholm, Sweden, announced it was renaming Anaview, the manufacturer of Class-D amplifiers, as ETAL. The well-recognized and regarded Anaview name continues as the product line name for the Group's Class-D amplifier range. KAMIC Group, a privately held conglomerate of companies in Sweden, and the parent company of ETAL, acquired Anaview in 2013, managing it as a unit of ETAL Group, an international supplier of inductive components.

Dan Phelan, President of ETAL commented, "Anaview has made excellent progress since we acquired the business, launching further innovative products and winning new customers. The combination of Anaview's design flair and experience and the strong ETAL production and distribution organization has proved extremely powerful. To reflect the growing integration of the business into the group we have decided to change the name of the company to the group name: ETAL. The Anaview name is extremely well recognized and loved in the international audio market as a brand of high-performance amplifiers, so we will continue to use it as a product name."

"The extensive experience of the design team behind Anaview Class D amplifiers is enabling audio companies worldwide to meet the highest demands for audio quality, cost efficiency, and green power, and achieve the shortest possible time to market. In creating their designs, they benefit from ETAL's expertise in transformer and inductive component technology. Inductive components in Anaview amplifiers are designed specifically by ETAL for Anaview and manufactured by the Group. The name change will make no difference to the exceptional design and quality of Anaview Class-D amplifiers. The team will continue with its keep-it-simple design philosophy using premium components for a robust and great sounding amplifier system," he added.

Since 1968, ETAL Group has been a leading supplier of in-house developed high-performance magnetic components, primarily transformers and inductors, for the telecom, power technology, automotive, and defense industries.

For more information, visit www.etalgroup.com or check out www.anaview.com.

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Dynaco ST-70 Series 3

The Inside Story—A Long Road



Since acquiring the legendary Hafler and Dynaco brands in 2014, Radial Engineering has been carefully working on the Dynaco ST-70 Series 3 tube amplifier. In an effort to create a truly worthy successor of one of the most successful tube amplifiers of all time, its design team has considered all the options needed to meet the expectations that come with David Hafler's work.

By
Dan Fraser
(Senior Design Engineer,
Radial Engineering, Ltd.)

As a veteran audio electronics engineer, I welcomed the news that Radial Engineering had acquired the Dynaco intellectual property and brand. I was one of the last engineers to receive a basic electronics education on tubes back in the 1960s. I repaired and rebuilt Dynaco products back in the 1970s, and though I worked for many years on Class-D PWM amplifiers, DSPs, digital audio distribution, and other leading-edge state-of-the-art technology, I immediately jumped on the ST-70 as the best introductory product for the line and something I really wanted to do. How hard could it be to bring it back, considering we already had the schematics and the sample amplifiers to measure?

Turns out, it was a lot harder than you would expect as the Dynaco output transformers are magic when it comes to high-frequency clarity and accurate build information is not to be had. When we conducted listening tests, we found that with the original tube power supply the bottom end was thin when played with modern music. And, it is the power supply in an amplifier that determines the quality of the bottom end. The power supply delivers the punch that modern music requires. Good low-frequency response is not enough.

Then, there was the question of whether to offer it as a kit. We debated this as it would save the cost of getting safety certification and reduce the costs in production. But in today's onerous legal environment, sending out products with high voltages that could potentially be used by amateurs made the legal risk too great. Instead, we chose to sell the ST-70 as a completed product and undertook the costs to have it properly safety certified.

Next, we considered the countless upgrades and modifications that were introduced by the DIY community over the years, many of which were very good. There is no sense in making a faithful reproduction of the old amplifier that would need to be modified out of the box.

Modern Parts

For the ST-70 Series 3, Dynaco has made extensive use of the most modern parts including: precision metal-film resistors, poly-composition capacitors, and miniaturized high-capacity power supply electrolytics (unavailable at the time of the original design). Newer parts are also more conservatively rated than some of the original parts. In compliance with general practices, a heavy-duty three-pin detachable grounded power cord is provided. Signal ground is the same as electrical ground and chassis. And all the signal connectors are gold-plated.

Tens of thousands of original Dynaco tube amplifiers are still in use around the world. In fact, more quality music systems have included Dynaco tube preamps and power amplifiers than any other combination. The original Dynaco Stereo 70 is a true classic still regarded with awe by audio enthusiasts worldwide. This "new ST-70 Series 3" reflects the illustrious history of Dynaco's involvement with vacuum tube designs as well as the latest advances in vacuum tube technology. We like to think of the ST-70 Series 3 as the update that David Hafler would have made if he were alive today (see **Photo 1**). Besides, there was no point in precisely recreating the old amplifier and adding some upgrades. If people wanted the old amplifier with upgrades, the aftermarket upgraders do an excellent job restoring and upgrading the old stock of Stereo 70 amplifiers at a better price than a new amplifier will cost. We feel people who are buying new want a better amplifier, but one that is still based on the work of David Hafler.

Styling

Next up…The styling came into question. The original ST-70 was a pretty utilitarian box that was okay in the 1950s. There have been many great looking replacement housings made for the ST-70 by third-party vendors, and really, the original looks pretty plain.

Today, with numerically controlled metal fabrication gear, the cost of making something look good is not a lot more than the cost of a plain box. So, we decided the case needed an update and our people created a better-looking amplifier that still retains the original "base-and-cage" design, yet brings a more modern and sexy look. We made it a little bit wider to get the necessary clearances for the high voltages based on current regulations and to match the upcoming Dynaco ST-1 tube preamp. Both are about 17" wide to follow the generally accepted standard in use by many manufacturers today.

The front panel is especially modern. It features adjustable recessed bias controls with pairing LEDs to optimize each tube and separate channel volume controls enable the user to balance the output for odd-shaped rooms. The rear panel is made with ultra-high-quality gold-plated RCA connectors and heavy-duty gold-plated "No Touch" speaker binding posts—compatible with modern safety standards. The modern IEC power cord connection is now capable of both 115 V/230 V operation with a simple rear-panel switch.



Photo 1: The new Dynaco ST-70 Series 3 tube amplifier intends to be an updated version, compliant with today's demanding global regulatory safety requirements.

To improve the amplifier's unit-to-unit consistency, we needed to eliminate the majority of the hand wiring and this meant using a PCB for the internal components and wiring to the output tubes as well. This is a modern glass-epoxy board with extra heavy traces that meet the modern standards for safety. The ST-70 Series 3 still employs 100% hand-inserted full-size discrete components—no surface-mount parts are used.

Once all the pieces were in place, Radial Engineering CEO Peter Janis unveiled the Dynaco ST-70 Series 3 at the 2016 High End show in Munich,



Photo 2: Peter Janis, CEO of Dynaco's parent company Radial Engineering, unveiled the Dynaco ST-70 Series 3 at the 2016 High End show in Munich, signaling the return of Hafler and Dynaco.

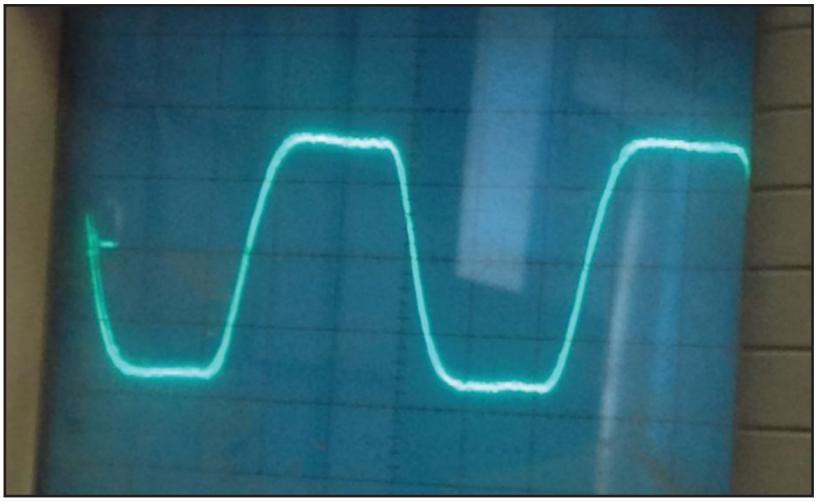


Photo 3: Here is the square wave response of the original A470 transformer at 20 kHz (1 W output).

signaling the return of the Hafler and Dynaco brands (see **Photo 2**). However, it is important to understand all the work that went into the new Dynaco ST-70 Series 3.

Power Supply

The original ST-70's power supply has long been the design area most vulnerable to legitimate criticism. We addressed this problem as follows:

- We designed an entirely new and larger dual primary power transformer with improved line regulation (due to added iron in its core). This improves bass response and enables the ST-70 Series 3 to be set for either 115 V or 230 V operation for global operation.
- The original, long discontinued 5AR4 tube has been replaced by solid-state rectifiers. This reduces the 5AR4's inherent losses while improving regulation and eliminating a failure-prone tube in the process. The benefits include more low-end impact, greater efficiency, and reliability.
- The original power supply filter capacitance has been made far larger; as large as we could fit in the space available while observing required safety clearances. This advancement has been desirable for a long time but has only recently become practical with the evolution of next generation capacitors that have been developed for computer power supplies and other switch-mode circuits. This benefits the ST-70 Series 3 with improved low-end dynamics and damping factor.
- The troublesome selenium rectifier used in the bias circuitry has also been replaced by a solid-state rectifier with additional supply capacitance. This of course improves reliability.
- The bias reference voltage is now derived from an IC voltage regulator for consistency. This improves the stability of bias settings for more consistent tube performance.
- The filter inductor has been reduced in value to complement the increase in filter capacitance with a resulting reduction of DC resistance. This makes more voltage available for use by the outputs and somewhat higher peak power output for a more impactful bottom end.
- The final filtering stage of the operating B+ voltage has been split into four streams, each with its own RC filter network to greatly enhance channel separation while lowering noise.
- The PCB has been especially carefully routed and optimized with advanced circuit development software to ensure that ground return currents from the power supply remain separate from the filtered power that operates the output stage and other separate ground returns for the driver stages. This reduces channel crosstalk, lowers overall noise including rectifier noise, and lessens overall low-level harshness or grit for more "air" when listening at low levels.

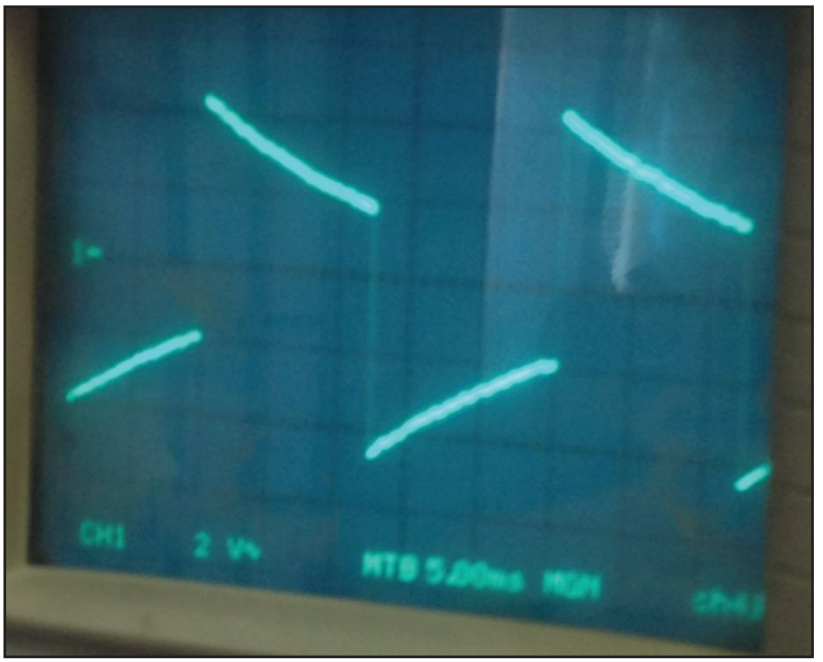


Photo 4: The bottom end is like this at 40 Hz square wave.

All of the changes serve to improve "stiffness" of the power supply by lowering the effective power supply impedance and improving regulation while increasing available voltage for maximum safe power output. This ultimately makes for a tighter bottom end. It is Dynaco's position that full electronic regulation of the plate power supply does not result in cost-effective sonic improvement.

Output Transformer

The output transformer inside the Dynaco ST-70 is often considered to be one of the primary contributors to its legendary sound. Before we set out to build a "new version," we heavily invested in the project by purchasing various ST-70s ranging from the original to the updated. This laid the foundation and set the goal in finding the right transformer for the job.

We began by taking these apart and then scouring the planet to find possible solutions. After testing various "off-the-shelf" output transformers, we found that we could not find any that had the high-end response of the original David Hafler-designed A470. The square wave response of the original transformer at 20 kHz is telling (see **Photo 3**). After reviewing the test results, we were able to confirm that the square wave response was part of what made the original amplifier so well-loved by audio fans. The image shows the square wave response at 1 W output as are all the rest of the tests herein. They are representative of the others we did at various power levels to clipping.

You can see the bottom end at 40 Hz square wave in **Photo 4**. The angle of the tilt indicates the low-end response. The less tilt the better the bottom end. Tilt is caused by both low-frequency roll-off and the ability of the power supply to deliver current at low frequencies.

Figure 1 shows the frequency response of a professionally upgraded ST-70 with the original Stereo 70 transformers in it. The amplifier we used to obtain these results already had upgraded coupling capacitors and a somewhat upgraded power supply.

Our first attempt at a custom transformer provided the 20 kHz square wave shown in **Photo 5**. This is as good if not better than every off-the-shelf transformer we tried. Please note the vertical scales are different. The original is set to 2 V per division while this one is at 5 V per division with 20 kHz at left and 40 Hz below.

However, the bottom end was stunning in comparison (see **Photo 6**). The greatly improved power supply is a large part of the performance, but the transformer was also very important.

For our second attempt—after exhaustive work by our transformer vendor over many months including testing of the original A470—we got the results shown in **Photo 7** at 20 kHz. While the rise time is not quite as fast, that was the trade off to get the excellent bottom end. The issue is that there should be no overshoot and no ringing or distortion of the wave shape. Our listening tests confirmed that the rise time was not an issue. The lack of ringing is what makes the high-end magical. **Figure 2** shows the frequency response with the new transformer.

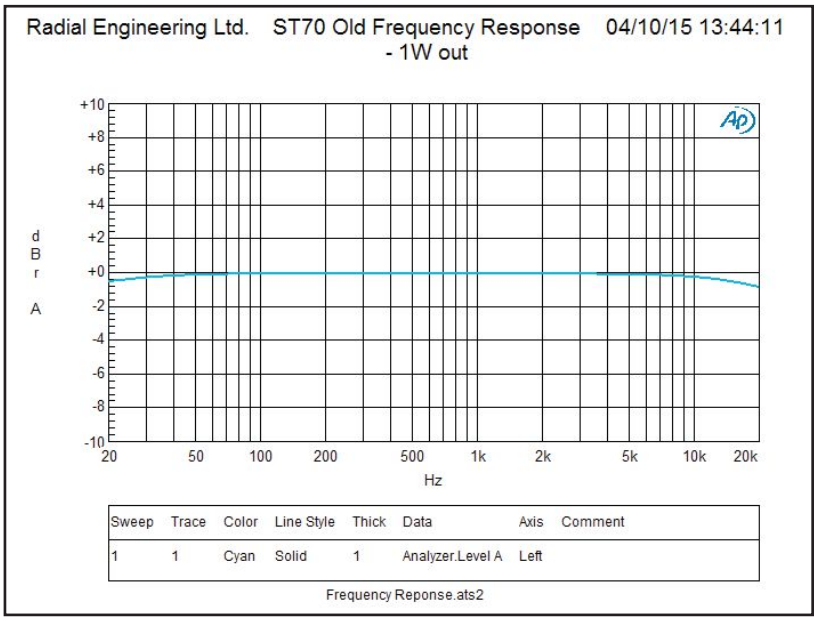


Figure 1: We obtained this frequency response for a professionally upgraded ST-70 with the original Stereo 70 transformers in it.

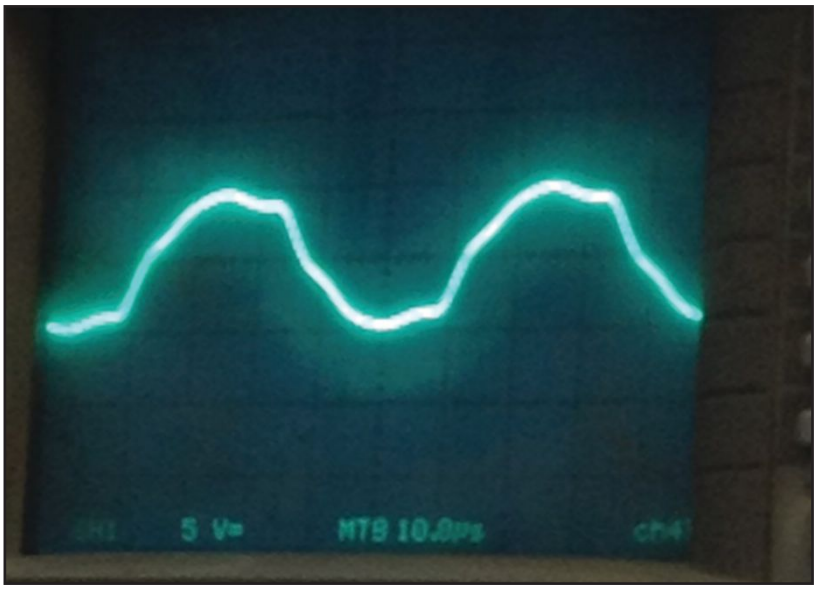


Photo 5: This was our first attempt at a custom transformer.

About the Author

Dan Fraser is one of the last electronics people to receive their basic education on tubes back in high school. A graduate of the Northern Alberta Institute of Technology, in Edmonton, AB, Canada, he has worked for more than 50 years in electronics in Alberta, California, and British Columbia. He built his first power amplifier in 1966 and has designed and built a variety of products for professional and high-end consumer audio in every technology from tubes to pulse width modulation.

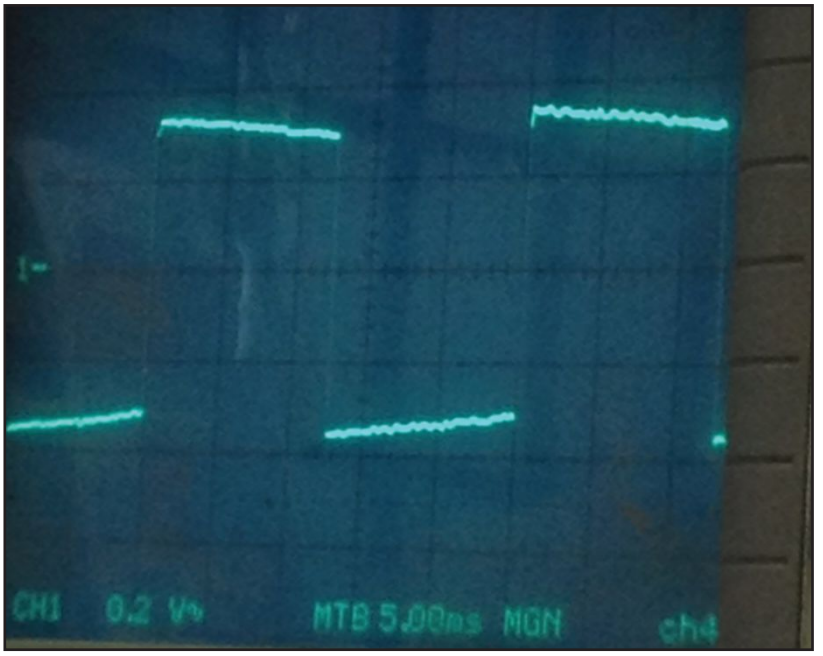


Photo 6: We achieved the bottom end shown here with our custom transformer.

Besides preserving the original transformer's high frequency and square wave responses, our transformer makers have extended the low-frequency response to do justice to current music, recorded with modern recording technology. As far as we can tell, we feel these may be the finest output transformers made for any tube amplifier ever anywhere near this price point.

Two output impedances are now offered, 4 Ω and 8 Ω with a selector switch, and feedback is taken from whichever impedance tap is selected with the switch. Everyone we surveyed felt that a 16 Ω tap was not required and the space that was freed up in the transformer allowed for heavier wire on other windings.

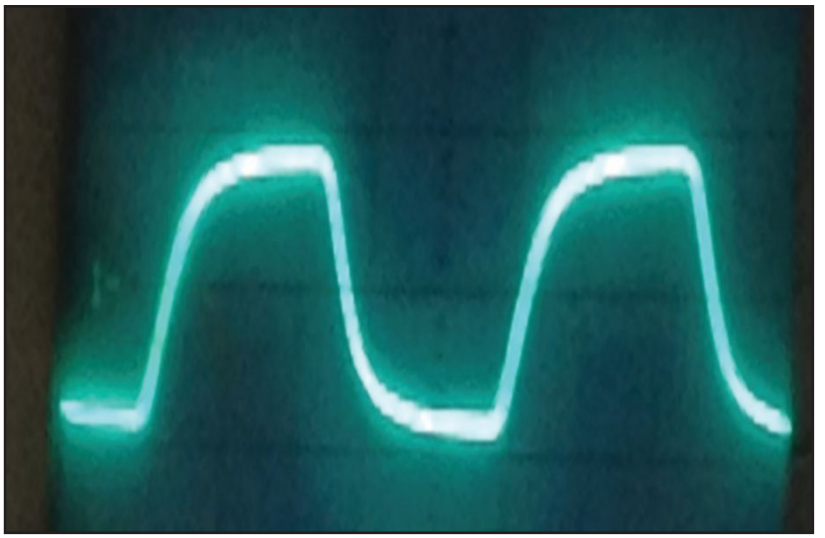


Photo 7: This is the results of our second try with new transformer.

Driver Stage

It is important to reiterate the philosophy behind the development of the ST-70 Series 3. When conceptualizing the product, we asked ourselves a simple question: What would David Hafler do if he were in our shoes today? The answer is simple: He would use the very best components, try to keep the costs under control, and deliver good value.

At this point, we ran into a brick wall. The original 7199 driver tube with pentode and triode sections that was serving as both the preamp and the phase splitter has long been out-of-production and unavailable as are all triode-pentode types except for expensive or hard to come by new old stock (NOS) tubes. David Hafler's approach would have been to find a solution that would enable the ST-70 to be produced without part shortages, while delivering the very best possible sound. So we searched and tested various options until we uncovered the best overall solution.

We recreated the design using a pair of EF86 pentodes and a 12AU7 dual triode as the phase splitter. This worked reasonably well. However, we discovered that the EF86 is only available from a single vendor and we felt the supply source was not secure and without competition, quality and price could be called into question. We built a prototype anyway and called it the 2T.

Upon further digging, we felt that it was likely that this driver circuit was developed by David Hafler to meet a target price point based on the available parts that were obtainable 50 years ago and may not reflect what he would have done today. During his illustrious design history, David Hafler had developed a number of more advanced designs—many of which were never commercially produced.

In fact, we discovered a design that used three dual triode tubes with a 12AX7 as a voltage amplifier and a pair of 12AU7 tubes as a long tail pair to act as a phase splitter. This configuration cost little more than the original design and had very good sound quality. However, the long tail pair phase splitter has unequal gain on the two phases, which then required a trimmer to balance the phase for lowest distortion. To adjust this in the field is difficult, requiring advanced equipment for best results. We built one and called this the Triple Triode (3T) design.

Then, there was the advanced Williamson circuit that Hafler had drawn (see **Figure 3**). This Williamson design used a dual triode as a voltage amplifier and cathodyne phase splitter. Another dual triode is used as an additional voltage amplifier and output tube driver. The cathodyne phase splitter used here and in the original ST-70 had the advantage of being as accurate as the resistors used in the circuit. With today's proliferation of high tolerance 1% resistors,

this made investigating this option completely viable. We called this version the Quad Triode. (4T)

The next step was to build prototypes of all three designs (2T, 3T, and 4T) for comparative listening. To ensure fair testing, all three designs used the same output transformers and power supplies. We set them up for blind testing and conducted multiple tests. To ensure there would be no listening bias or sonic memory to contend with, we built a switcher that instantly switched between amplifiers and carefully made sure each had the exact same output level.

In measuring, we found after warm-up and bias adjustment the triple triode (3T) unit with the long tailed pair phase splitter had the lowest distortion by a very tiny margin, but it had to be laboriously set up with an Audio Precision test system.

The quad triode and pentode-triode designs required no balance adjustment. The quad triode design measured a slightly higher distortion through the critical 500 mW to 5 W range. The pentode-triode design measured to have about twice the distortion of the quad triode design.

In the listening phase of the testing process, we found the sound quality of the pentode-triode design to have the lowest sound quality. Still good but not as good as the other two.

Between the triple triode and the quad triode, they seemed indistinguishable in the mid and high end. Both were excellent. However, the quad triode with the cathodyne phase splitter was felt to have a punchier low end. Likely because the dedicated drivers were able to feed the EL34 grids from identical lower impedances, giving it more punch.

Our listening panel decided that while the quad triode design cost a little more to make the improved sound, the quality was worth it. All of the variations followed the original design philosophy, which is still valid today. The low open-loop gain philosophy was maintained, thereby minimizing Transient Intermodulation Distortion (TIM). TIM distortion is regarded in many circles as the type of distortion to which the ear is most sensitive.

This is why we use the lower gain 12AU7 tubes for all four driver circuit tubes as the 12AX7 had too much gain and the dynamic distortion levels seemed higher. While this approach of lower open loop gain is controversial in some circles, listening tests show a sound quality closer to true tube sound by minimizing the effect of the feedback.

Output Stage

The output tubes remain the fine EL34/6CA7, using the Ultra-Linear principle David Hafler and Herb Keroes developed under US Patent 2,710,312. However the troublesome cathode resistor originally used is now a separate high-quality metal film part for each cathode.

A further improvement in distortion, particularly in the higher frequencies, was achieved by moving the feedback point from the 16 Ω tap of the output transformer to the actual loaded tap by the double pole impedance selector switch. The switch uses one side of the switch to direct the 4 Ω or 8 Ω tap to the output terminals. The other side directs the feedback point to the tap selected and changes the feedback divider ratio so the overall gain remains the same whichever tap is selected. Transient response is improved.

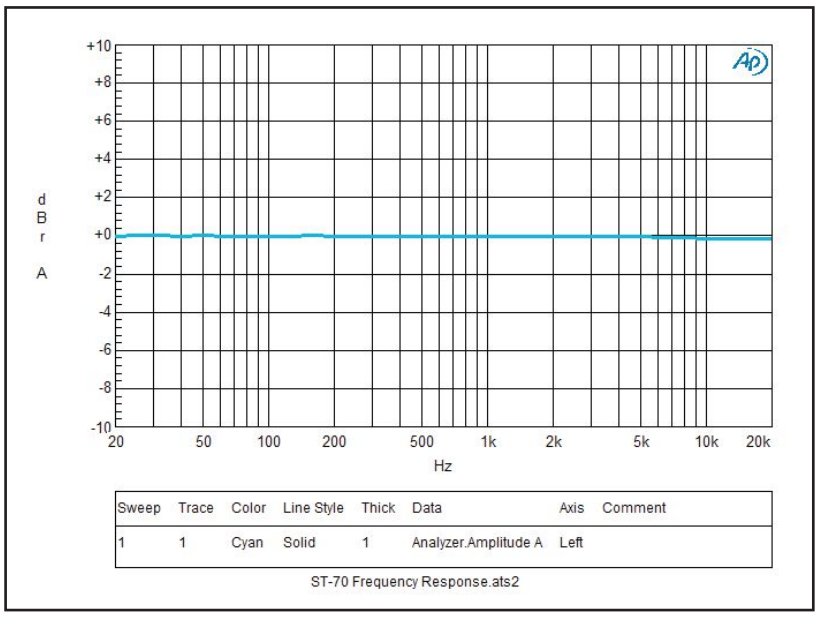


Figure 2: We obtained this frequency response with the new transformer.

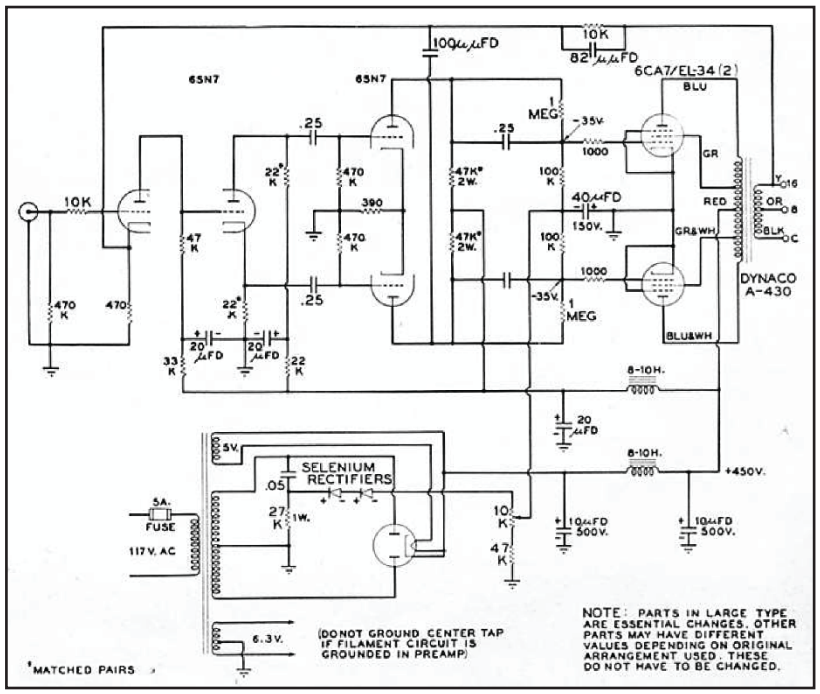


Figure 3: David Hafler drew this advanced Williamson circuit drawing.

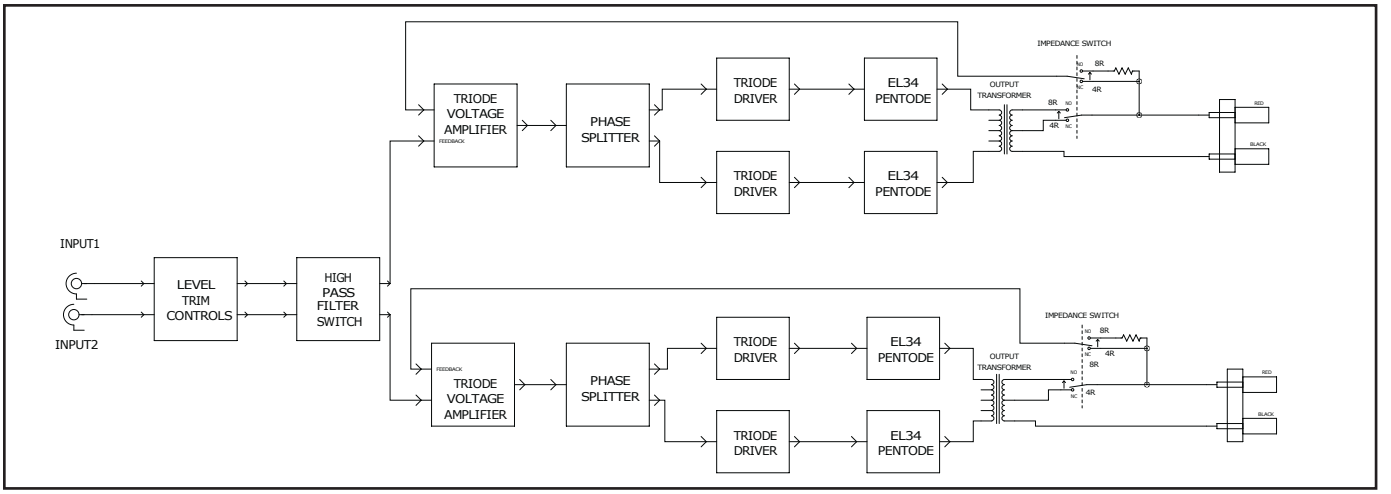


Figure 4: The block diagram details how the Dynaco ST-70 Series 3 functions.

Input Filter

Due to the extended low-frequency response that the improved power supplies and the inherently lower damping factor tube amplifiers provide, operation with acoustic suspension speakers may result in an exaggerated low-frequency response. Hence, a passive high-pass, three-position filter switch is provided with a bypass position and two different frequencies of gentle 6 dB per octave roll-off. As well, there is a passive radio frequency filter well above 30 kHz to prevent picking up AM radio signals.

For best performance, the driving device should have an output impedance of 5 kΩ or lower. Operation with older equipment with very high output impedance, such as the PAS-3X, may experience some high-frequency roll-off, especially with longer connecting cables.

Block Diagram

Each channel features a dual triode device operating as a voltage amplifier and "cathodyne" phase splitter. Feedback is applied to the Triode Voltage Amplifier. Following is a second dual triode acting as a driver stage and providing additional open-loop voltage gain (see **Figure 4**). Each channel drives a pair of EL34/6CA7 power pentodes operated in the classic Hafler Ultra-Linear mode. The power pentodes drive a custom made output transformer with both the extended high frequency response of the original transformers and extended low-frequency response. The impedance taps on the transformer are selected with a rear panel switch.

The original Stereo 70's PCB material was exposed economical phenolic, which deteriorated with age, creating unpredictable electrical changes that resulted in sonic differences. The ST-70 Series 3 unit uses military-grade, double-sided epoxy fiberglass PC material and shields the recessed PCB with a solid metal chassis above it. The board is populated by high-grade modern parts. Sensitivity to internal wiring



Photo 8: Adjusting the bias on the new Dynaco ST-70 Series 3.

Never Compromise on Transformers

The only parts David Hafler never skimped on were his transformers. Other than that, he used the cheapest parts he could find. Ed Dell always said that Hafler really didn't believe that quality parts mattered—carbon film resistors were just as good as metal film and ceramic disc caps were just as good as polypropylenes. I'm sure cost played a role, as well. Either way, this is a good part of the reason why Dynaco and Hafler products have been vehicles for so many modifications over the years. They were good, basic designs that usually benefited from parts upgrades. Of course, many of those modifications also involved design changes, as well.

All that said, based on what I have learned about the new ST-70 Series 3, it should be a significant improvement over the original (which was one of the highest-value audio products ever made).

—Gary Galo



Photo 9: The ST-70 Series 3 is the first Dynaco product to be relaunched since the brand was acquired by parent company Radial Engineering.

placement is eliminated since wiring to the output tubes is accomplished via the PCB. In fact, nearly two-thirds of the trouble-prone original point-to-point wiring has been eliminated! The bias adjustment is now done with indicator LEDs (see **Photo 8**). After a warm up period and with no signal playing, all that is required is to set a pair of LEDs to equal brightness. To accommodate unequal aging between initially matched pairs of output tubes, separate adjustments are provided for both tubes of each pair.

Conclusion

Recreating and improving a classic is a lot more work than any of us expected (see **Photo 9**). Our goal was to do justice to the original product, add the necessary improvements, project the vision of the original designer into a modern version that will be just as successful as its predecessor.

Is the New Dynaco ST-70 Series 3 Really Class A?

audioXpress asked this question to Dynaco's engineering team that was responsible for the new Dynaco ST-70 Series 3. In response, they agreed the Class A/Class AB issue is controversial. Dan Fraser, the engineer who led the project said, "The original was Class A up to about 10 W and Class AB above that. The Series 3 is the same."

Still, Dynaco admits that only half of the audiophile community will echo Fraser's comments. The other half will say that it's only fair to define an amplifier's class at full output (i.e., right before clipping), in which case the ST-70 Series 3 is running Class AB. Even Dynaco's own team doesn't agree on the classification. The Dynaco Series 2 manual called it Class A, but there's no mention of Class A or AB in the Series 1 documents.



An original Dynaco Stereo 70 Series II is shown. (Image courtesy of the Steve Hoffman Music Forums)

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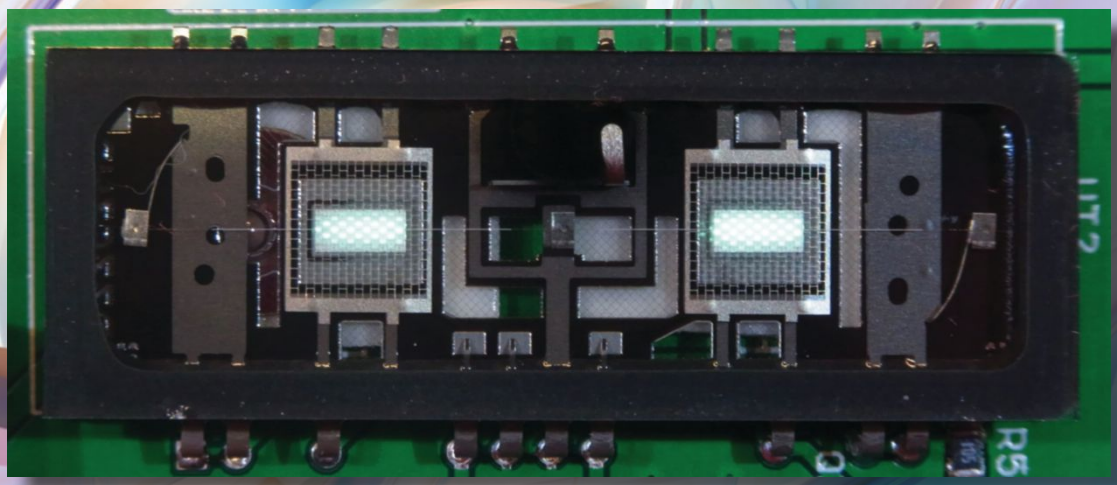
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The Korg Nutube



Learn more about the Korg Nutube 6P1—a new miniature, low-power dual directly heated triode device developed by Korg (of musical instrument fame) and Noritake Itron, a Japanese manufacturer of vacuum fluorescent displays. Although it is a true directly heated triode, it is manufactured differently than traditional vacuum tubes, which gives it some unique attributes.

By
Pete Millett
(United States)

Electrically, the Nutube 6P1 is somewhat comparable to the battery-powered "hearing aid" tubes. It operates on very low voltages—the filament is rated at 700 mV, and plate supply voltages can be as low as 5 V.

Nutube 6P1 Typical Characteristics (each triode)		
Filament voltage	Vf	0.7V
Filament current	If	17 mA
Plate voltage	Va	12 V
Grid voltage	Vg	+2 V
Plate current	Ia	32 µA
Amplification factor	µ (mu)	14.5
Transconductance	gm	54 µS (54 µA/V)
Plate resistance	Rp	330 kΩ
Grid current	Ig	6 µA
Maximum Ratings:		
Filament voltage	Vf	0.8 V
Plate voltage	Va	80 V
Plate dissipation	Pa	1.7 mW

Table 1: These are the Nutube 6P1's electrical characteristics.

Electrical Characteristics

The Nutube is a low power device—the filament consumes only 12 mW of power per triode, and the plate dissipation is rated at only 1.7 mW. The consequence of such low-power operation is that, compared to most traditional vacuum tubes, it also has low transconductance (gm). Refer to **Table 1** for a summary of the 6P1's electrical specifications.

Even though the numbers are a bit different than a common tube (e.g. the 12AX7), a first look at the plate characteristic curves will seem familiar (see **Figure 1**). The curves are typical for a triode and are quite linear. However, when selecting an operating point and load line, you must observe the maximum plate dissipation and maximum permitted plate voltage, which significantly constrains operation. Also notable is the fact that to get much of an output signal with a low plate voltage, you must operate the grid in the positive region—Class A2.

Since much of the Nutube's advantage is its ability to operate on low-voltage power supplies, most applications will utilize a positive grid bias. This means that some grid current will flow, and a somewhat low impedance source (e.g., a source

follower) must be used to drive the grid.

Figure 2 shows the grid current. You can see that the grid current, although only tens of microamps, is significant when compared with the low plate current.

Mechanical and Physical Attributes

The Nutube is manufactured by Noritake Itron in Japan, using a process similar to the one used to make vacuum fluorescent displays. It is, in fact, a one-pixel vacuum fluorescent display (VFD) device. The plate is coated with a phosphor just like a VFD, and it glows a blueish-white color when plate current is flowing.

A close look at a 6P1 triode shows a fine coated wire running across the device (see **Photo 1**). This is the filament. Even though it is heated enough to emit electrons, it does not get hot enough to show a visible glow. Below the filament is a fine, open mesh—this is the grid. Below that is a metal plate with the phosphor (glowing) applied—this is the plate. Two triodes are packaged together into one glass package, which is again borrowed from VFD manufacturing. It looks a bit like an old 40-pin DIP IC package, but only has pins along one edge (see **Photo 2**).

The two filaments share a common center terminal. In most applications, this terminal is at ground potential. Because of this, certain circuits—such as cathode followers or differential amplifiers—can be more difficult to implement, since they would require a floating filament supply.

One notable side effect of the construction is that the filament wire, under tension, is a lot like a piano string—when subjected to a shock or mechanical impact, it vibrates at an audible frequency (5 kHz or so). This microphonic behavior seems identical to that of the sub-miniature battery tubes and could cause problems when the Nutube is used in low signal applications. In some cases, it might be desirable to mechanically isolate or shock mount the tube.

The package is a glass sandwich, 45 mm × 16 mm, and is designed to mount to a PCB with pins spaced on 2 mm centers (see **Figure 3**). A small metal disc on the back of the package is used to seal the port where the air is exhausted.

Reference and Example Circuits

The basic application circuit provided by Korg uses simple JFET source followers to buffer the signal in and out of the Nutube (see **Figure 4**). An input buffer is used to support the relatively low impedance looking into the grid and bias source. The output buffer is needed to drive anything but high impedance loads due to the low transconductance and high plate resistance of the tube.

This circuit is intended to operate from a single power supply between about 5 V to 30 V, plus it

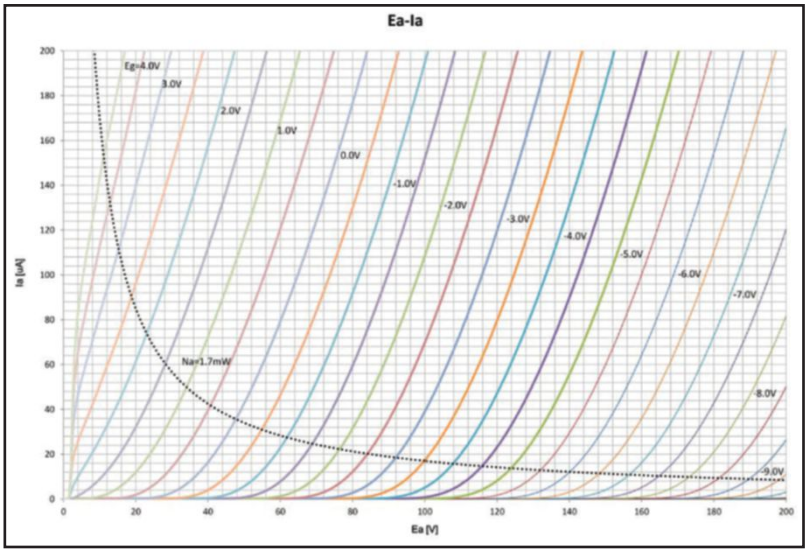


Figure 1: The curves are typical for a triode and quite linear.

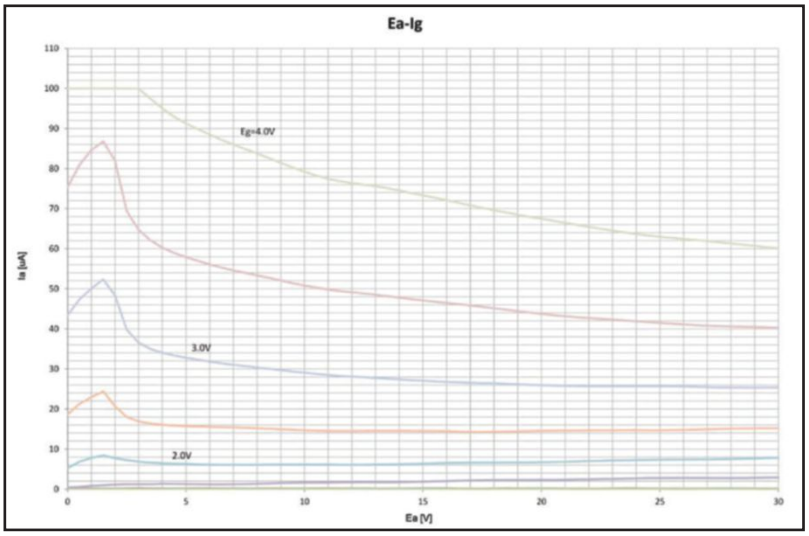


Figure 2: The grid current is significant when compared with the low plate current.

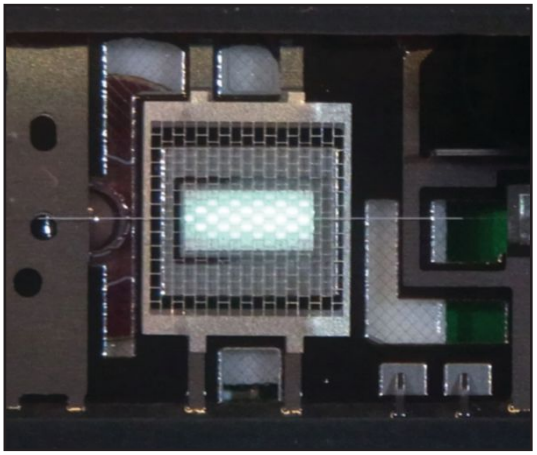


Photo 1: The 6P1 triode has a fine coated wire running across the device.

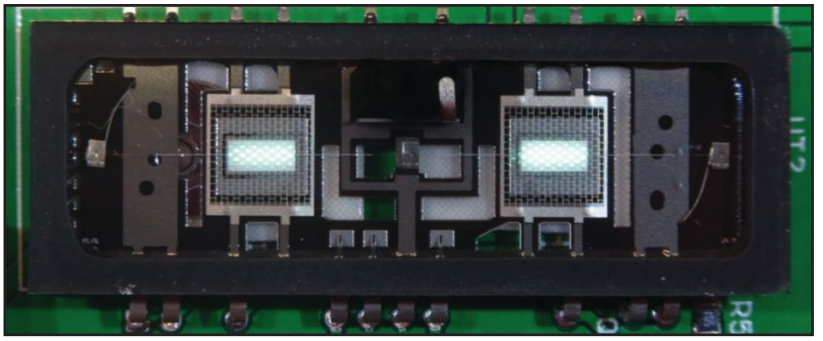


Photo 2: The Nutube resembles an old 40-pin DIP IC package, but only has pins along one edge.

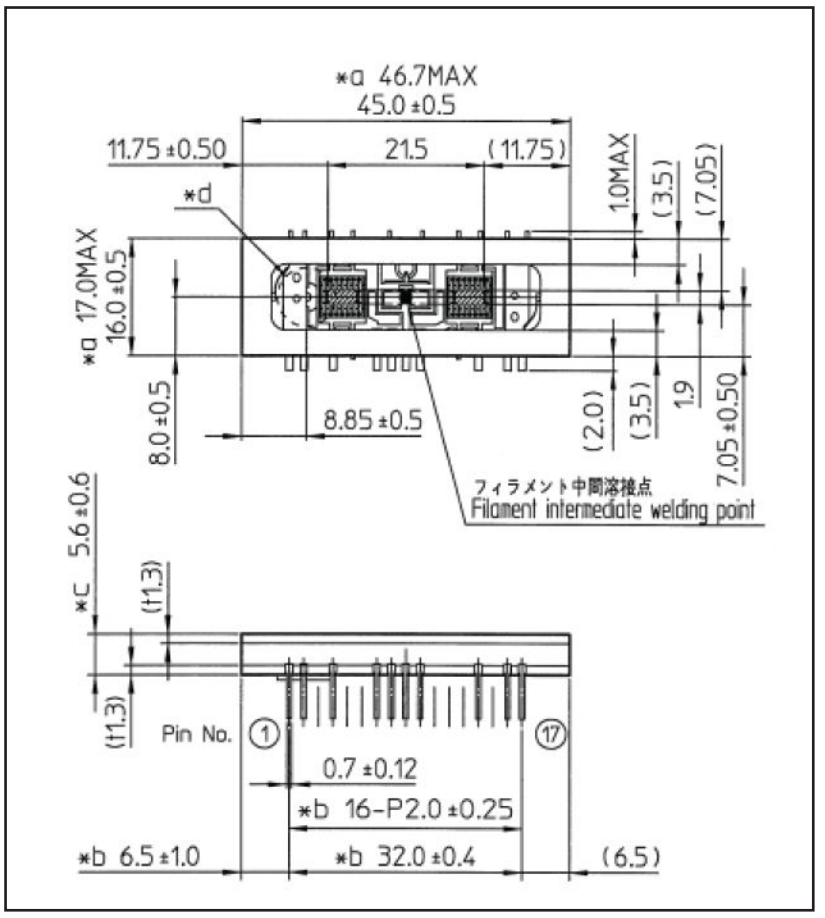


Figure 3: The package measures 45 mm × 16 mm and is designed to mount to a PCB, with pins spaced on 2 mm centers.

About the Author

Pete Millett is an electrical engineer, currently employed as a Technical Marketing Engineer at a semiconductor company, where he defines and supports mixed-signal power ICs. Previously, he worked as a board-level hardware design engineer in the computer and consumer electronics industries for more than 30 years. The design and construction of vacuum tube audio equipment has been a hobby of his for many years.

needs a regulated 3.3 V supply, typically from an low dropout (LDO) regulator. Bias is adjustable from 0 to +3.3 V. To support the grid current, a relatively low value resistor (33 kΩ) is used to provide DC bias to the grid. This mandates that a large (10 µF) coupling capacitor be used to couple the audio signal. There is no doubt there could be some bias shift with signal level in this arrangement but with the values shown, it doesn't appear to be significant. Filament power is obtained from the 3.3 V supply with a dropping resistor.

Figure 5 shows another example. In this circuit, ±18 V supplies are used, as well as a higher voltage B+ voltage for the plate (in this case, about 75 V). Both triode sections are connected in parallel, to increase gm and lower the noise level. A DC-coupled JFET follower is used to directly drive the grid at zero bias, and an op-amp is used to buffer the output.

I also used this as a gain stage in the Apex Sangaku headphone amplifier. Note that an op-amp with low input current offset and noise is needed, as a large resistor (1 MΩ in this case) is necessary at the op-amp input due to the 6P1's high Rp.

Audio Performance

As with any vacuum tube, the audio performance—gain and distortion in particular—are dependent on the circuit implementation. Bias point and plate loading make enormous differences in the performance of an amplifier stage.

I ran some tests using the Korg application circuit, at the recommended bias and plate load, to determine the audio performance. All the tests were run with a grid bias voltage of +2.5 V, and a plate load resistance of 330 kΩ. With a single +12 V supply, the circuit provided a gain of about 2.6 times. With 0.5 V RMS in, and 1.35 V RMS out, total harmonic distortion (THD) was around 1%. The THD increases with the level as one would expect.

In **Figure 6**, the horizontal axis is the input voltage. The green line shows the THD, read on the left scale. The yellow line shows the output voltage, read on the right scale. You can see the compression of the waveform at higher levels, causing the yellow trace to bend downward. The THD reaches 5% at about 3.1 V out. At this operating point, with 0.5 V RMS input, a Fast Fourier Transform (FFT) graph shows that the second and third harmonic distortion products were nearly equal in level at -40 dB, with dropping levels for higher order products (**see Figure 7**).

Using the same operating point and load, things greatly improve at a higher supply voltage. **Figure 8** shows the same circuit running from 24 V. You can see that the gain has increased to about 3.5 times. THD is lower, with 0.7% at 0.5 V in and 1.8 V out.

A 5% THD isn't reached until the output reaches 6.5 V RMS. The FFT at 24 V shows that the second harmonic product has dropped, and now the third is the highest, at about -43 dB (see **Figure 9**).

By changing the bias and/or plate load, you can tune the harmonics however you want. A small change in bias will change the relative levels of the second- and third-order products. Negative feedback can also be employed to trade lower gain for lower distortion.

Applications

The Nutube 6P1 is intended to be used in various audio circuits, mostly in musical instrument applications. Its low power requirements enable its

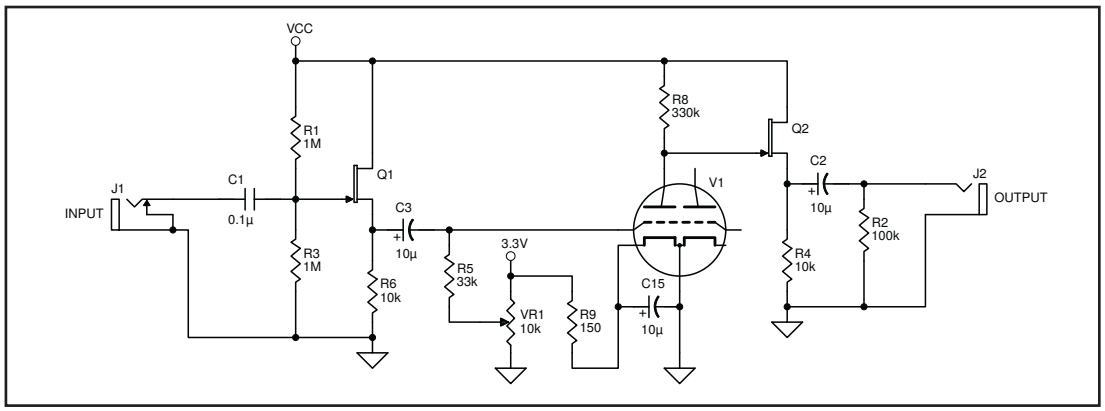


Figure 4: The Korg application circuit uses simple JFET source followers to buffer the signal in and out of the Nutube.

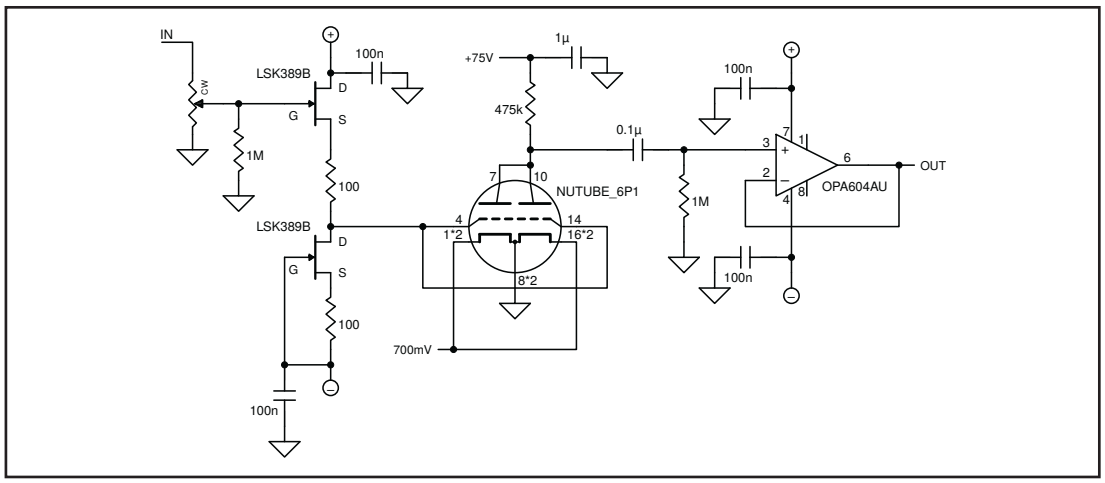


Figure 5: This is an example of a split-supply application.

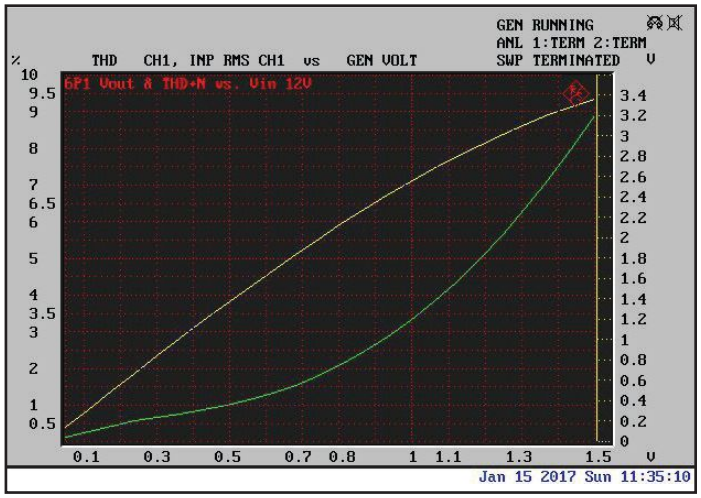


Figure 6: With 12 V gain and distortion, the THD increases with the level.

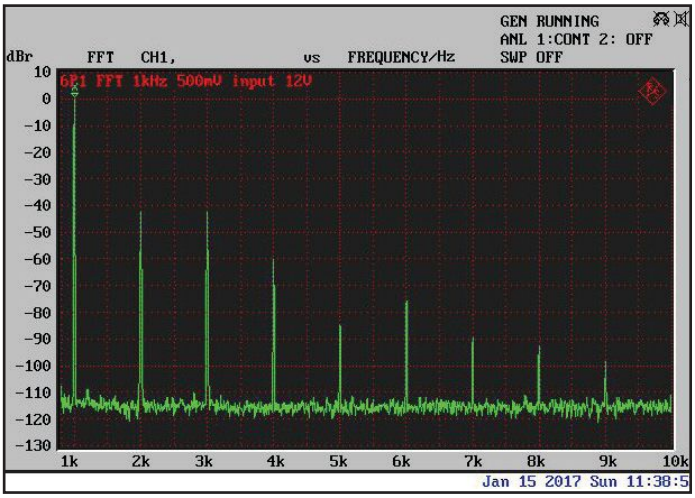


Figure 7: At this operating point, a Fast Fourier Transform (FFT) shows that the second and third harmonic distortion products were nearly equal in level at -40 dB, with dropping levels for higher order products.

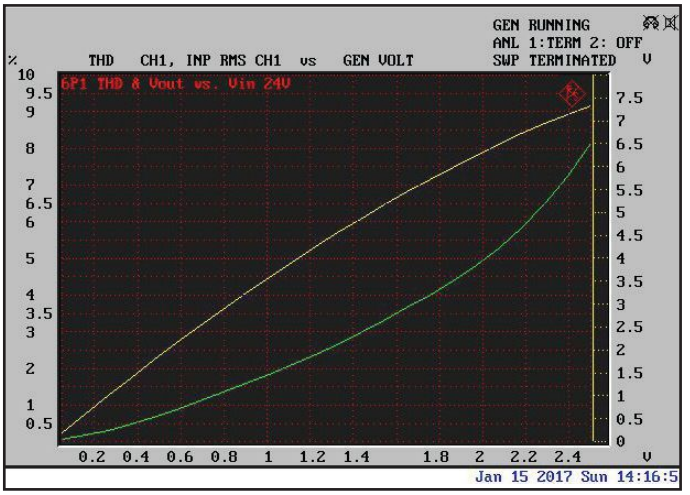


Figure 8: Using the same operating point and load, things improve considerably at a higher supply voltage. Here is the same circuit running from 24 V.

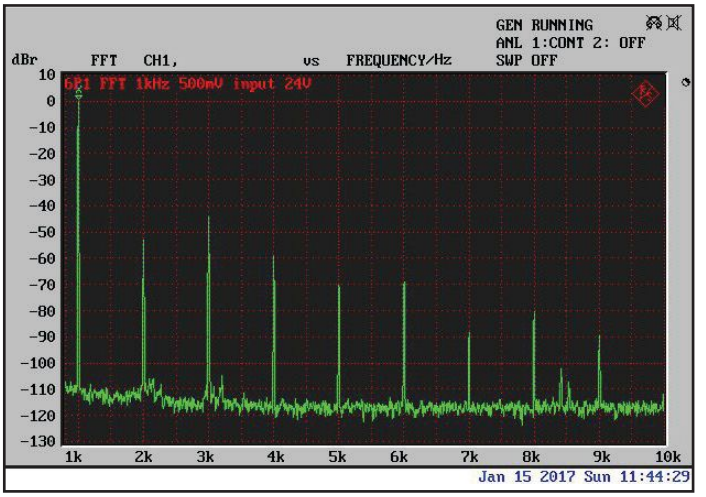


Figure 9: The FFT at 24 V shows that the second harmonic product has dropped, and now the third is the highest, at about -43 dB.

use in battery-powered systems, ideal for a product such as a guitar effects pedal. However, due to its low gm and correspondingly high noise, it is difficult to use in low-level applications (e.g., the input stages of microphone or phono preamps). In some cases, the Nutube can also find use in hi-fi audio equipment.

As you might conclude, based on the audio performance measurements, I view the Nutube not so much as an active amplifying device as a signal processing element. Yes, it does provide voltage gain—but if accurate voltage gain is your goal, you are much better served by using other devices, both tube and solid state.

In a guitar amplifier, it can be used to provide distortion similar to that of a conventional tube, with much lower voltage and power requirements. In hi-fi, it can be used to emulate the harmonic characteristics of single-ended triode power amplifiers. Not everybody's "cup of tea" to be sure, but it opens up some interesting possibilities.

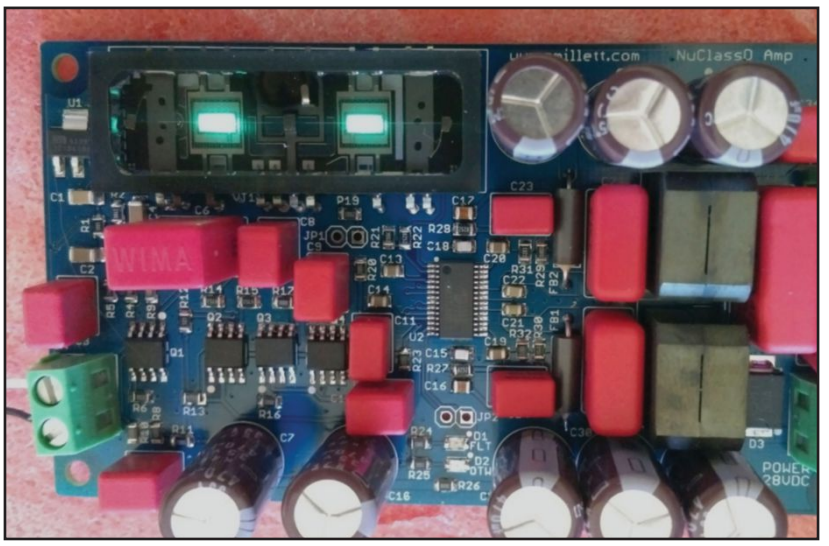


Photo 3: As an experiment, I designed and built a hybrid amplifier using the Nutube 6P1 in conjunction with a self-oscillating Class-D amplifier IC, the MP7770 from Monolithic Power Systems.

As an experiment, I designed and built a hybrid amplifier using the Nutube 6P1 in conjunction with a self-oscillating Class-D amplifier IC, the MP7770 from Monolithic Power Systems (see **Photo 3**). This combination yields about 50 W output power, and harmonically, it is much like a large single-ended triode amplifier. It runs on a single 28 V power supply and each channel occupies only a few square inches of PCB space, certainly smaller and much less expensive than a big tube amplifier.

In this amplifier, I set the MP7770 Class-D circuit up for slightly lower gain than it would normally have and used the Nutube 6P1 as an input stage, feeding a JFET phase splitter. Since it is powered from a single 28 V supply, I used a circuit similar to the Korg reference circuit, with a couple of differences. I connected both triodes in parallel, and also added some plate-to-grid feedback. With a positive grid bias, it's easy to add feedback by simply adding a resistor (see **Figure 10**).

This type of feedback stabilizes both the DC operating point and reduces distortion. With this arrangement, it is easy to tune the stage to have whatever gain and harmonic profile you want, as you can vary the plate load resistor, bias voltage, and negative feedback.

The MP7770 amplifier has quite low distortion by itself (I measured about 0.02% at 1 W), but as with most Class-D amplifiers, it does have low-level harmonics that extend to many times the fundamental. A theory of harmonic masking says that

higher amplitude, lower-order harmonics mask the audibility of higher-order harmonics. So the hope here was that the sound of the resulting hybrid amplifier will be dominated by the Nutube input stage, with its relatively large amounts of low-order (mostly second harmonic) distortion. It does, in fact, measure similarly to a large single-ended tube amplifier. People who have heard it agree that it does indeed sound like a single-ended triode amplifier but has improved damping due to the low output impedance of the Class-D circuit.

Conclusions

Though it won't make conventional small tubes (e.g., the 12AX7) obsolete, the Korg Nutube can be a useful device for audio DIYers. I expect it will find many uses in musical instrument amps, guitar effects, and hi-fi equipment. Its low power and low operating voltage makes it a natural for experimentation by those who are not familiar or comfortable working with the high voltages typically used with tube circuits.

For more information, visit www.korgNutube.com. The Nutube 6P1 and supporting evaluation hardware is available from www.Nutube.us.

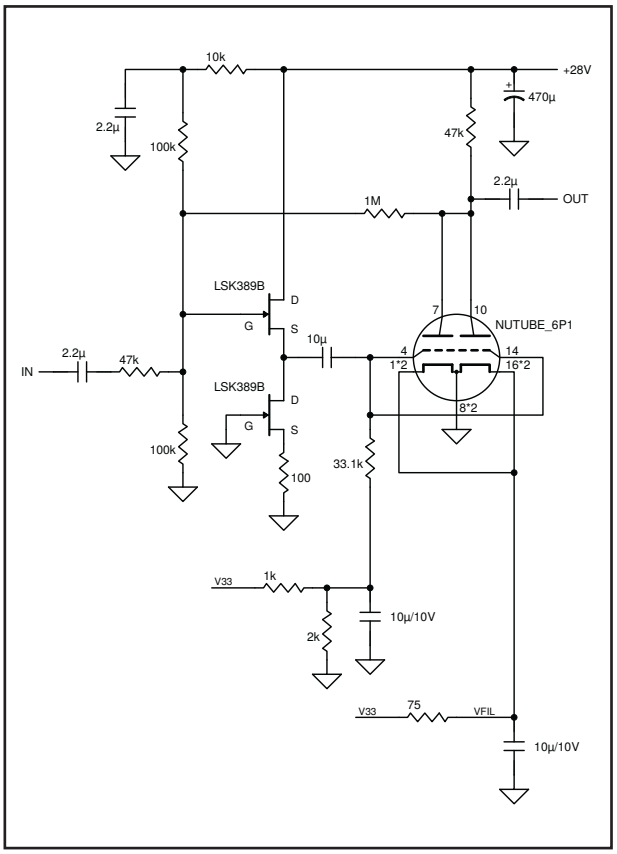


Figure 10: I set the MP7770 Class-D circuit up for slightly lower gain than it would normally have and used the Nutube 6P1 as an input stage, feeding a JFET phase splitter.

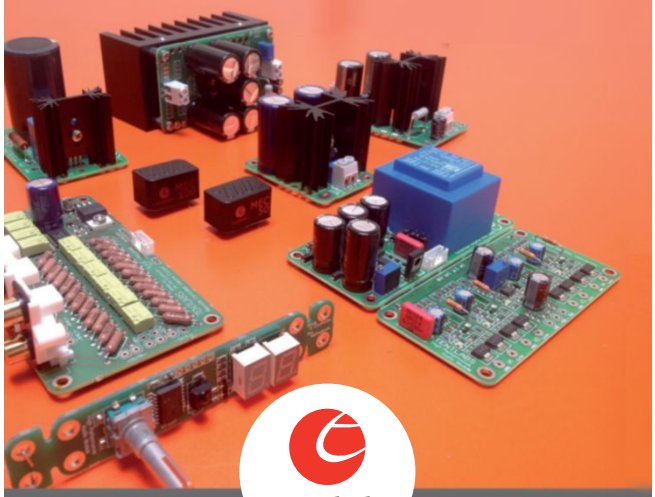
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Kismet 5: The 20th Anniversary Edition

The Evolution of Its Design

My first Kismet amplifier project appeared in *Glass Audio* (*audioXpress*' predecessor) in 1996. I could never have imagined 20 years later I would be writing about the fifth generation of the Kismet amplifier! When I made the first amplifier, it was to demonstrate (mostly to myself) how to get good results with simple and inexpensive choices.



Photo 1: I built the Kismet 5 about 20 years after sharing the original Kismet design. The iteration is an improvement but stays true to the original's low-cost overall approach.

By
Fabio Camorani
(Italy)



Photo 2: This is the original Kismet, which I built back in 1996.

I have always been fascinated by high-quality, low-cost designs. I think that cost-is-no-object products are easier, but they do not always sound better. (It seems impossible, but several high-cost products out there sound really bad.) The challenge is to get very good results from low-cost designs. I spent and still am spending a lot of time on this. What works? Which aspects are more important? And what sound is a good sound?

The Kismets

Over the past 20 years, I have found the different variations of the Kismet topology reflect the changes happening in my life. After all, in 20 years there have been many changes. I met the girl who later became my wife, we had two daughters, and I created two small audio brands and one small label for re-issuing

very good masters. I also obtained a Master Black Belt in Lean Thinking.

During those 20 years, I came across several mentions of my Kismet design on the Internet. And I also found several schematics that appeared to be based on my original design (even if sometimes those authors did not mention it). This makes me proud of the Kismet and provided me with the challenge to improve the design, while maintaining the overall approach and keeping costs low—the result is Kismet 5 (see **Photo 1**).

The original 1996 design contained a 6922 common cathode plus a single-ended 2A3 with a Tamura F475 and a 5U4G rectifier with a CLC filter (see **Photo 2**). Then, I developed the second-generation Kismet, just using the series regulated push-pull (SRPP) input stage. It only cost one more double triode (plus a socket) than my original design. However, I never published this variation.

In 1998, I shared my design for the Kismet 3 in an article titled, "Kismet 3: The Last Generation," in *Positive Feedback* (June/July 2002). As a result, I received a Kismet iteration from another DIYer, which contained an SRPP of 5687 (or 5965—then 6414—in the 3i version), without other significant changes (see **Photo 3**).

At the time, I considered Kismet 3 to be my last Kismet evolution. Time passed and I never made any further improvements to the design, but I did store away some thoughts taken from my studies of other amplifiers and gained some fine-tuning knowledge through discussions with some DIYers.

After several years, thanks to the DIYers who continued to contact me for suggestions on improving the Kismet 3, I decided to write down some of those improvements and Kismet 4 was born. Kismet 4 still had the 2A3 with F475, with the 6414 in SRPP but it contained a different power supply, with solid-state diodes and a few other modifications. However, that version was also never published because I did not have the time.

Then, four years ago I got the chance to work on a 4P1L pentode (during my research of triode connected tetrodes/pentodes) and I recalled that the 20th anniversary of Kismet would be coming up and thought…Why not a Kismet 5?

A "New" Power Tube

Without going into detail about the loctal 4P1L (there is tons of information and schematics about it on the Internet), I decided to try it in the Kismet topology, leaving the 2A3. The 4PIL, made by Svetlana is a copy of the RL2, which was used during World War II by the German Wehrmacht (see **Photo 4**). It's a directly heated pentode and triode connected so it is extremely linear! Some claim it's the most linear triode ever measured.



Photo 3: A DIYer sent me a photo of his Kismet 3i.

This tube triode strapped (G2 and G3 connected to the plate) shows a mu of about 8 and a plate resistance of about 1,200 Ω, so the transconductance is 5.6 mA/V, with a plate dissipation of 7.5 W (plus a screen that is 1.5 W). Thus, putting two in parallel, we are not far away from the 2A3, with similar dissipation and plate voltage, as well as plate resistance (600 Ω vs. 800 Ω). I like high transconductance tubes!

The tube's inexpensive cost, its linearity, and its similarities to the 2A3 convinced me to use it in the Kismet topology. Over the past 20 years, the original RCA 2A3 has become extremely expensive (for such a common tube) and the Chinese 2A3s are getting expensive and "strange." (They call them 2A3s but they are definitively different tubes with "identical" specifications.) A year ago, I found a small batch of



Photo 4: The Russian Svetlana 4P1L tubes are similar to the German Wehrmacht RL2s used during World War II.

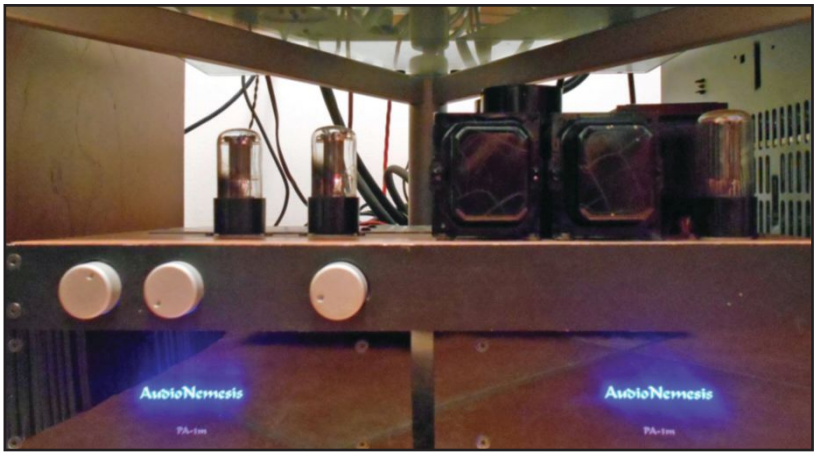


Photo 5: I fell in love with the 6SN7, after listening to an old SRPP preamp many years ago.

Chinese 2A3 tubes from the 1990s (a copy of the RCA tube) and I purchased them all. But, they are getting more difficult to find.

The high gain of the 4P1L in triode mode enables me to use my favorite low gain signal triode: the 6SN7. This is impractical with the 2A3, if not impossible as I discovered during the Triodino 3 project I helped to design with friends in Italy (see Resources). I fell in love with the 6SN7, after listening to an old SRPP preamp many years ago (see **Photo 5**).

Sometimes They Return

Since 1996, I have often written about the importance of having a "balanced sound." This is always one of my primary design goals. And after 1996, I often used the SRPP. I know that some DIYers say this circuit has a "strange" sound, claiming different defects. Despite it's imperfections, I always liked it for its energy and other aspects. I made very good preamplifiers with it. But prior to that, I used the mu-follower because I liked its fine grain. I used it for the input stage of the Triodino 3, and did not use it again for several years. About five years ago, I rediscovered the mu-follower. It seems to give me a more balanced sound in my circuits. So, what I have concluded is that it's again a matter of balance: I changed the circuits and the SRPP could be exceeded by the mu-follower for a better balance.



Photo 6: For the Kismet 5, I chose the Sicte 6SN7 and the Fivre 6SN7, two top sounding tubes.

Anyway for the Kismet 5, I chose a mu-follower of 6SN7 for the circuit. Which 6SN7? The Sicte 6SN7 (see **Photo 6**). I prefer it over the Fivre. I have several US-made 6SN7 tubes but… it's up to you and your taste in the sound balancing. A good and inexpensive alternative is the 7N7 (loctal socket, like the 4P1L). You should give it a chance.

Anode Power Supply

During the last 10 years, I conducted extensive listening tests on power supplies, both stabilized and not stabilized. I tried several solutions, compared many ideas and parts (e.g., bipolar, MOSFETs, capacitors, etc.) and listened to their impacts in many different applications, sometimes with surprising results (bad and good). And I have come to some conclusions—both on the impact on the final sound and on the best topology. In a Kismet-like design, a stabilized power supply for anode power is out of the question, because it's too complicated and expensive. So once again, I went with a standard CLC filtering. But nothing is preventing you from trying to stabilize it for your own project.

After the first three generations of this amplifier and my early designs, I left the standard rectifiers in their storing box. All those high impedance rectifiers do is slow down the speed of music. I found them to be very good sounding because they are smooth, but I wanted dynamic energy! Live music is pure energy. So I concluded that I needed very low impedance rectification.

In the fourth generation of the Kismet, I used solid-state high-speed diodes. The best solid-state diodes are too expensive for this amplifier, so I compromised. For Kismet 5, I chose a couple of damper diodes, which have very low impedance and high current compared with standard rectifiers. So, back to tube rectifiers in Generation 5!

TV dampers have been developed to dampen oscillations in the deflection system of TV systems during the fly back period of the electron beam. This application is very critical, handling high peak currents and high reverse voltages without arc-over. As far as I know, J. C. Morrison was the first to use these

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very special diodes in audio amplification, back in the 1990s on *Sound Practices*. I see they are common now among DIYers, not in the audio business, despite their exceptional characteristics and super low price.

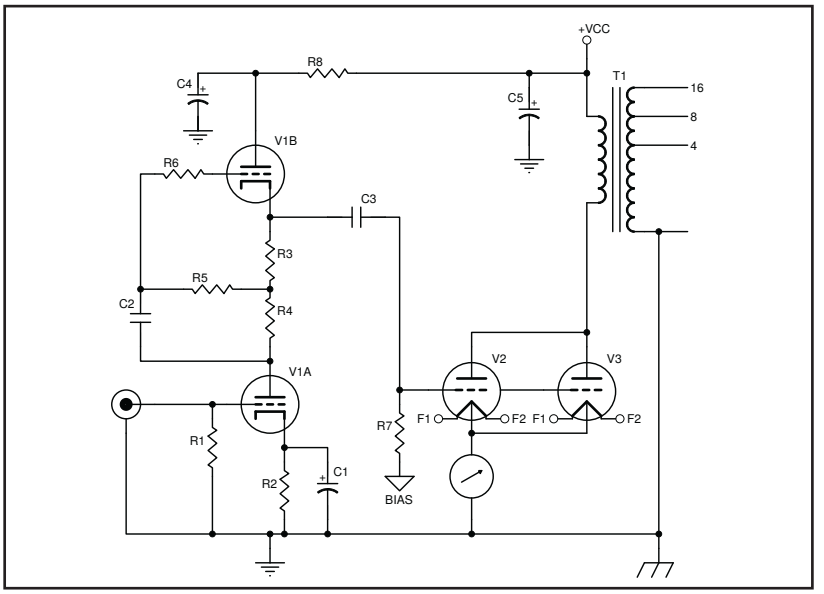


Figure 1: The signal schematic shows the design for the Kismet 5's right channel. The left channel is, of course, identical.

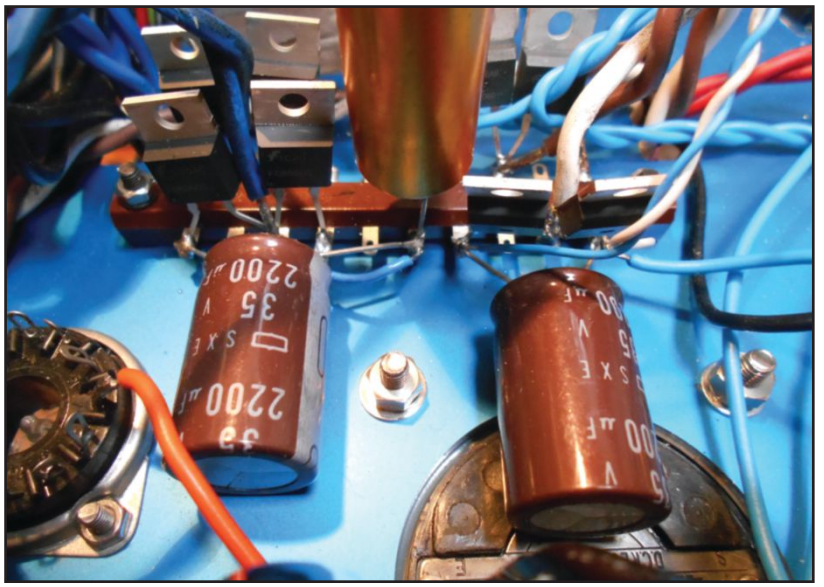


Photo 7: The inner view of the Kismet 5 details the diodes and filtering before CCS.

About the Author

Fabio Camorani became an Audio Engineering Society (AES) member in 2000. He received an Electronic Engineering degree at the University of Bologna in Bologna, Italy. Fabio published his first audio article in 1994. He started his own audio brand AudioNautes in 2000. In 2004, he added the less expensive AudioNemesis brand. In 2010, Fabio created the AudioNautes Recordings to reissue top-quality discs. Fabio is also a Lean Master (Lean Thinking is a business methodology similar to the Toyota Production System in a more general way). Fabio also does triathlons.

There are many different models out there, at various different heater voltages. Most of them are perfect for most audio applications, and they are very reliable. They also have good insulation between the cathode and the heater.

In the Kismet 5, I used the 25CG3 (a 25 V version of the 6CG3), which is able to handle 350 mA DC current (>1000 mA steady-state peak) and with up to 9,000 V peak inverse plate voltage, with a 9 W plate dissipation (see **Figure 1**). It is really a high efficiency diode! Damper diodes also have the damper characteristic! And this is extremely useful, as we see later.

But in this case, the filter is not a pure capacitor input one. I love choke input filters; however, to avoid a big loss of voltage (meaning bigger power transformer, so higher cost), I chose an uncommon option: the "hybrid" filter. The "critical value" for choke input is 2 H (Vdc/I) and a good practice is to double it (4 H). I used a standard 10 H for better filtering. The input capacitor value is small, too small for a pure capacitor input, too big (bigger than zero) for the inductive one. I also fine tuned the value to get the right final voltage on the tubes! It's also a good compromise from the sound point of view. The filter's choke is, in my prototype, a Tamura A395 because I already had used one in my laboratory. It's not the best choice because I used it at its maximum level of current—a 200 mA model is a better choice. Thanks to the hybrid input filter, you don't need an expensive choke-type to avoid vibrations. A good, serious model is okay (with low Rdc).

CCS for the Filament

A few years ago, I also started playing with Constant Current Source (CCS) circuits. Over the past 15 years, I have experimented with stabilized power supplies, and now I use shunt regulated type only. When I started experimenting with them, the change from the series resistor to a simple CCS was dramatic. But which CCS type is essential for a good and balanced sound? The CCS has probably the biggest impact on the sound! This is clear to everyone who has tried CCSs and listened to them. However, I often see CCS in audio circuits (mostly for anode loading), that are similar to one another when not identical. But, I don't see many comments on their sound. I suppose that is because the designers just tested/copied one and that was it. I tested several topologies and I identified what I consider more balanced and found the one I liked the most, and then I improved it over the years.

In this case, I used CCS for filaments. The first time I did it I was really impressed by the sound results. So I decided to try this solution for the Kismet. For my CCSs, I made a PCB, but it was too

big and expensive for this project. I had to find a simpler solution. What I propose is a CCS made with a properly connected LM1084. In CCS connection, it does well. Of course, feel free to try different solutions with better performance and test them with listening sessions! There are better solutions, but they would probably be more expensive and more complex. I put the filaments of the 4P1L of each channel in parallel, so the CCS is giving 650 mA for about 4.2 V across each filament. R30 and R31 are for fine tuning. The basic parallel configuration (R23-24 and R28-29) was a bit high for all my 4P1L batches, resulting in a slightly lower current (about 640 mA without considering the tolerance chain). I then used a parallel resistor (2 W) to tune the right value "on board."

An important consequence of the CCS is the intrinsic protection of the filament. The filaments (of the 4P1L) when cold have a very low resistance. Using no stabilization (AC) or a normal DC voltage, we have a peak of current at switch on, exceeding the normal value. With a CCS, this will never happen! We will always have the right current value and the voltage across the filament will slowly increase, with the temperature, preserving the filament itself.

For this design, the bias for the 4P1L is fixed; however, you could try the filament bias thanks to the CCS. You just need more anode voltage to allow it. If you try it, please let us know!

I also used a CCS for the 6SN7 heaters, both channels in series (600 mA, then 12.6 V). In this case a LT1086 is enough. Feel free to compare with voltage regulators. Per my usual design, the heater power supply of the 6SN7 is not referred to the ground, but to an intermediate value. This is essential for reliability because for the SRPP and the mu-follower, I used two triodes inside the same bulb. They use the same heater with different cathode voltages, so we need to use an intermediate value.

Other diodes are fast-soft recovery type (see **Photo 7**). They are not r cheap but their cost has dropped the last few years and I could not resist.

The Output Transformer

I intended to use my usual 5 kΩ load, and my favorite Tamura F-475. But to keep the cost of this project low, I decided to look for something less expensive but that would provide similar results. However, I could not find anything special, confirming my original choice. The one change is that over the past 20 years, I have met many good designers.

In my products, I always use Tamura or custom-made transformers, from MC step-ups to power outputs. So I asked one of those designers to do something for this amplifier. I also requested low losses and a few other specs. When I got them, I could not have hoped for anything better! After some trials I saw that 5 kΩ to 6 kΩ is good for one 4P1L in triode connection, not for a parallel operation. I discovered that 3 to 3.5 kΩ is good for parallel operation.

The designer did a wonderful job. I was convinced the results would be good because his designs follow ideas similar to mine. For example, I don't like transformers with extremely extended bandwidth. In general, these performances are only sacrificing other aspects and the final result shows a lack of "balance." And, I don't like "universal" designs. They do a lot of things, but nothing is done well. These transformers have 4 Ω-8 Ω-16 Ω outputs.

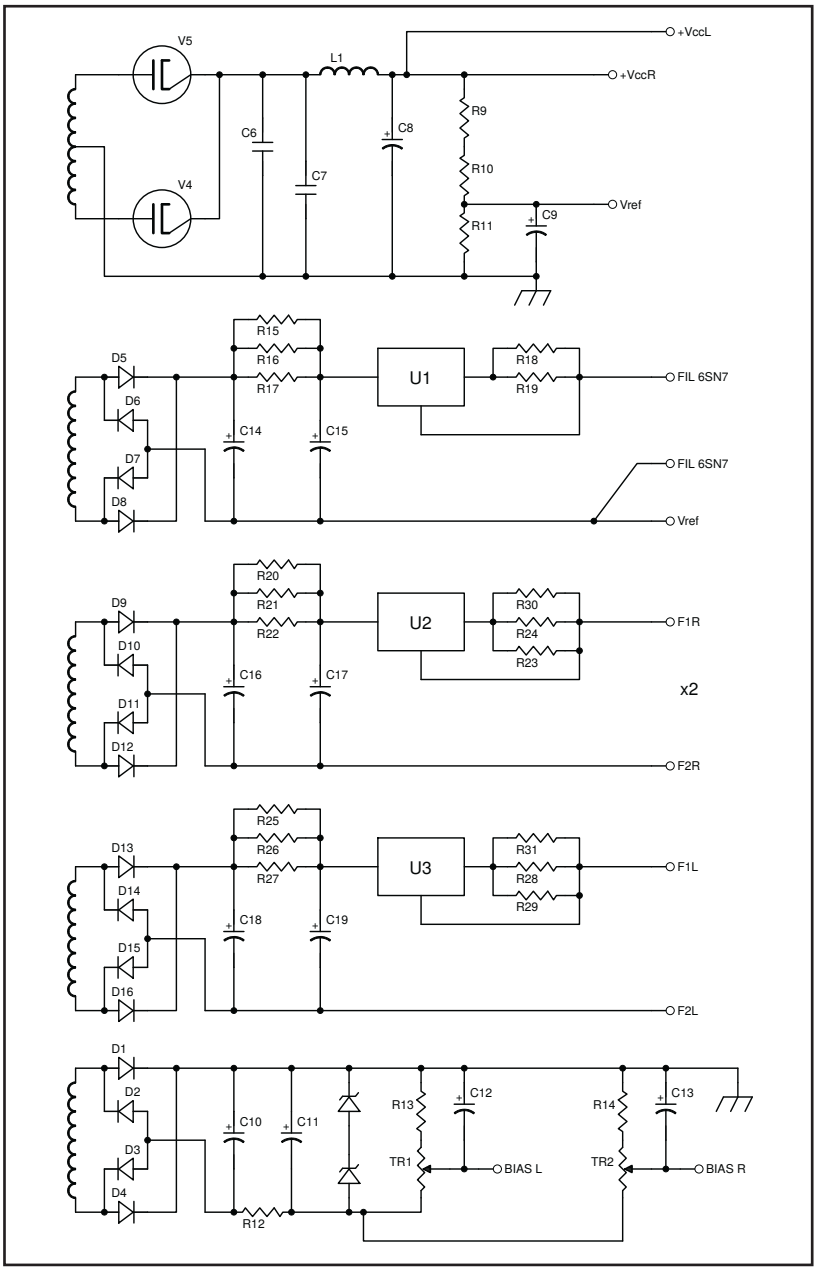


Figure 2: Here is the Kismet 5's power supply schematic.

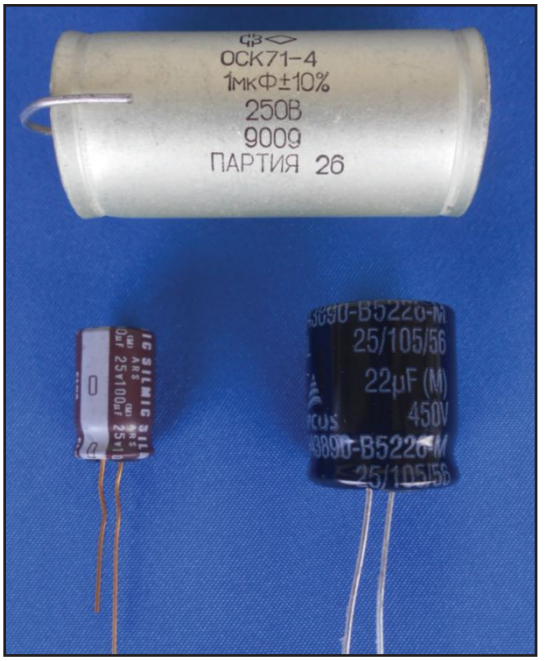


Photo 8: This is the Russian polystyrene capacitor, Elna Silmic II, and the Epcos 400 V.

Of course, you can use your spare output transformers or a pair from your preferred brand. You'll not get the Kismet 5, but you will get a similar amplifier with a different sound—maybe not very different, it depends on the transformer you use. A good choice would be the Tamura F-912 (2.5 kΩ) or F-913 (3.5 kΩ). They are not 3 kΩ (but they will fit well). Although I have not tested them, I will sooner or later.

Passive Parts and More

After 20 years, I still think the passive parts I used in the original Kismet design provided the best balance at a low cost.



Photo 9: Here is a front view of the completed Kismet 5.

Starting with the interstage capacitor, the market is full of polypropylene capacitors but only a few of them are really balanced (you can try the Clarity Caps ESA). For Kismet 5, I used a polystyrene Russian model. It sounds really good and is super inexpensive.

The first capacitor of the anode power supply (C6 and C7 shown in **Figure 2**) is a polypropylene type. In my case, I used a parallel of two good Solen capacitors. One is 4.7 μF and the second is the one to be fine tuned, also 4.7 μF. All the other capacitors are very good Epcos electrolythic type.

The bypass capacitors are Elna Silmic II, which are inexpensive and readily available. And the capacitors used for the filaments after the bridges are again very good Epcos (see **Photo 8**).

Regarding the resistors, I suggest good and reliable carbon 5% models in the signal path. You could also choose 1% metal models. However, pay attention to the applied voltage. For the CCS, you must use reliable higher power resistors.

For the signal path I used Acrolink wire for internal wiring. I got it from a Japanese friend and it sounds really good. RCA and Binding Post connectors are inexpensive but good gold-plated types, which I have tested many times during the years. I made the signal path as short as possible, with hard wiring and a central ground bus.

I also used two small instruments to check the current of the 4P1Ls. One instrument per channel, with a blue light to complement the light blue color of the aluminum ground plane.

I have included a complete Parts List in the Supplementary Material section of the audioXpress website (www.audioxpress.com). However, to summarize the passive parts, it's a matter of balance. All the parts along with the schematics play an important role, but the final sound is the result of everything working together. I never understood upgrades of amplifiers done using super expensive capacitors that cost as much as the rest of the amp! It's nonsense! The final result depends on how well all the parts work together. So feel free to try your favorite parts, but remember that the final balance depends on all the parts working in unison.

Again, the bias of the output tubes is fixed, as with all the Kismet's iterations. I understand the advantage of the automatic bias and, in some cases, it's very good, but I prefer the fixed version whenever possible. For this amplifier, I also used a shunt regulation with Zener diodes followed by two separated trimmers for fine tuning. I always use multi-turn trimmers for more precise regulation. R13 and R14 are used to avoid 0 V bias. Please notice the last filter capacitor of biasing: They are Silmic IIs. But, you could also try some ERO 1822 10 μF, 100 V instead.

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Chassis

I don't have anything special to say about the chassis. I made it out of wood and 3 mm aluminum ground plane because it is effective and inexpensive. I designed the ground plane with CAD software, and the supplier made it with excellent quality. I think it works for this project. I like to see the tubes and their lights, that's why I used an open chassis. You can do it yourself if you prefer, following your taste. But, always consider safety (first!) and hum aspects.

More important is that I used rubber isolators for all the transformers and the sockets. In particular, I used normal rubber O-rings to isolate all the "irons" from the chassis to avoid at least some of the vibrations they can introduce. Moreover, I used silent blocks to isolate the sockets of the output tubes—rubber dampers with screws are useful and easy. The only problem is that they are delicate. If you want to insert a tube into the socket, be careful not to break it. It is better to insert the tubes holding the sockets with the other hand. I also used silicon O-ring tube dampers for the 6SN7s.

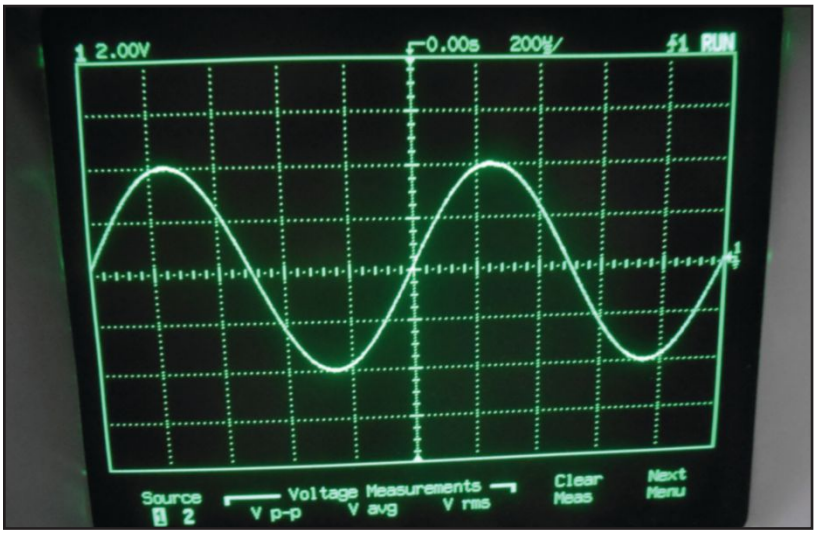


Figure 3: This is the Kismet 5's sine wave at 20 Hz.

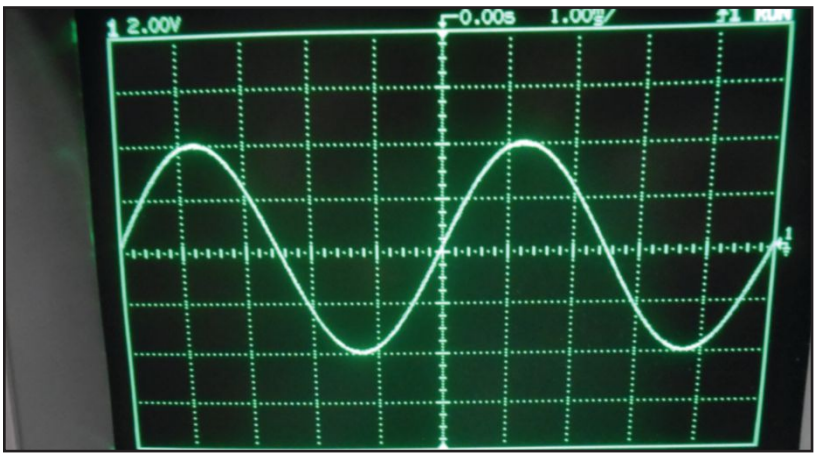


Figure 4: The Kismet 5's sine wave is shown at 1 kHz.

Assembly and Testing

There are many more parts to the Kismet 5 when compared to the original Kismet. This is mainly due to the CCSs. However, in my opinion, it's worth the effort. But, this amplifier is definitely more complex. It's not that basic and requires some experience to avoid potential disasters. It's a good idea to use a chassis bigger than the one I used, for additional space and added safety. Always safety first, folks! My version has a ground plane approximately the size of an A4 paper and is rather full. There is a little noise/hum, and it needs some adjustments.

I put all the input/output connections, as well as the on/off switch, on the ground plane. The instruments are on the front panel (see **Photo 9**).

When complete, the CCSs will need some fine tuning. I discovered that different batches of 4P1L tubes could have different filaments Rdc, so it also has a different current requirement when hot.

I tried to match the filament Rdc of the tubes, because I have more than 100 4P1L tubes in my personal warehouse. After connecting almost all the parts inside, I started from the CCSs, connecting them and then doing the fine tuning. Of course, you should get about 6.3 VDC across each 6SN7 heater and 4.2 VDC across each 4P1L filament. Then, I started with the other power supplies one at a time, leaving the anode one until last.

My suggestion is to always use a variac to verify and test every single step. The optimum working point for the 4P1L in a triode connection may change from batch to batch. I did draw the curve for my batch finding some variations over the standard datasheet (my batch was pushing more current). I started from the working points you can find in many projects

Project Files

To download additional photos and the complete Parts List, visit http://audioxpress.com/page/audioXpress-Supplementary-Material.html.

Resources

F. Camorani, "Kismet, a Simple 2A3 SE Amplifier," *Glass Audio*, June 1996.

——, "TRIODINO III: A Direct Heating Single-Ended Triode Amplifier," *Costruire Hi-Fi,* 1996.

——, "Kismet 3: The Last Generation," in *Positive Feedback*, June/July 2002, http://www.positive-feedback.com/Issue1/kismet.htm.

on the Internet and from the datasheet itself. Using those points and the first 5 kΩ load, I got good sound and low power but it took some experimenting with the simulations, the load, and the bias to adjust and tune it. After all that, I settled on 250 V, 30 mA per tube (about -23 V), and a 3 kΩ load. Feel free to adapt these numbers to fit the batch you have.

For my Kismet project, the instruments should show 60 mA of current for each channel (30 mA for each 4P1L). I used the trimmers to fine tune the value. You will need some stabilization in temperature and break-in, and some matching first, of course. But I've seen that 4P1Ls from the same batch are similar (apart from filament resistance). So I got my biasing after a few hours of break-in and several switch on-offs and tuning. I also found that a very interesting point could be 300 V and 27 mA per tube (about -30V) with 5 kΩ load. But I did not test it on my bench until now.

Rember, CCSs take some seconds to warm up the filaments. But the damper diodes will take much longer to give some voltage. In my amplifier, it takes a bit less than 25 seconds before the instruments show some current and about 35 seconds to have stable current. This will change a bit in the following minutes. Don't worry if you see the current going up more than 60 mA, it will go down in a second.

Some Measurements

During testing, the frequency response at 1 W showed -0.5 dB at 20 Hz and -1.5 dB at 20 kHz, but it still needs some break-in. Then, the sine wave at 20 Hz, 1 kHz, and 20 kHz with some voltage output showed a pretty nice shape (see **Figures 3–5**). I took the measurements before the final improvements, so now they are even better.

The final note is the super soft clipping when the grid becomes positive: The driver can push it well into the positive side giving the required grid current making it soft. This makes the clipping visible on the negative side of the grid first.

In Conclusion

I have used the Kismet 5 for a few months in my system. My colleague Marco Pagogna said there has been a big improvement in sound after several hours of usage and a continuous improvement in sound quality, which has showed no sign of stopping after 100 hours. Pagogna also reported "a clear magic," perhaps typical of many single-ended amplifiers. "Lots of harmonics, uncommon richness, three dimensional, and all I need to listen to music," he said.

I might add that the bass response is really a surprise because it is very deep. Compared to my reference amplifier (which was several times the cost of the Kismet) and with my Goto Unit drivers, the Kismet 5 does lack a bit of precision and clarity. But, it will still enable you to enjoy your favorite music. Happy 20th anniversary to the Kismet design.

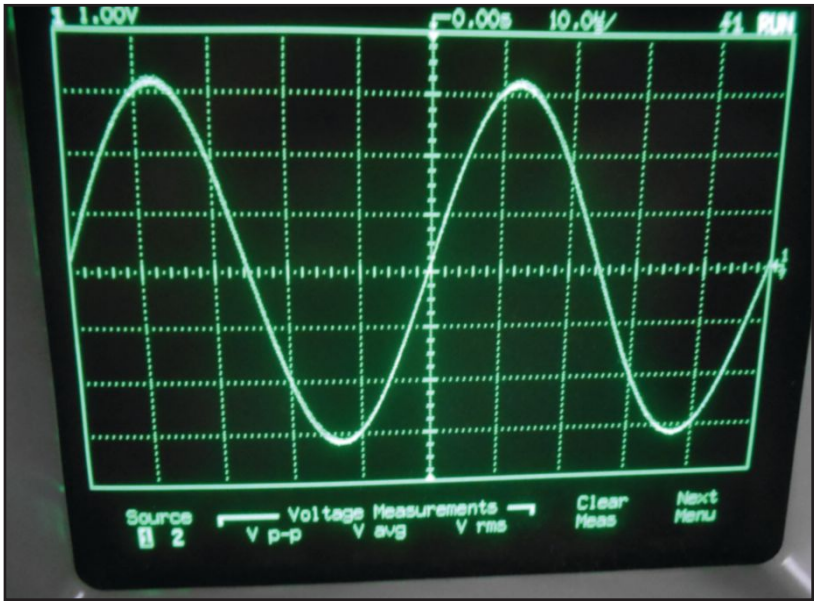


Figure 5: The Kismet 5's sine wave is shown at 20 kHz.

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High-Quality Hi-Fi Circlotron Power Amplifiers



In the 1950s, a special circuit principle for tube power amplifiers called "Circlotron" was invented. Designed primarily for studio applications, this amplifier was only moderately successful because of its greater component complexity and the associated higher price. Furthermore, the power output was lower compared to that of conventional push-pull AB amplifiers. This article describes power amplifiers based on the Circlotron principles and offer good hi-fi characteristics.

By
Gerhard Haas
(Germany)

When the Circlotron power amplifier was invented in the 1950s, the component selection was limited. Even if this power amplifier design had better basic characteristics than the traditional push-pull AB amplifiers, it was mostly used in studio settings. Today, a huge component selection as well as new materials are available, enabling Circlotron power amplifiers to reach high performance. Due to the more compact design, the electrolytic capacitors for the power supply can be generously sized. This not only guarantees good hum filtering, but also sufficient bass reserves in case of powerful bass impulses.

Where there only used to be half-wave rectifiers or selenium rectifiers in tube technology, today there are powerful silicon bridge rectifiers. They can be used for building compact and powerful high-quality power supplies. Likewise, the amplifier circuit can be generously dimensioned and equipped with high-quality components. Finally, the performance of modern mains transformers and output transformers is superior to those that were available when the Circlotron amplifier was invented.

The Design

First, I will explain the Circlotron amplifier's working principle in comparison with the standard push-pull AB amplifier. **Figure 1** shows the output stage ratios of a push-pull AB amplifier. Each power tube receives a signal half from the previous phase inverter. The two half-waves U1 and U2 are put together to form a full wave inside the output transformer. For each half wave, only one half of the primary winding is used. The total internal resistance R_I of the power amplifier is made up of the internal resistances R_I of the two power tubes as well as of the two plate resistances Ra. The internal resistance R_I is calculated as:

$$R_I = 2 \times R_I + 2 \times R_a = 2 \times \left(R_I + R_a\right)$$

The power amplifier is fed by a single voltage source U_B. If one tube is fully driven, the other one is blocked. Since the two primary transformer windings are identical, the output transformer changes the voltage U1 by the factor of 2 toward the other side, which also applies to U2. Therefore, at the blocked

tube, there is the voltage 2× U1 or 2× U2, respectively. If the correct load exists at the transformer's secondary circuit, the power amplifier works flawlessly. However, two extreme cases need to be observed if the power amplifier is at full level. The first extreme case is when the transformer's secondary circuit is shorted. In this case, the output voltage is automatically limited by the copper resistances of the transformer windings. Furthermore, the unstabilized voltage (U_B) will normally drop because of the heavy load, which is also beneficial for current limiting. If there is a total short of the secondary winding, the output transformer will become warm, especially if the amplifier is at full level for a longer time. However, the power amplifier will not suffer any damage if it is switched off in time.

The second extreme case is when the transformer's secondary circuit is not loaded, thus operated in idle mode. In case of full level control, the primary voltage at the anode of the blocked tube will increase to twice the value. Since the power load of the power supply is low without the secondary load, the voltage U_B remains high. However, the main problem arises if the transformer's ferrite rod is polarized in one direction by the half-wave. Now, if the other half-wave sets in, the magnetic polarity of the ferrite rod needs to be reversed. Thus, the magnetic energy stored inside the transformer core will be converted into electrical energy—resulting in high voltage peaks. If it is not possible to dissipate this energy in a defined way, there might be voltage flashovers inside the power tubes and at the tube sockets. Voltage flashovers can destroy these components and some resistors that are directly located next to the power tubes. It is not necessary to have a problem with all the components simultaneously, often it is sufficient if one component is faulty, which then trips the fuse.

Figure 2 shows the Circlotron power amplifier's circuit design, which requires two identical operating voltage sources U_B. With the design, the power tubes obtain a half-wave from the phase inverter. However, the output transformer's primary winding is always in full operation and alternately driven by both power tubes. The operating voltage sources U_B represent a short circuit for the alternating current. From an AC point of view, both power tubes are connected in parallel. This means the internal resistance only amounts to one-fourth of the internal resistance of a push-pull AB amplifier. Therefore, the output transformer's transmission ratio is lower. Also, the power amplifier is protected against shortening because of the afore-mentioned properties. As you can see in **Figure 2**, this power amplifier is also protected against idle operation! The output transformer can only be maximally driven with the voltage U_B because

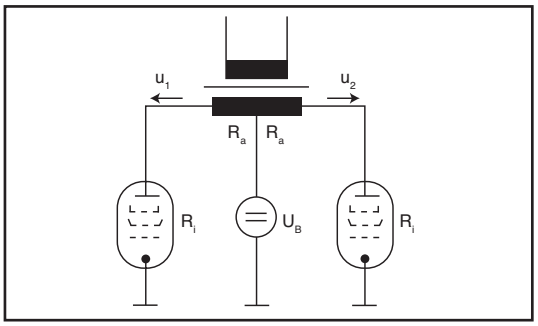


Figure 1: These are the output stage ratios of a push-pull AB amplifier.

it is driven with alternating current and because the magnetic energy is completely dissipated while reversing the polarity of the alternating current being fed. This means the voltage cannot increase as with the classic push-pull AB amplifier. For example, if a mains transformer is connected to the power grid, the secondary voltage without a load will only be slightly higher than with a load, however, it will not increase. That is how the Circlotron power amplifier behaves without a load, which means the circuit design is very reliable in operation.

Design Limitations

One problem with Circlotron power amplifiers is their efficiency. If enough tubes are connected in parallel, the output transformer can be omitted. So, there is enough current available to operate the speakers with an impedance of 8 Ω or less. The speaker is directly connected to the tube cathodes. So, the frequency response is not limited by the transformer characteristics and will exceed the audio range, if properly designed. There is a series of power amplifiers built this way. At full power, the current consumption is up to four times the output power. Therefore, the efficiency is only about 25%! The efficiency decreases because of the heating power of the many tubes. Therefore, there is a substantial heat build-up with higher amplifier power outputs.

Another option is to replace the transformer with an inductor tapped at the center. The center

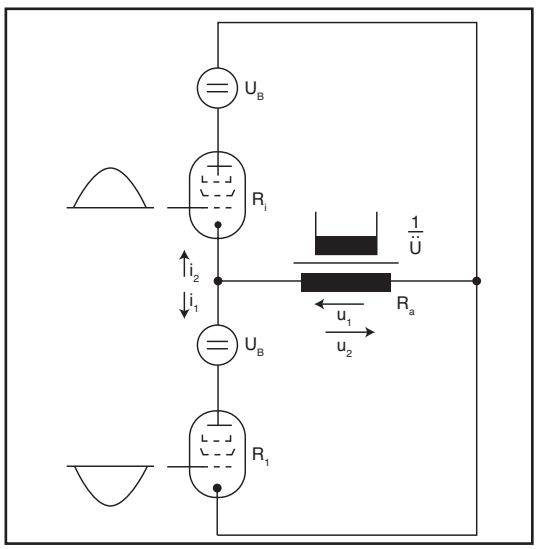


Figure 2: This is the circuit design for the Circlotron power amplifier.

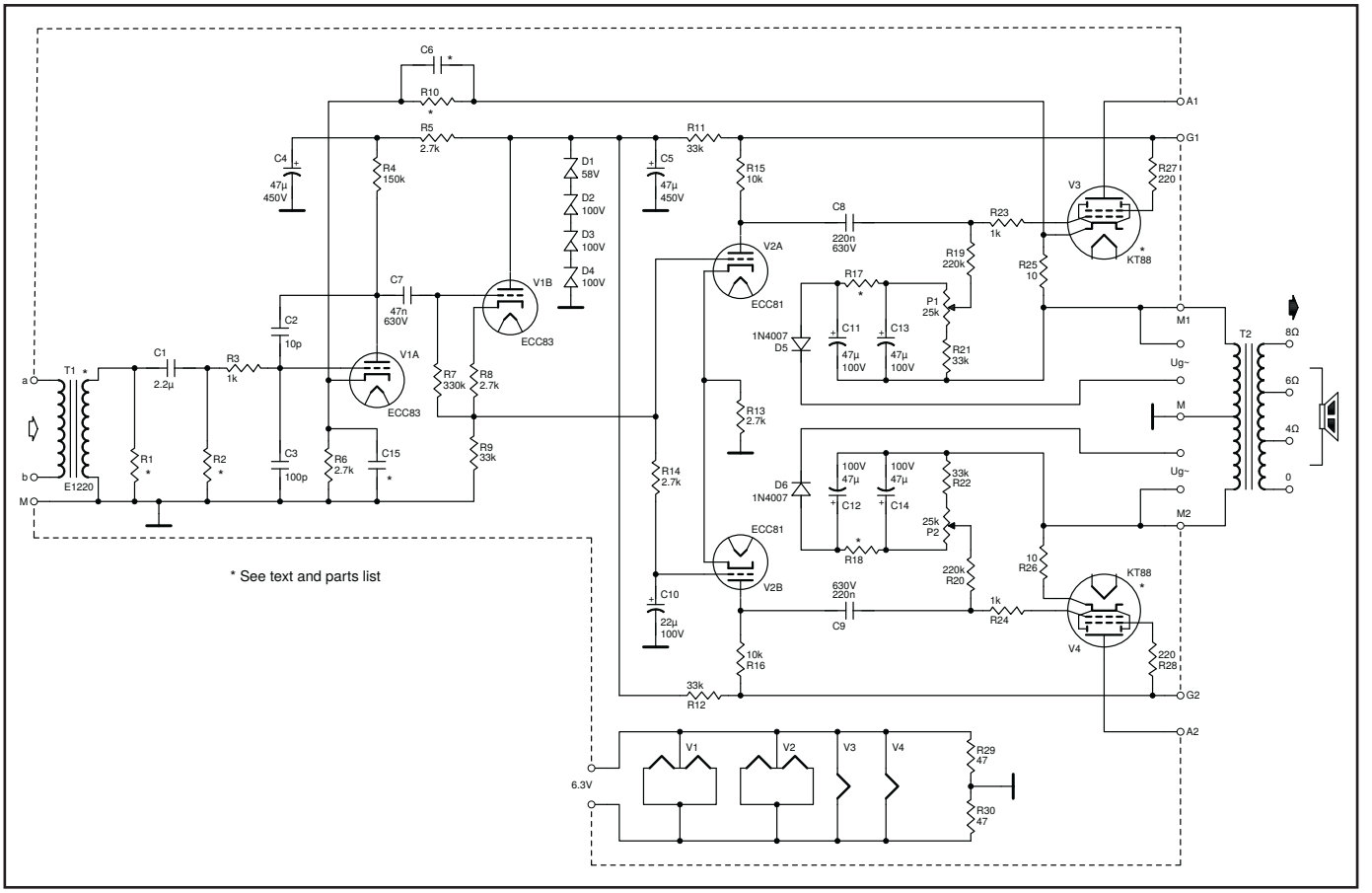


Figure 3: The amplifier circuit is shown with two power tubes.

tap must be connected to ground, which means two supplementary tubes are sufficient. The transformer acts like a tapped auto-transformer and transforms the higher tube output impedance into the lower speaker impedance, making the efficiency considerably higher.

The problem with both options is that there is no galvanic separation between the amplifier section with high internal voltages and the speaker. If there is a failure inside the amplifier, the high voltage may appear at the speaker terminals. This could lead to life-threatening injuries and destroy the speaker. The power amplifiers I am presenting are equipped with electrically insulating transformers that are checked for 4,000 V between the primary and the secondary windings, offering a high degree of safety. By optimizing the mains transformer, the power supply, and the output transformer, the total efficiency is approximately 30% at full level control.

The Circlotron power amplifiers also had frequent problems with residual humming at the amplifier output. When this power amplifier design became popular in the 1950s, half-way rectifiers were preferred. Normally, the electrolytic charging capacitors were dimensioned to 50 µF and less, because the rectifiers could not tolerate high voltage peaks. But, using tubes to build bridge rectifiers was disproportionate. Selenium bridge rectifiers were only available to a limited extent or not available at all. For one-way rectification of the nominal voltage, the hum voltage is twice as high as with bridge rectification. Today's powerful and surge-resistant silicon rectifiers enable the use of charging capacitors with high capacitances, keeping the mains hum low.

As good as the silicon rectifiers might be, especially with the Circlotron power amplifier, there is another problem that would not normally impact other amplifier circuits because the signal current doesn't flow via the electrolytic charging capacitors.

About the Author

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



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The charging capacitors maintain a certain voltage. If the voltage delivered by the transformer becomes higher than the voltage of the charging capacitor, the rectifier diode will open and stay open until the transformer voltage's peak value is reached. Once the transformer voltage drops, the rectifier diode will close. However, before it completely closes, the charge carried inside the barrier layer would need to be cleared out. For a short time, the diode represents a short circuit. As long it is in short circuit mode, the current flows from the charging capacitor back into the mains transformer. In this way, spikes are generated at double the rate of the mains frequency that break through the amplifier circuit, appearing to be its output. These spikes will not only be perceived as humming with double the mains frequency because of the bridge rectifier, but due to its signal type there are harmonics that can even be heard in the tweeter. These annoying spikes can be efficiently suppressed. I will be described this in greater detail when we get to the power supply circuit.

Using Two Power Tubes

Figure 3 shows the amplifier circuit with two power tubes. The input features a transformer. In this way, it is possible to feed the audio signal. It is particularly advantageous when the amplifiers are built as monoblocks. This avoids ground loops. This one is coiled 1:1+1, which enables the secondary windings to be connected in parallel or in series. The PCB layout includes wire bridges so that the input transformer can be operated in 1:1 or 1:2 mode. The assembly layout makes it clear that there are four solder points on the transformer's secondary side. When inserting a wire bridge in the center, the transformer works at the ratio of 1:2. The input sensitivity is doubled, the gain is virtually increased by 6 dB without noise. This can be favorable when connecting low-level signal sources. However, normally the transformer is only used for galvanic isolation and is operated in 1:1 mode. To achieve this, the two outer wire bridges are inserted. When using the transformer, R2 will be omitted and another wire bridge is inserted for C1. For R1, use a resistor with 33 kΩ, which is a real load to the transformer and provides a linear frequency response. The transformer and R1 form the grid bleeder of V1a. Without transformer C1, use R1 and R2. R1 provides DC discharge of C1. C2 forms the grid bleeder of V1a. If no transformer is used, C1 should not be omitted. Otherwise, if DC components come from the signal source, the working point of V1a will be shifted. R3 and C3 suppress high-frequency interferences.

To suppress the oscillating tendency, at V1a, C2 is connected from the anode to the grid. This tube contributes to the signal amplification. V1b is connected as an impedance converter and forwards the signal at low impedance to the phase splitter V2, which is designed as differential amplifier. The grid of V2b is AC connected to ground via C10. Signal control is accomplished via the cathode. The Circlotron design requires every power tube to have

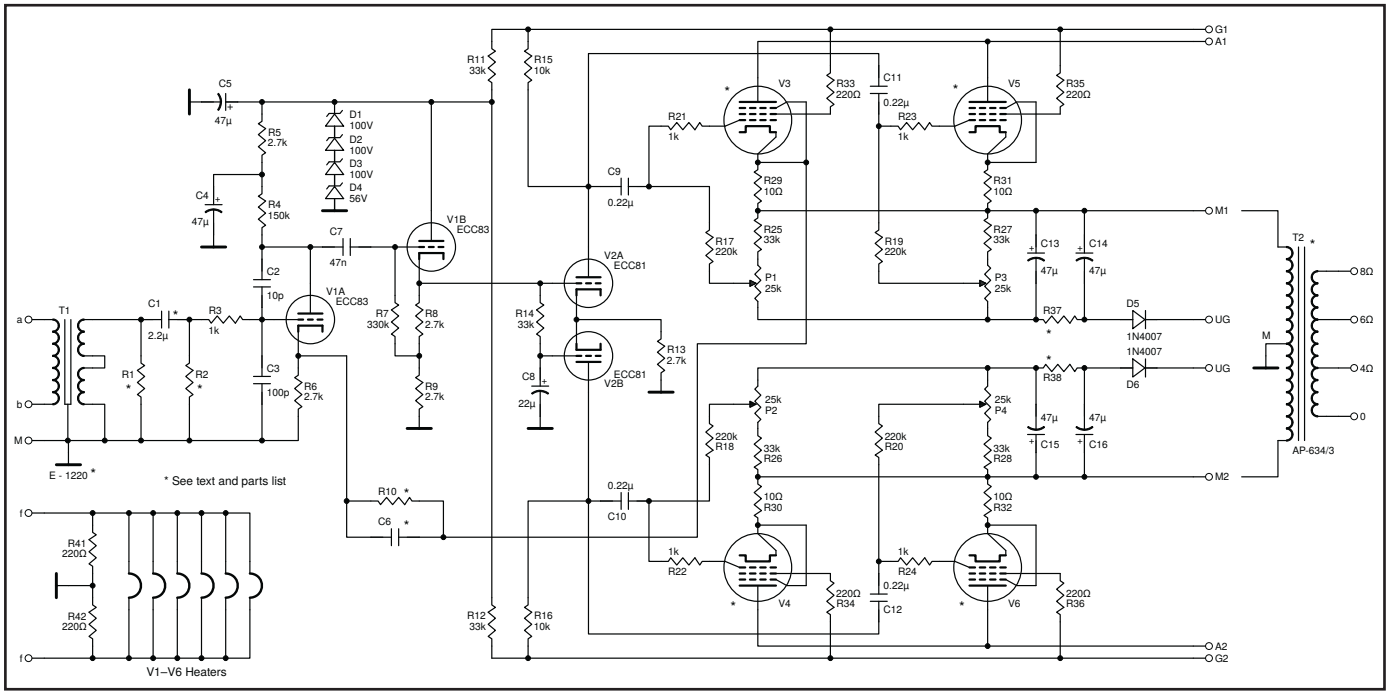


Figure 4: This circuit design corresponds to the configuration shown in Figure 3.

its own grid bias. For the grid bias generation, two separated transformer windings Ug~ are needed. Use the trimpots P1 and P2 to set the idle current. It would also be possible to insert a resistor bridged by a capacitor in each cathode pipe of the power tubes to generate the grid bias. Depending on the tube type, approximately 30 to 50 V would drop here, which would then not be available for the maximum signal level at the amplifier output. In this case, the output power drops by about 20%. It is possible to measure the idle current of each power tube via the voltage drop at the cathode resistors R25 and R26. If the idle current is supposed to be 65 mA, for example, the voltage drop at the cathode resistor must be 650 mV.

The negative feedback is achieved via R10. Choose its value so that the circuit works in a stable manner and offers good distortion. If there is an oscillating tendency, a capacitor with 10 pF can be used for C6. If that is not sufficient, C6 can be increased. However, first check whether the oscillating tendency is caused by an unfavorable amplifier design and cable routing. The input voltage to drive the amplifier up to a total harmonic distortion (THD) of 1% is 1.46 V, which is a value that virtually any preamplifier can deliver.

The operating voltage of V1 is filtered by R11, R12, and C5, and stabilized with the Zener diodes D1–4. In this way, V1 always has a stable and hum-free operating voltage, regardless of the control. R5 and C4 decouple the operating voltage of V1a. If the level drops with high frequencies, it could be due to the design and to the use of an output transformer. This can be compensated by using C15, which should be a film capacitor. The Parts List gives the reference values. It can be found in the Supplementary Materials section of the audioXpress website (www.audioxpress.com).

Two More Considerations

There are two more important things to consider. Depending on the manufacturer and the type of tubes, it is possible that the idle currents can't be correctly set. For example, you might be able to set the idle current of a KT 88 from manufacturer A, whereas the tubes from manufacturer B might pull too much current even if the trimpots are set to maximum. Furthermore, depending on the batch of tubes, even from the same manufacturer, the idle current setting might prove difficult. If this occurs, then R17 and R18 need to be modified so that a safe setup is possible. Instead of trying it with expensive tubes, it is easier to adapt the resistors. If it's impossible to set the correct idle current setting, it is probably not due to a faulty circuit, but because the tube tolerances are substantial. You should also check whether the circuit shows an oscillating tendency. If so, it will be impossible to set the correct idle current.

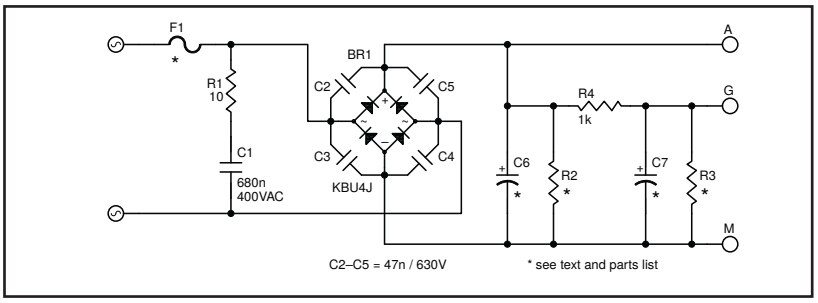


Figure 5: This power supply circuit is suitable for all Circlotron power amplifiers.

The circuit can be operated using the EL 34 and the more powerful E 34 L. The measured values specified in the technical data (provided in the Supplementary Material) are based on the E 34 L. It is also possible to use the 6550 A instead of the KT 88 without any modifications. If necessary, the 6L6 GC can be used instead of the EL 34, however, you should check the harmonic distortion's behavior. With two KT 120 tubes, you can achieve an output power of 80 W. The type of tubes you use is a matter of taste. To me, the bulb shape of the KT 88 is attractive, and the KT 120 is even more so. The rated power at low frequencies can only be transmitted by a bigger core with a larger iron core cross-section and the corresponding winding. The MD 85 B core has a sufficiently large iron cross-section and can be wound with thicker wires, which reduces the copper losses and increases the power output.

If you want to use the slightly smaller MD 74 core due to lack of space or for economic reasons, you must give up some power. No matter which transformer you use, the circuit dimensioning basically stays the same. Due to the higher copper resistance of the smaller transformer core's windings, the output power is slightly lower. The present measurement results

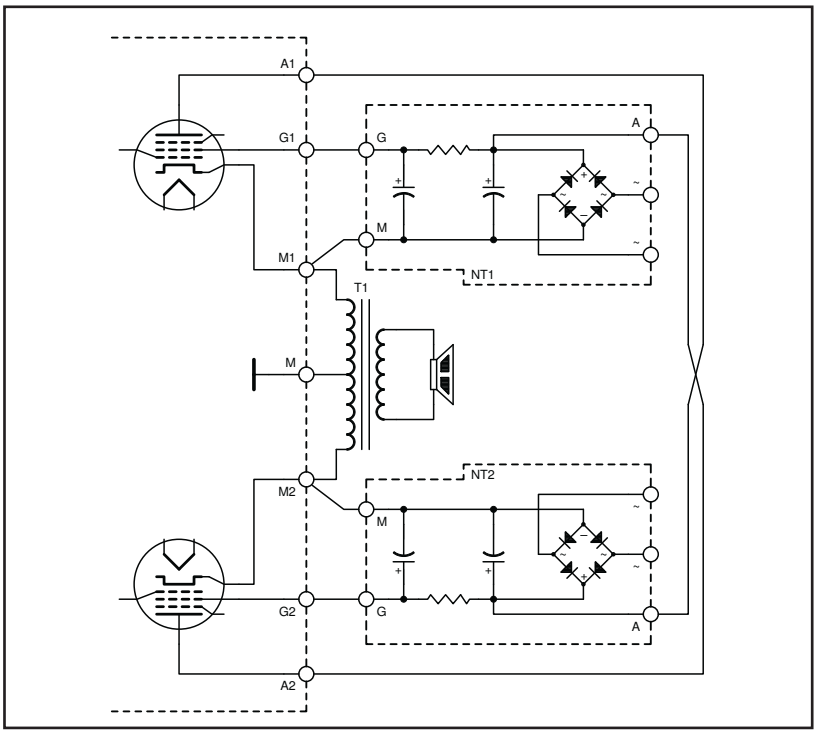


Figure 6: Two power supply circuits are needed for each power amplifier.

were determined using a 600 Ω source resistance of the generator. The technical data also specifies values that can be reached with the smaller transformer.

The amount of the harmonic distortion as well as its spectral distribution depends on the tubes you use. Depending on the manufacturer and the design, the power tubes—especially the ECC 81—can have a significant impact on the distortion properties. The ECC 81 was developed as a VHF tube. Processing frequencies around 100 MHz requires the tube capacitances to be low. The original ECC 81 has small anodes. The anode plates are about 11 mm high. The market offers different variants of the ECC 81 with different system configurations. Some versions of the ECC 81 feature the large anode of the classic ECC 83 with a height of about 16 to 18 mm. These tube types can cause a higher harmonic distortion factor, especially the third harmonic. In this case, changing the tube type and the manufacturer will help.

Figure 4 shows the circuit that corresponds to that configuration shown in **Figure 3**, which is only expanded by two more power tubes. By doing so, it is easy to reach a power output of more than 100 W. In contrast to the version with 2× KT 88 or 2× EL 34, the achievable output power is almost doubled. Now, you only have to decide whether you prefer 80 W with 2× KT 120 or the visual appeal of four power tubes and slightly more output. Basically, everything stays the same—except two more power tubes are used and the power supply and the mains transformer matches the power requirements. The output transformer also needs to have a bigger core to transmit the higher power at lower frequencies.

The Ultimate Power Supply Circuit

Figure 5 shows a power supply circuit suitable for all Circlotron power amplifiers. It is suitable for all the power amplifier options that we have mentioned. The placement is adapted to the power amplifier type. Each Circlotron power amplifier requires two power supply boards. As already mentioned, measures must be taken inside the power supply to suppress spikes. An efficient way to achieve that is by bridging each rectifier diode with one capacitor. This suppresses the current backflow form the charging capacitor C6 toward the transformer winding, which would otherwise occur each time the diodes block. Since the power source is inductive due to its mains transformer and the load is capacitive, and the current flows through a rectifier diode with non-linear characteristics, there is a phase shift between the current and the voltage that leads to further disturbances.

Use R1 and C1 to compensate for this. In contrast to classic amplifier circuits, the signal current flows directly through the respective power tube and the charging capacitor of the power supply, in this case through C6. There should be little interference voltage, otherwise it will appear at the amplifier output. The voltage for the screen grid is decoupled and further filtered via R1 and C7. After switching the operating voltage off, R2 and R3 ensure that all capacitors are safely discharged.

The Parts List, provided with the Supplementary Material, contains proven reference values for the compensation units. Depending on the circuit design, the amplifier circuit layout, and the mains transformer, it might be necessary to make adjustments. Choose the dimensioning with the least interference voltages. Basically, you can assume the given values to be correct. If the interference voltage values are okay, no further measures need to be taken. Useful measurements can only be carried out using the power supply and the amplifier featuring a proper design.

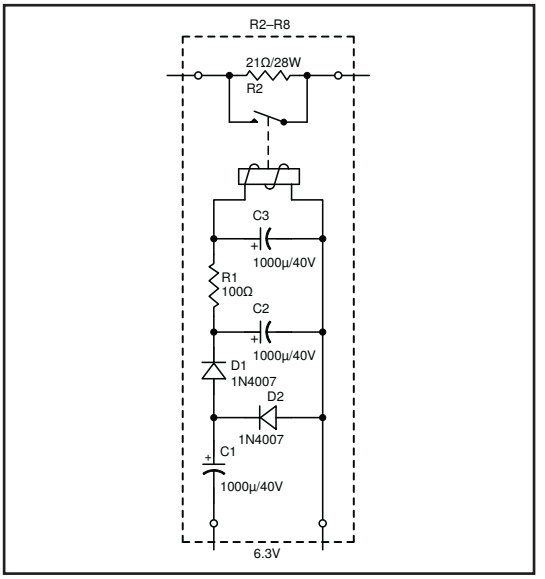


Figure 7: I recommend using the power-on current limiter with the Circlotron power amplifiers.

Start Up

When starting the Circlotron power supplies, there is another very important consideration. The mains transformers listed in the power amplifier Parts List have very low impedances. The electrolytic capacitors inside the power supply feature high capacitances that are necessary for good hum filtering and they also need to have sufficient reserves for playing back powerful bass impulses. If the electrolytic capacitors are stored for a longer time, they could show an increased leakage current. When you switch on the power supplies, the high-capacitance discharged electrolytic capacitors with increased leakage current almost act like a short circuit. Even if the fuse is correctly dimensioned, it can trip.

During the first startup, replace the fuse with

a resistor that is about 470 Ω to 1 kΩ and can withstand at least 10 W. With that, only the high-voltage power supply is operated at the intended mains transformer for up to 30 minutes. In this way, the electrolytic capacitors start up in a gentle way and the leakage current reaches its low value. The resistors may become warm due to the leakage current and because of the phase shift. Therefore, I would recommend resistor types with ≥10 W. After the initial startup, the resistors can be replaced by the respective fuses and the power supplies can be supplied with voltage for some time. This way you can safely start the electrolytic capacitors, the leakage current can reach its normal value, and the power supply is ready for operation.

When handling the electrolytic capacitors, you should make sure that they are sufficiently discharged. Capacitors that are already 47 µF charged at 400 V can cause an electrical shock and injuries! You should use a measuring instrument to check if the voltage has reached values of about 30 V or below, then it is possible to continue the work without risk.

The power supply circuit is required twice for each power amplifier! **Figure 6** shows the correct wiring. The high-voltage power supplies must be consequently connected as shown in the picture. The most important aspect is that the anode lines are wired cross-wise. This is the only way the layout will work correctly.

One final important point. When switching on parallel push-pull (PPP) power amplifiers, several things will happen in the moment they power-on. The transformer's iron core must be magnetized. The transformers we used featuring the MD cores have a very low impedance, which increases the power-on surge. Furthermore, directly behind the rectifier, there are relatively high capacitances that must be charged first and that represent almost a short circuit during the moment of power-on.

Finally, in contrast to warm tube heating, this tube heating features only a sixth of the resistance when it is cold. These factors taken together cause a high-power surge at the moment of power-on. Although the mains transformer of the largest PPP power amplifier presented here only features about 300 VA, the circuit breaker might trip. However, the time-lag fuse inside the amplifier will not trip, although its current value is clearly lower.

I would recommend that you use the power-on current limiter shown in **Figure 7** with the Circlotron power amplifiers. The heating voltage with 6.3 V is rectified by the voltage doubler circuit D1, D2, C1, and C2. R1 is dimensioned so that during operation 12 V are present at the relay coil. C3 and R1 together provide a delay. The time until the voltage at C3 ramps up and the relay is energized takes about one second. Then, the mains transformer is magnetized, the power supply's electrolytic capacitors are charged, and the tube filaments heat up increasing their resistance. Next, the current limiting resistor located in the mains line consists of the parallel connection of R2–R8 and is bridged by the relay contact. We used seven resistors with a load-bearing capacity of 4 W, which avoids overloading. The relay features two switch contacts connected in parallel, which provides a lower impedance. But, you are assured that if one contact fails, the second one will still work. If total contact failure occurs, the resistors would overload and burn through. **Figure 8** shows how to integrate the circuit into the amplifier setup. ax

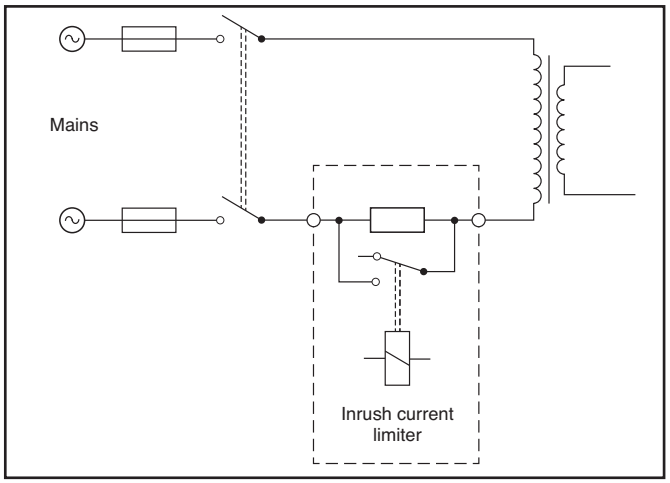


Figure 8: Use this design to integrate the circuit into the amplifier setup.

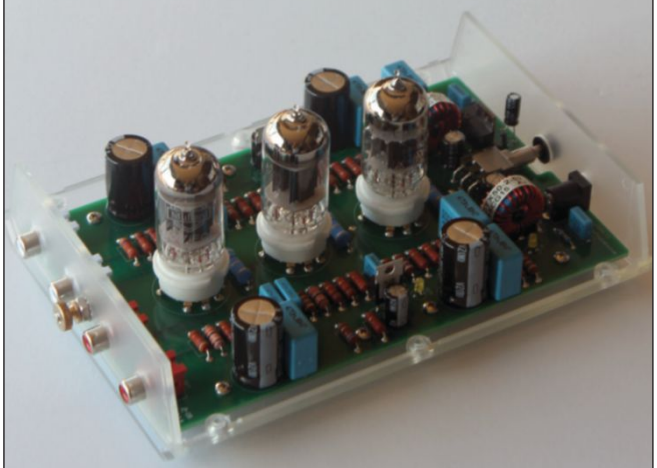
Project Files

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Four Gadgets for Tube Enthusiasts



Photo 1: These four tube amplifier gadgets will help you keep your audio equipment finely tuned.

I want to encourage and attract new hobbyists to the fascinating world of tubes. To help new tube enthusiasts get started, I want to recommend four tools that you can build yourself and that will be immensely useful when constructing your own tube projects.

By
Bruce Brown
(United States)

I build, service, and restore tube audio and guitar amplifiers. There are many tools and tricks that are helpful in setting up or servicing tube equipment. First, you need a good digital voltage meter (DVM), a signal tracer, a sine wave generator, a variac (metered is best), and a variety of hand tools. An oscilloscope is also handy, especially in tracking down parasitic oscillation, but it isn't a first-buy tool.

Over the years, I have built many of the projects that appeared in *Glass Audio* and its successor *audioXpress* to help me work with tube gear. Almost every week, I use the electrolytic reformer/tester that I built many years ago (see Resources). I also have several little devices that make it easier to work on tube gear and want to share these helpful accessories. They are all inexpensive and straightforward to build. I will also tell you some ways that I use them and why in an effort to encourage and attract new hobbyists to the fascinating world of tubes.

The four tools I am going to show you how to build and use include: a line voltage monitor, a variable resistance box for setting bias and decoupling resistor values, an inexpensive bias meter, and a simple tool for determining the impedance of unknown transformers (see **Photo 1**).

Line Voltage Monitor

Many tube amplifiers were built when the AC line voltage was in the 110-to-115-V range. This presents some real problems with older tube equipment power supplies and tubes. The line voltage in my area varies from a low of 119 to 123 VAC, depending on the time of day. If the equipment was designed for 110 V on the primary with 495 VDC on the first section of the filter capacitors and today's voltage is 120, you may well be in the range of 525 V, on capacitors rated at 500 V. Even if the capacitors are rated at 525 V, the voltage may go as high as 550 V until the output tubes warm up.

I have a good AC volt meter on my variac, so I can run the line voltage where ever I want, but it doesn't tell me the current line voltage. One day I was surfing the Internet and ran across some self-powered AC digital LED voltmeters with a range of 0 to 200 V. They were inexpensive and I bought a couple to try. They come on at about 20 V and are surprisingly accurate.

All you need to build this device is one of the modules, a small plastic utility box, and an AC plug that you can separate. Most small plastic utility boxes come with both an aluminum and a plastic front cover, I would suggest you use all plastic—it is much easier to use, and can be cut with many different tools.

Start by cutting an appropriate hole in the bottom of the case for your AC plug. You can start the holes with a drill bit and finish with a file or you can cut the hole with a scroll saw.

I use marine "Goop" to glue in things like this. Once it sets up, it is hard to remove, so get it where you want. Let it set up for about 24 hours. Carefully trace the outline of the bezel on the front cover and cut the hole. (Most of these modules come with a bezel for the front and the meter module mounts from the back.) Connect the wires and you are ready to plug it in. Some modules have a trimmer pot to adjust the calibration, but mine agreed with both my bench meters (see **Photo 2**).

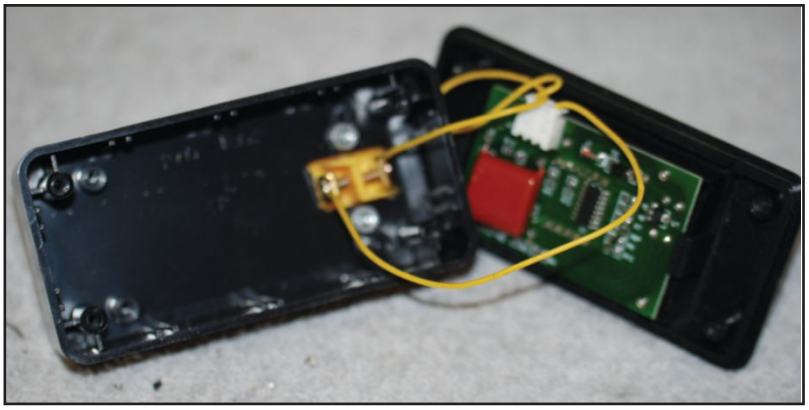


Photo 2: This is an interior view of the line voltage monitor.

Larry's Bias Box

This variable resistor substitution box enables you to fine tune a preamp/AF amplifier section by making it easy to adjust the power supply decoupling resistors for the tubes and the power supply transformer you have chosen—especially if it came from your junk box and does not have the exact voltages you wanted. It also is helpful when selecting cathode resistors in power amps with "automatic" bias, as well as resistors in "fixed" bias supply systems. You could use a commercial resistor substitution box, but they usually use 0.5 W resistors and are sometimes quite large and cumbersome. In the past, I have soldered some clip leads to a 2 W pot and used this for the selection of bias resistors. This obviously presents some danger with high voltages and is not recommended.

But, there is another option. A couple of years ago, an old friend and "tube buddy" died suddenly and his family asked me to help sort some of his equipment and parts. One of the pieces of equipment he had built was a compact BUD-style box with three 2 W pots inside. I had seen this on his bench in the past and had asked him about it. He said it was for setting bias resistors and compensation networks and offered to make me one, but he never had a chance to do it. So in his honor, I want to take you through the construction and use of the "Larry Box" (see **Photo 3**).

If you have a well-stocked "junk" box you probably have all the parts you need. I used 1 kΩ, 10 kΩ, and 100 kΩ, 2 W pots, but you can add a 1 MΩ unit to give your box a range of up to 1,111 kΩ. One of my pots had a locking nut so I used it in the 100 kΩ pot. You will also need a pair of binding posts or pin jacks. I used a pair of them so that I could easily connect my DVM to one set and the circuit to the other set. I also had a new BUD box, so I spaced the pots side by side, but you can do it however it makes sense to you.

Wiring is simple, just series wire each pot with the next. (Make sure you wire the pots to give increasing resistance as the pot is tuned clockwise.) Your binding post connections are connected to the lugs of the first and last pot. I used a label maker to identify the value of each pot on the front panel. Connect your ohmmeter to the binding posts and check to make sure you have the variable range and action you need (see **Photo 4**).

A word of caution when using the bias box, especially if you are using it to set bias voltage on output tubes. Use your DVM to preset a value in the range you anticipate for the circuit. For setting "fixed" bias resistors, it is better to start at a higher negative voltage on the grids of the output tubes than you

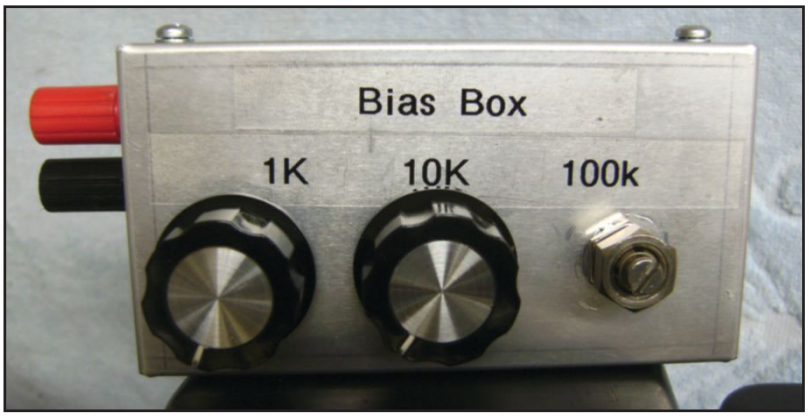


Photo 3: Larry's Bias Box enables you to fine tune a preamp/AF amplifier section by making it easy to adjust power supply decoupling resistors for the tubes and the power supply transformer.



Photo 4: If you have a well-stocked "junk" box you probably have all the parts you need to build this variable resistor substitution box.

anticipate. This will prevent the output tube(s) from getting the dreaded red plate glow.

Author's Note: Always have a load on the output transformer, no signal on the input, and be extremely careful taking these readings as they are very high voltage close to where you are working!

Fixed Bias Circuit

Some amplifier designs ground the cathode or have a closed circuit jack between the cathode and ground, alternatively a small value resistor is connected between the cathode and ground. To use this procedure, you need to have a resistor between the cathode and ground (or build the adapter discussed later). I commonly use 1 Ω, 5%, 2 W resistors. The reason I do this is it makes it easy to determine cathode current. If you measure 20 mV across the resistor, the cathode current is 20 mA.

Connect the box between the bias voltage supply and between the balance resistors going to the grids of the output tubes. Connect your DVM or other meter across the 1 Ω resistor. Slowly bring up the voltage on the amplifier, watching the bias voltage and the cathode current (detailed in the next project) on your DVM.

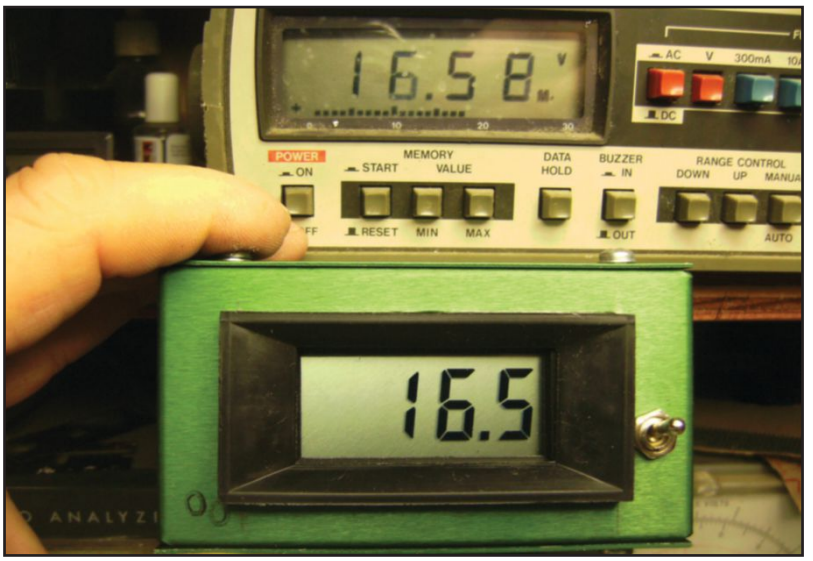


Photo 5: The Tube Bias Meter is a small digital meter with tube socket adapters for measuring output tube cathode current.

Adjust the bias box, while monitoring the cathode current until you have it where you want it. Tube manuals list the cathode current maximum for output tubes. I try for no more than 70% of this number to lengthen the life of the tubes.

Guitar amps typically run at maximum cathode current for the best sound, but it does shorten tube life. I usually will let the amp idle for 20 minutes to make sure everything is working correctly. Power the amp down and discharge the filter caps. Disconnect the box and measure the value of the needed resistor. Select a fixed resistor as close as you can get and wire it in (or if you want to have an adjustable bias use a pot or a cermet trimmer). Recheck your current with the fixed resistor in place.

Some commercial amplifier schematics only show the output tube grid voltage and don't specify the cathode current. Some amplifier schematics only show the grid voltage. If this is the case, just use your voltmeter to attain the specified bias voltage measured on the grids of output tubes. (This is common with McIntosh Power amps and guitar amps) Once you get the correct voltage on the grids, you can replace the pots with a fixed resistor.

You can also use this box to vary the cathode resistor value on single-ended output tubes (and small push-pull amps with "Automatic" bias) and see the effects of varying the output tube current draw. Disconnect the cathode resistor. (Leave the electrolytic cap in place, if there is one) Adjust the pots for approximately the same resistance as the resistor that was there and power up the amp. Listen to the amplifier and vary the bias a bit and you can hear the difference. Watch the plates of the output tube(s) and if they start to glow, go the other way.

You can measure the actual cathode current with a method called "shunt bias." Set your DVM for current measurement and connect one side to the plate of the output tube and the other to the B+ feeding the output transformer. Again, have a load on the amplifier output but do not feed a signal to the input. Adjust the pot(s) while watching the cathode current. Do not exceed the current recommended in your tube manual. I generally run about 70% of the rated plate current. Guitar amps (vintage ones) tend to run the cathode current at or above the maximum recommended value to get that "sweet spot."

Many guitarists believe that amps sound best with the output tubes run at full blast. And, the amplifiers

were designed when tubes were cheap so replacing them wasn't a big issue. Many single-ended amps used a 2 W carbon comp resistor for the cathode bias resistor. If you measure the value in many of these amps, the values have drifted a long way and the tubes may be running cold (not great sounding) or hot (glowing plates). Getting the value of the cathode bias resistor right can have a dramatic effect on the amplifier's sound.

I once had a customer who owned a Fender Champ from the 1970s that he asked me to recap because it had developed a little hum. The cathode resistor had drifted from 150 Ω to more than 400 Ω. I selected a cathode resistor to give me about 90% the cathode current for a 6V6 tube. He called me up the same night he got the amplifier back and wondered what I had done to it? He was thrilled: He said it hadn't sounded that good for the entire time he had owned it.

Tube Bias Meter

I recently noticed that several online vendors were selling small digital meters with tube socket adapters for measuring output tube cathode current. After I thought how cool that was (not having to lift cathode ground and insert 1 Ω resistors), I realized that the price was pretty high. I dug up a used BUD box, a digital panel meter, and a few odd parts and built my own (see **Photo 5**). I had everything on hand, but even if you have to buy everything new, it will cost less than $20 to build. If you want to use your DVM instead of building the standalone meter, skip this section and go on to building the socket adapters.

You can build your meter as a single measuring circuit, or do as I did and put in a double-pull double-throw (DPDT) switch with a pair of jacks for the leads to the tube socket adapters. This will enable you to switch from using the socket adapters (discussed later) to the binding posts, for direct voltage measurement.

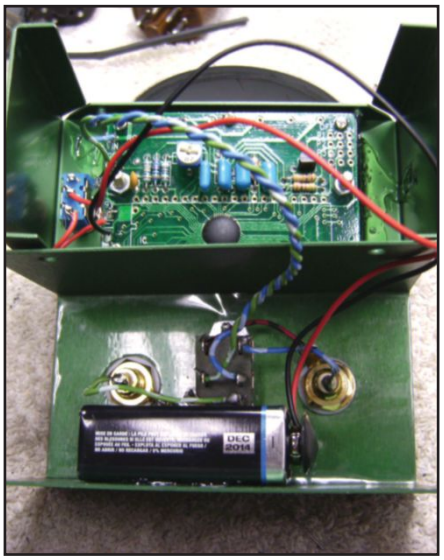


Photo 6: I used a pair of RCA jacks with insulator washers, to isolate the jacks from the case.

I used a pair of RCA jacks with insulator washers, to isolate the jacks from the case (see **Photo 6**). I used a nibbler to cut out the opening for the PM-128 meter in the other part of the case and I wired everything. Before you assemble the box, you will need to connect some jumpers and resistors to set the meter's range.

My meter is a P-128 model, which is adjustable for 200 mV, 2 V, 20 V, or 200 V. The directions that come with these meters are poorly translated (or maybe were badly written in Chinese and perfectly translated). I wired mine for 20 V with a single decimal point. This required taking out a bridge, adding two resistors, and putting in a bridge for the decimal point. I then wired a small toggle switch and a 9 V battery connector.

I checked the meter accuracy by connecting a known voltage to one of the connectors and verifying the accuracy of the voltage reading with an accurate DVM. You can make minor adjustments to the meter with the small trim pot on the back of the unit. Once everything checks out, assemble the box, and set it aside for now.

About the Author

Bruce Brown has been a hobbyist in electronics for more than 45 years. When he was 12, Bruce started collecting discarded TVs and using the tubes and parts to build things. He continues to recycle electronic items to build new projects. He restores vintage tube equipment and builds custom hi-fi and guitar amplifiers. He has two websites: www.vintagetubeaudio.com and Junkwerks.us. Bruce can be reached at Tuninfork@aol.com.

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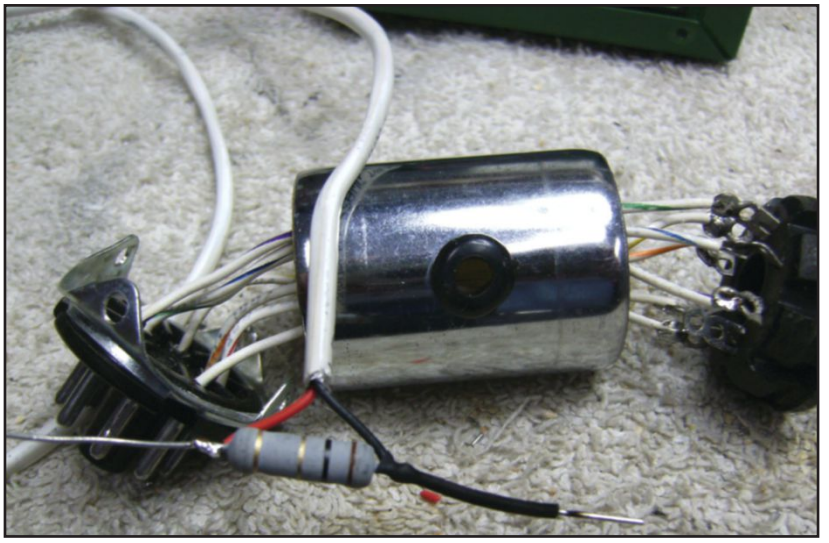


Photo 7: The Tube Bias Meter's interior shows the double-pull double-throw (DPDT) switch and a pair of jacks for the leads to the tube socket adapters.

The most difficult part of this project was making the tube socket adapters. You will need to make adapters for each tube and type you use. I use a lot of 6L6, EL34, and 6550 tubes, which all have the same basic pin connections. If you want to make your own octal ones, it is pretty easy. Find some bad octal tubes, break the glass (be careful to cover the tube with cloth and carefully dispose of the glass) and unsolder the pin wiring. Then, find a suitable octal socket and something to mount between them. I used a piece of metal sink trap pipe and glued the finished adapter together with "Goop" (see **Photo 7**).

Start by cutting and stripping enough wires to connect the pins from the socket to the plug. In place of the wire, use a 1 Ω, 2 W resistor in between the pins for the cathode connection (Pin 8 for 6L6. EL34, and 6550). Make sure you insulate the resistor wires and connections. Before you glue everything together, solder a piece of two conductor zip cords to each side of the resistor, these will be your leads to the meter.

You can also buy tube socket "savers" online for both octal 7" and 9" tubes. Make sure that the ones you buy can be disassembled. I contacted several sellers before I found a company that had both ones that had been glued together and ones that weren't. The unglued ones are commonly used to make adapters when replacing an obsolescent tube with a different one so you do not need to rewire the chassis socket. Use the same procedure when installing the 1 Ω resistor in each adapter between the cathode pins from the socket to the plug. If you commonly use tubes with different pin connections, you can make adapters for each type. (Be sure to label each for which type of tubes they fit.)

You might also want to make up a cord with a small set of alligator clips on one end, with the appropriate connector to plug into your new tool. This enables you to clip into a circuit temporally or to just use it as a 0 to 20 VDC voltmeter. I also made up a wire set with a one-quarter phone plug on one end for use with some of the older amplifiers that have the bias jack already installed.

I have a few suggestions on the use of the bias meter:

- Some amplifier schematics do not identify the cathode current for bias adjustment, instead they specify the voltage that is to be applied to the grids (e.g., McIntosh in its MC 30s, 40s, etc.) You can still use this device to measure cathode current, but you can also just use your to measure the voltage on the grids.
- Some experimenting with amplifier sound and performance, at different bias settings, is fun and quite interesting. Do not go to extremes. Do not exceed the cathode current specified in your tube manual. Also be sure to watch the plates of the output tubes, if they start glowing, shut the amplifier down.
- If you do not feel comfortable building your own adapter, you can buy a complete kit from the Tube Depot (www.tubedepot.com). You can use this adapter with the meter that I described how to build or with your DVM. It works exactly the same, plus it has some additional features.

Impedance Box

Often, I have bought output transformers of unknown linage or pulled them from amplifier chassis I have purchased. A simple circuit and four-function calculator will enable you to determine the primary impedance of any transformer. You will need a decent DVM to measure AC voltage, in addition to a simple circuit (see **Photo 8**). You need a 6.3 or 12 V filament transformer—the current rating is not important, 0.5 A is more than enough. It is helpful to have a variac, but it is not absolutely necessary. Connect an AC cord to the primary of the filament transformer. I used a surplus BUD box and mounted some binding posts to the filament transformer secondary. This was done to make it easy to connect the primary of the unknown transformer and the AC voltmeter.

Before you use the box, first use your ohmmeter to determine the primary and the secondary. The highest resistance is the primary (if it is a push-pull output you will also have a center tap) and the lowest is secondary. You may have more than one secondary, depending on whether it has 4 Ω, 8 Ω, and/or 16 Ω secondaries. If this is the case, I usually use the middle resistance, assuming an 8 Ω secondary.

Connect the primary of the unknown transformer to the box and connect the AC voltmeter to the

Resource

C. Hansen, "Electrolytic Capacitor Reforming," *Glass Audio*, June 1997.

primary side also. Turn on the variac and adjust the AC until you read 5 VAC on the primary. Now, disconnect your meter from the primary and connect it to the secondary. Let's assume that you measure 0.2 VAC on the secondary. The 5 V is divided by the 0.2 V, which equals 25 (primary volts/secondary volts = turns ratio). Take this number, square it, and then multiply by the impedance of your secondary (4 Ω, 8 Ω, or 16 Ω, depending on what secondary load you want).

Turns ratio × Turns ratio = Impedance ratio. Multiply this by your selected secondary load. For example, 25 × 25 = 625 × 8 Ω = 5,000 Ω (5 kΩ:8 Ω).

If you do not have a variac, the procedure is the same but your numerators will differ. Assume that when you connect the filament transformer to the primary of the unknown transformer, you just have the filament transformer plugged into the wall. Let's say the voltage reads 6.85 V and the secondary reads 0.27 V. (6.85 V divided by 0.27 V = 25.4 turns ratio or 25.4 × 25.4 = 645 and 645 × 8 Ω equals 5.16 kΩ or probably 5 kΩ:8 Ω. If the secondary only has one set of two leads and you are driving a 4 Ω load then the primary impedance is 2,500 Ω.

You can use this same procedure for matching transformers or if you want to experiment with using filament transformers as output transformers (not ideal for hi-fi, but it works with guitar amps and tube radios).

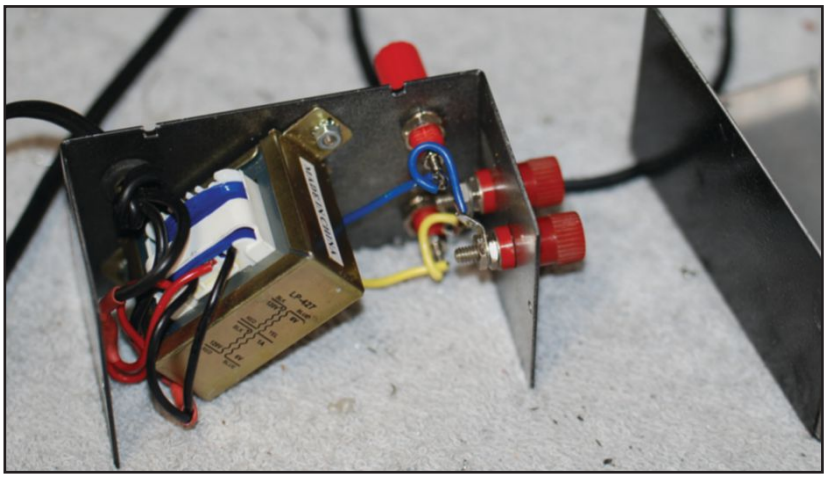


Photo 8: This simple circuit and a four-function calculator will enable you to determine the primary impedance of any transformer.

The Tools

I probably use at least one of these tools each week, and I hope you will find them useful. I can be reached at Tuninfork@aol.com.





Inrush Current Protection for Tube Amps



Learn how to balance the use of transformers, rectifiers, and filter capacitors to increase the lifespan of components, reduce mechanical and thermal stress, and protect older tubes.

By
Scott Dorsey
(United States)

Tube amplifiers with solid-state rectifiers are hard on input transformers, rectifiers, and filter capacitors. When the power is first turned on and the power supply capacitors are completely discharged, they present almost a complete short to the transformer. The transformer passes as much current into them as it is capable of doing, often resulting in a characteristic "Ba-Whangg" sound as the flux in the core rapidly changes. Magnetostriction in the core materials causes the core to flex and laminations to be spread apart, resulting in that sound and eventually weakening the core.

There is also considerable stress on the capacitors themselves, especially older capacitor designs that have enough series resistance that the inrush current can briefly result in considerable heating internally until the capacitor charges.

And, don't forget the rectifiers. Admittedly, modern replacement rectifiers are inexpensive and replacing them is considerably easier than replacing a transformer or can capacitor. Still, constant shock to the rectifier can result in premature failure, and when things fail, it is always at the most inopportune time.

Reducing Mechanical and Thermal Stress

The key to increasing the lifespan of components is to reduce mechanical and thermal stress on them. This is especially important with older tube power amplifiers where the transformers may be custom and the capacitors may be difficult to readily replace.

Replacing those capacitors with modern types, especially with higher valued ones, increases the inrush current demand and puts further stress on the power transformer. Even if you don't put a higher value capacitor in place, the reduced equivalent series resistance (ESR) of modern capacitors is going to greatly increase the inrush current demands and the stress on the power transformer.

So, what can be done? The first possibility is to use a tube rectifier that not only has considerable series resistance, but also starts out cold with infinite resistance and slowly warms up to pass more current. This is a fine solution to the inrush current problem, but the series resistance permanently limits the ability to drive current. The resistance of the tube rectifier combined with the capacitance of the filter capacitor forms a low-pass network that limits the amplifier's ability to drive current at low frequencies. Of course, you could add still more capacitance to reduce the B+ voltage sagging at low frequencies, but then you've just increased the starting current demand on the tube and caused entirely new problems.

The second possibility is to ditch the power transformer and use an "impedance protected" transformer that is inherently current-limited and turns into a constant current source (CCS) when a short is applied to it on startup. This is expensive and requires considerable modifications to vintage

equipment. If your primary goal is to protect a vintage power transformer, then the last thing you want to do is replace it.

Thermistors

The technique of using thermistors as inrush current protection devices has been around for many years, and you will find them inside most computer power supplies. Switching supplies in your computer have the rectifier and main filter capacitor directly connected to the AC power line with no transformer to limit inrush current. To prevent them from blowing the fuse when first turned on, a series thermistor is a normal addition to any computer supply.

The typical application uses an NTC thermistor, which is a device that has a negative temperature coefficient (NTC). Unlike a conventional resistor or light bulb, when it gets hotter, the resistance drops. These devices are normally specified by cold resistance and maximum operating current. So you might find a device sold as a "1 Ω, 20 A NTC thermistor," meaning that it has 1 Ω cold, and after it warms up, it's able to deliver as much as 20 A. The resistance drops down to nearly zero once it comes up to temperature.

The thermistor is a sort of trick semiconductor and it operates by the same principle as the thermal runaway effect that destroys output transistors on power amplifiers. It's a block of semi-conducting material and as the temperature is increased, the number of charge carriers available to carry electrons across the device increases. This is the same mechanism by which the increased temperature of a transistor increases its beta (which in an amplifier output stage means more current and still increased temperature in an endless loop until failure). Unlike many other semiconductors, though, they are symmetric and can be used on an AC power line.

So, say we have a Harman-Kardon Citation II amplifier, which has a steady-state current while running at about 3 A, but pulls nearly 20 A for a fraction of a second when first turned on. And, we want to reduce the transformer's "Ba-Whangg" and prevent damage. We know we need a thermistor rated for at least 3 A, and we might want to derate that by a factor of two, so let's say at least 6 A. If we pick something with a cold resistance of 10 A, that will limit our momentary load to 12 A from a 120 V circuit, or even less given the resistive losses in the transformer.

Pulling out the Ametherm catalog and looking for something with a 10 Ω cold resistance and a 6 A running current gives us the Ametherm SG27, which is $1.33 at Mouser. That's inexpensive protection for a transformer that would cost hundreds of dollars to rewind if it failed.

About the Author

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

This same math can be applied to any device with any load characteristic. Just figure out the maximum current it should be allowed to pull on startup and the maximum current it normally pulls in continuous operation. After you have done the math, install the device and see how it works.

Measure the voltage drop across the thermistor and if it is more than a couple volts during normal operation, go to a higher current thermistor. If it continues to show signs of heavy transients on startup (making sounds or shaking for instance), consider a higher resistance thermistor. However, you'll find the simple math should get you very close to the right thermistor the first time.

Mount the thermistor close to the power input, in series with the transformer, and if necessary, install standoffs so that the thermistor is well supported by both legs and is sitting in free air. The thermistor does get hot in normal operation, so you want it away from wiring and components by at least a quarter inch. Don't put heatshrink tubing on the thermistor body. You will sometimes see that done on PC power supplies to speed up the heating of the device, but you don't need that for a tube amplifier.

Several vendors make NTC thermistors that are specifically designed for inrush current limiting and, therefore, are rated in terms of cold resistance and maximum operating current. My inclination is to buy from Ametherm because its applications engineers and my local representative are helpful and friendly, but a number of other vendors such as AVX, Cantherm, EPCOS, U.S. Sensor, Murata, and Honeywell make these products. There are also several small Chinese companies making inrush current protection thermistors, which can be found on online. However, I can't speak for the quality of products purchased from random Internet vendors.

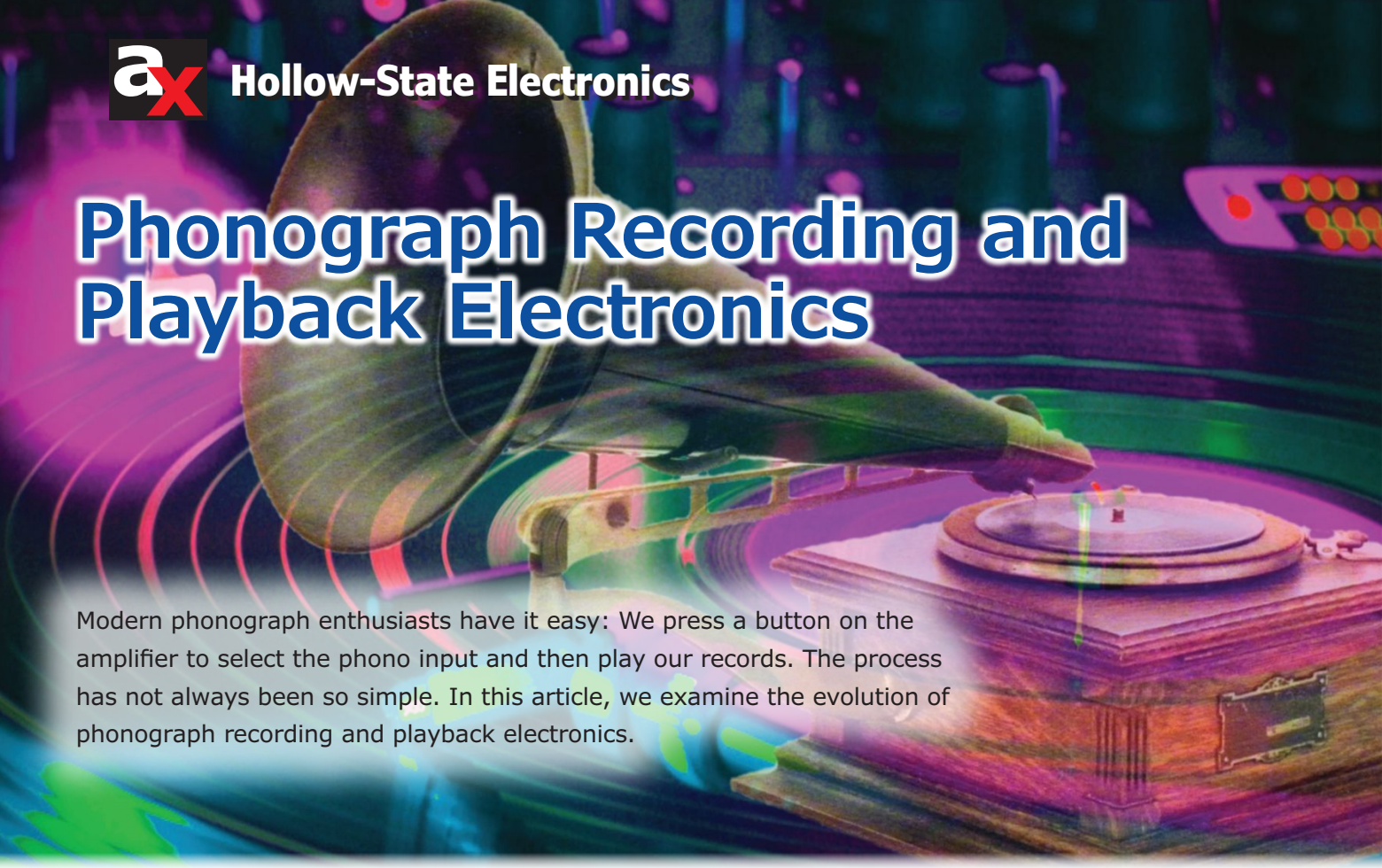
Protection Equals Peace of Mind

Adding thermistors is a simple retrofit that can be used on all tube amplifiers and might be worth adding to some solid-state amplifiers that have large reserve capacitance and high inrush current. It will extend the life of the transformers and the capacitors with no real disadvantage. It is also worth considering inrush current limiting when designing new amplifiers, especially with high efficiency transformers that might have high peak currents.

Source

SG27 Thermistor
Ametherm |
www.ametherm.com

Phonograph Recording and Playback Electronics



Modern phonograph enthusiasts have it easy: We press a button on the amplifier to select the phono input and then play our records. The process has not always been so simple. In this article, we examine the evolution of phonograph recording and playback electronics.

By
Richard Honeycutt
(United States)

Since the early days of hollow-state phonograph disc recording and playback electronics, many changes have taken place that affect the design and operation of recording and playback electronics. Western Electric introduced its first electromagnetic disc cutter in the early 1920s, and in 1924, the Columbia Phonograph Co. began producing Orthophonic records using this process. In 1925, RCA Victor released its last discs made using the acoustic process, and in 1930, RCA Victor began selling its first electrical phonograph record player.

Early Phonographs

The original Edison cylinders were cut using a vertical or "hill-and-dale" motion, but the disc record invented by Emile Berliner in 1887 used a lateral (horizontal) stylus motion, which reduces sensitivity to turntable rumble. Although vertical recording continued for some years, lateral recording was preferred by consumers, and soon was used for the majority of commercial records.

Figure 1 shows the schematic of a Westrex cutterhead amplifier employing four parallel/push-pull 807 triodes. The circuit is similar to that of a common audio power amplifier, except that it has a provision for a feedback signal from the cutterhead.

Especially when cutting a record using lateral modulation, the cutterhead creates a wider groove at low frequencies, due to its constant-velocity characteristic. (Since higher frequencies produce more rapid undulation of the groove, if the stylus motion has constant velocity, its excursion must decrease as frequency increases.) If the groove becomes too wide, the maximum time length of music on a disc is reduced, and the playback stylus may not be able to track the groove well.

To correct for this problem, the audio signal sent to the cutterhead was reduced below a "bass turnover point"—typically 250 Hz. For flat playback, a complementary bass boost had to be provided in the playback electronics. This bass boost on playback had the unfortunate result of increasing the effects of turntable rumble and record-warp noise, especially at sub-audible frequencies, which in turn created distortion. To prevent this problem, a bass shelf was later added to the playback curve, limiting the bass boost below a frequency of 30 to 50 Hz.

Early electrical recordings had an upper frequency

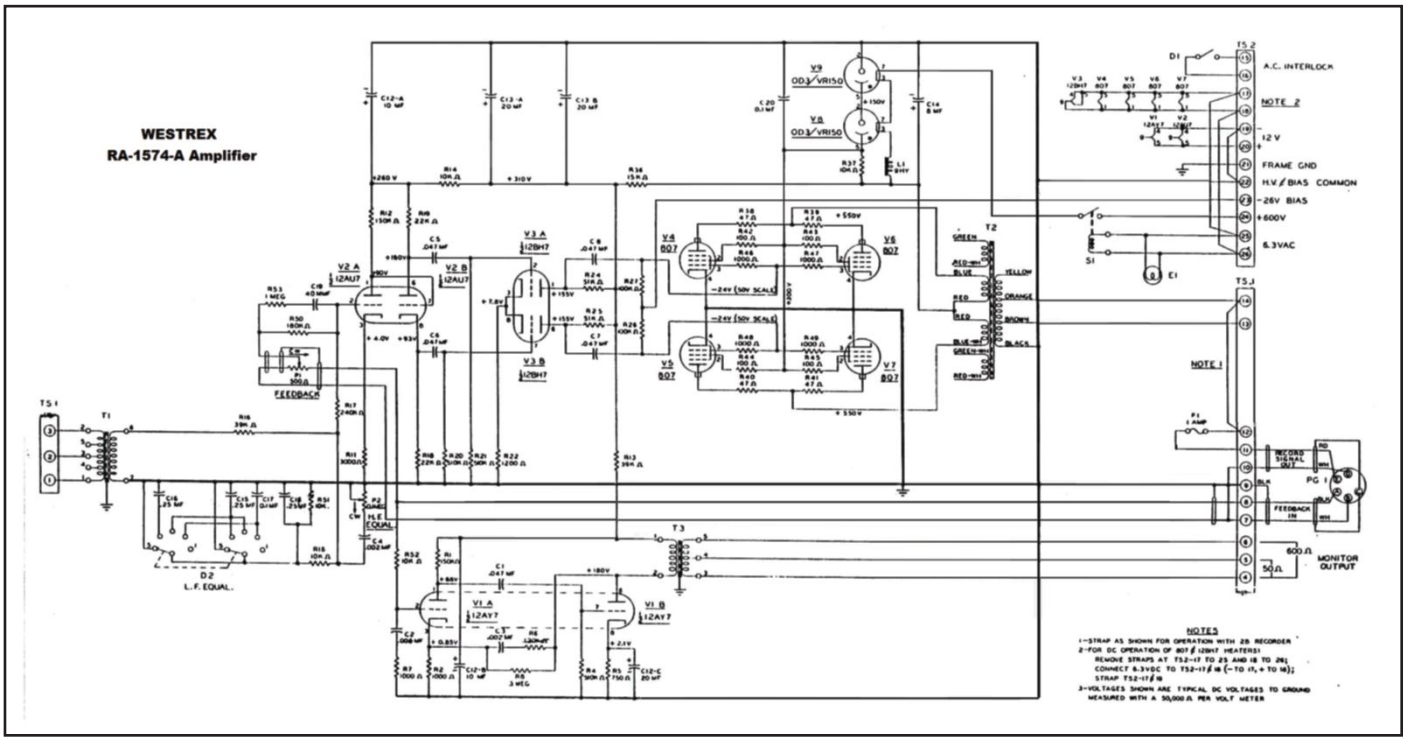


Figure 1: The Westrex RA-1574-A cutterhead amplifier was widely used to cut mono master records.

limit of about 5 to 7 kHz, but even at those frequencies, the combination of surface noise and small groove excursion due to constant velocity recording made the upper frequencies difficult to hear. Adding a high-frequency boost in the recording stage provided effectively constant amplitude recording of the treble, which increased the signal level above the "treble turnover point." A corresponding treble cut in the playback electronics restored the recording-reproduction chain to nominally flat, simultaneously reducing the audibility of surface noise.

There was no standardization of the actual bass or treble turnover points or the slopes of boost and cut. Bass turnover points ranged from 250 to 1,000 Hz, and treble turnover points (where treble boost was used, not including a 1.5 dB peak at 6.6 kHz caused by cutterhead resonance) to 2.5 kHz. Often instead of specifying a treble turnover point, the number of decibels cut at a certain frequency, such as 10 kHz, was given.

Recording Characteristics

The recording equalization that was used varied from one record manufacturer to another, and among sessions for a given manufacturer—even from one side of a record to the other. In 1949, to address this confusion, which particularly affected radio broadcasters using records as source material

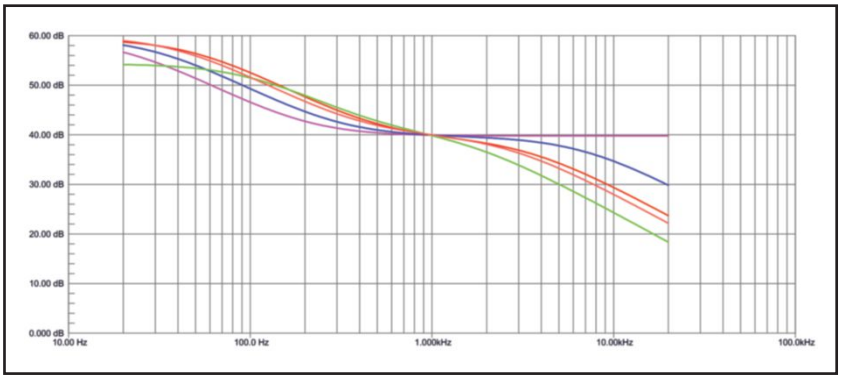


Figure 2: These are some of the equalization curves needed for accurate playback of late 78-rpm and early 45- and 33 1/3-rpm records. See Table 1 for the key. (Graph courtesy of Gary Galo)

Color	Phonograph Type	EQ Curve
Magenta	Early Electrical 78	20 Hz/200 Hz/Flat treble
Yellow	c. 1930 Electrical 78	30 Hz/300 Hz/-5 dB at 10 kHz
Light Cyan	Late Electrical 78	50 Hz/500 Hz/-10dB at 10 kHz
Dark Cyan	AES LP	40 Hz/400 Hz/-12 dB at 10 kHz
Green	Columbia LP	100 Hz/500 Hz/-16 dB at 10 kHz

Table 1: The color key decodes the EQ curves shown in Figure 1.



Photo 1: The McIntosh hollow-state model C-8 preamp included a bank of switches to provide selectable record equalization.

("canned music," its detractors called it), the National Association of Broadcasters (NAB—later called the National Association of Radio and Television Broadcasters or NARTB) issued standards for both vertically and laterally cut records. The NAB curve for laterally cut records was quite similar to the Orthoacoustic curve developed by NBC in the 1930s. The NAB curve employed additional bass boost below 100 Hz, and a treble turnover point of 1.6 kHz, the high-frequency boost at 10 kHz being 16 dB.

In 1951, the Audio Engineering Society (AES) developed a standard playback curve intended for use by hi-fi amplifier manufacturers. The bass and treble turnover frequencies were 400 Hz and 2.6 kHz, respectively. Some of the latest 78s and many early LPs were recorded using the AES curve.

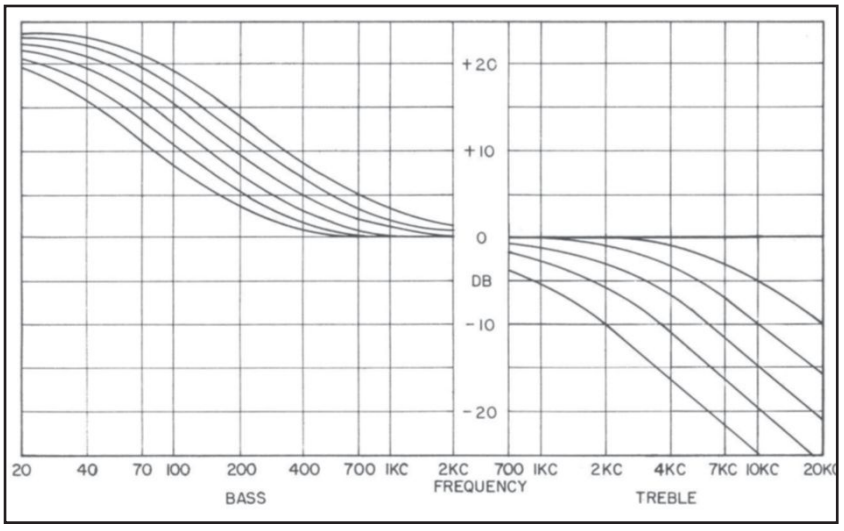


Figure 3: The McIntosh C-8 could provide all these EQ curves in order to help a listener optimize the equalization for a particular record.

In 1949, Columbia debuted the 33 1/3 rpm LP. Also in 1949, RCA introduced the 45-rpm record. The battle between RCA Victor's 45-rpm record and Columbia's 12" LP also involved a difference in the recording characteristics. RCA Victor was using the "New Orthophonic" curve, with a 2.1 kHz treble turnover, whereas Columbia was using the LP curve. Between 1953 and 1956, several standards organizations around the world adopted the RCA Victor New Orthophonic curve as the standard throughout the national and international record markets. The curve was adopted by the Recording Industries Association of America (RIAA) soon after its founding in 1952.

Figure 2 illustrates the variety of playback equalization curves needed for proper reproduction of records in the early 45-33 1/3-rpm era. **Table 1** shows the color key to the graphed curves.

Magnetic Cartridges

Note that the reasons for record and playback equalization of records has nothing to do with the playback cartridge. A magnetic cartridge has a flat frequency response when playing back a recording created with a constant velocity process.

Magnetic cartridges work by translating groove undulations into a variation of magnetic field strength in the vicinity of a coil. This can be accomplished by keeping the coil fixed and attaching a tiny magnet to the stylus shank (i.e., the moving magnet or MM design), or by keeping the magnet stationary and letting the stylus shank move the coil (i.e., the moving coil or MC design). MM cartridges are more commonly used, but MC designs are considered superior by a small but vocal group of audio enthusiasts. The debate over the superiority of MM or MC designs has no clear winner. The discussion is often muddled by cost and design considerations (e.g., Can an MC cartridge requiring another step-up amplification outperform well-made MM cartridges that need simpler front-end stages?).

Several considerations apply:

- MC cartridges offer very low inductance and impedance, which means that the effects of

Resources

"1931 Recordings of Leopold Stokowski and the Philadelphia Orchestra," The Stokowski Legacy, www.stokowski.org/1931_Electrical_Recordings_Stokowski.htm.

G. Galo, "Archival Phonograph Preamplifier," *Linear Audio, Volume 5,* https://linearaudio.net/volumes/786.

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Photo 2: The Altec Model 440B preamp also provided selectable phono equalization.

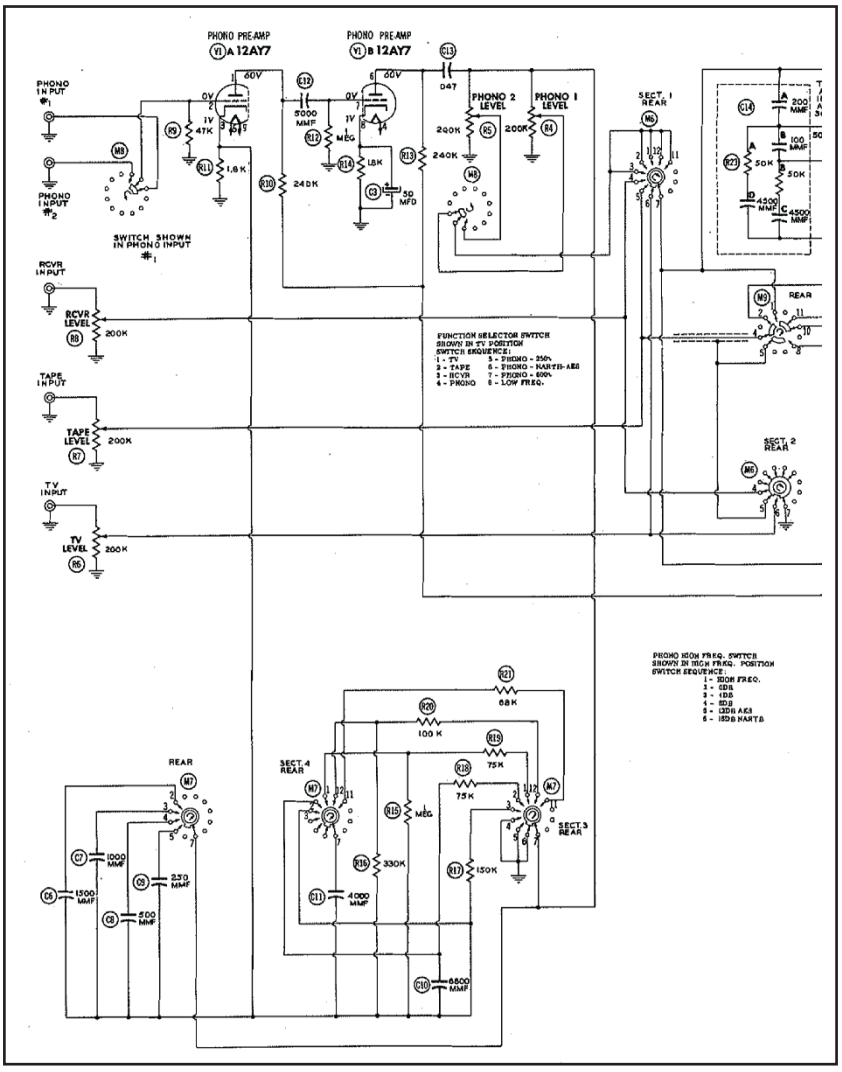


Figure 4: This partial schematic of the Altec 440B preamp shows the phono equalization circuitry.

cable capacitance are unimportant; unlike MM cartridges, whose inductance and impedance are much higher. This higher impedance makes MM cartridges more susceptible to high-frequency attenuation in the cable.

- It is often believed that MC cartridges have lower moving masses, making them capable of better high frequency and transient responses and reduced record wear. However, some of the best MM cartridges have lower moving masses than some MC designs.
- Other design factors—including the stylus tip shape, mounting (bonded vs. nude), cantilever material, and cantilever design—have a significant influence on the sound, and this may account for more variation of sound quality than the MM vs. MC design.

It is true, however, that MC cartridges require higher preamp gain than MM cartridges.

From the late 1930s into the 1950s, piezoelectric crystal and ceramic cartridges outsold magnetic cartridges, primarily because of their lower cost. However, "crystal" cartridges (as both types were usually called) did have two advantages: Their higher output required less gain in the preamp, and they did not require bass cut in the playback electronics. (However, contrary to common opinion, crystal cartridges did not provide the treble cut necessary to achieve a flat response. A few record players and hi-fi/stereo preamps and amplifiers did include the correct treble cut for a crystal cartridge connected to the appropriate input, and using the correct selector switch settings.) However, most devices using crystal cartridges used speakers whose high-frequency response was weak enough to prevent the lack of treble equalization from making the sound too strident.

Hi-Fi Recordings

Arguably, the first high-fidelity recordings were made by Bell Labs in 1931 and 1932, which you can hear on The Stokowski Legacy website (www.stokowski.org/1931_Electrical_Recordings_Stokowski.htm). However, the term "high fidelity" did not make its way into audio marketing until the 1950s. By 1955, an avid listener to phonograph records might well have a few pre-1925 acoustic-process records, many 78-rpm records of various vintages and manufacturers, a growing collection of LPs, and even a few RCA 45-rpm classical or popular records.

If the audio enthusiast had a limited budget, the playback system would include a record player with a crystal cartridge and probably no playback equalization. But those who were able to invest in

better systems had several options. One of these systems included the McIntosh C-8 preamplifier (see **Photo 1**). By using the Record Compensator switches to customize EQ settings, the listener could approximate the optimum playback EQ for most records. Even if the actual EQ used in recording was not known (which was usually the case), adjustments could be made by ear so that the reproduced sound was reasonably natural. **Figure 3** shows the various response curves the C-8 could provide.

Photo 2 shows the Altec Lansing 440B preamp, a somewhat simpler although less versatile unit, that provided multiple equalization options. From the partial schematic provided in **Figure 4**, you can see the function switch includes settings for PHONO, PHONO 250~, PHONO NARTB-AES, PHONO 500~, and LOW FREQ. The record compensation switch has settings for HIGH FREQ, 0 dB, 4 dB, 8 dB, 12 dB AES, and 18 dB NARTB. The meanings of these designations are described in the Altec Lansing 440B preamp's user manual and summarized on the printed guides affixed to the inside of its hinged front cover.

An example of newer phonograph preamps featuring selectable equalization is the Sentec EQ10 (see **Photo 3**). The more simple, but less versatile EQ11 (not pictured) is also available. For more information, DIYers might be interested in Gary Galo's article "Archival Phonograph Preamplifier," which appeared in *Linear Audio. Volume 5* (https://linearaudio.net/volumes/786).



Photo 3: The Sentec EQ10 is a versatile hollow-state phono preamp that provides great flexibility in choosing equalization.

Author's Note: I would like to thank Gary Galo for his invaluable assistance in researching information for this article.

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Industry Calendar

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

May 18–21, 2017

High End 2017

MOC Munich,
Munich Germany
www.highendsociety.de

Let The Music Play! The new theme for High End 2017 reflects the idea of addressing people and conveying a positive, emotional experience with high-quality music reproduction. The High End show is the world's foremost meeting place for the high-end audio industry, including emerging consumer audio technologies and personal audio. The High End is indisputably the number one international meeting point for audio enthusiasts and has been an institution for many years. The High End show is organized by the High End Society, the trade association for high-quality electronic home entertainment. This year will be the 36th edition. On May 18, the show is only open to registered trade visitors and journalists with prior accreditation.

In 2016, more than 500 exhibitors (most of them representing multiple brands), from over 40 countries, presented their products and services, while more than 20,000 visitors, of which 7,000 are high-profile trade visitors from all corners of the world attended. In February 2017, the show was already completely booked, while the demand from companies for exhibition opportunities was still high. The High End show displays the entire spectrum of the audio market, from small to exclusive music systems, car sound and headphones, right to record players and traditional valve amplifiers, using all four halls and 140 adjacent demonstration booths in the MOC complex.

May 20–22, 2017

142nd AES Convention Berlin 2017



Conference: May 20–May 23,
Exhibits: May 20–May 22,
Maritim Hotel Berlin,
Berlin, Germany
www.aes.org/events/142/
aeseurope.com

Following 2016's two highly-successful Audio Engineering Society (AES) conventions in Paris and Los Angeles, the AES Berlin convention will be the association's largest European pro audio event of the year, drawing professionals and students from around the world. The exhibition promises to be a decent display. The Tech Tours program will include several renowned Berlin recording studios, tours of market-leading Berlin-based audio manufacturers, broadcast and film sound production facilities and the Museum für Kommunikation (Museum of Communications).

With one of the largest numbers of papers and engineering briefs accepted of any recent AES convention, the work of more than 120 researchers will be showcased. Topics range from 3D Recording and Reproduction and IEMs and Hearing Protection to Bluetooth and Digital Filter Design. Workshops bring panels of experts together to discuss a range of provocative and practical topics. And, the tutorials will provide a varied horizon of presentations on important audio fundamentals, for young people and seasoned professionals alike, on 3D and Binaural Listening, Audio Forensics and Sound Design, to name but a few.

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