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Voice Market Update

Connected Audio Products and Voice Recognition

By J. Martins

Voice Capture Challenges

The Threshold of the Golden Age of Voice

By Mike Klasco

Voice Assistants

The Essential Next-Generation Consumer Interface

By Simon Bryant (Futuresource Consulting)

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Turning Up the Innovation on Smart Speaker Designs

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Audio Electronics

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Hollow-State Electronics

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By Richard Honeycutt

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Here's to Another Five Years!

I just returned from the 145th Audio Engineering Society (AES) Convention in New York, where I was reminded that it was there—five years ago—that we introduced the redesigned and re-energized edition of this magazine. Even though my work with the title had been initiated earlier, it was indeed a privilege to publicly showcase the first results at a forum such as the 135th AES convention in NY, in 2013, where the feedback I received was highly enthusiastic and encouraged me to pursue our vision for the new *audioXpress*.

As it was in 2013, this year we had a vibrant and well-attended AES conference, where we engaged once again in interesting discussions with manufacturers, some of the best industry pioneers and researchers, many of whom are we privileged to have as colleagues and regular contributors to *audioXpress*.

As we felt five years ago, the ideas that have guided our vision for *audioXpress* remain solid as ever, as is the need for a reliable, trustworthy, and relevant source of technical information, which only an independent and highly focused publication can provide. Also, our editorial principles have remained strong and consistent. We continue to believe in the need for a broad perspective in the audio industry, providing an expanded scope that is immune to the application silos and vertical markets, even with all the challenges that encompasses. We remain determined in our expressed mission to be the reference information source about innovative solutions and technologies in audio electronics, acoustic and electro-acoustic design for DIY audio enthusiasts, audio product designers, electronic and audio engineers, the industry, and its channel partners (agents, consultants, integrators, and opinion leaders).

As the 145th AES convention served to reinforce, audio remains one of the most exciting and innovative disciplines, as consumer-focused innovations continue to evolve in tandem with the exciting progress made in broadcasting, professional audio, music, and the installation/systems integration markets.

I am also confident to claim that we have strongly consolidated the publication's audience around its roots and tradition, while evolving the title as a platform for a wider global audience, in the pages of the magazine, the website, social media, and our weekly e-newsletter *The Audio Voice*. A diverse platform where we will continue to focus on innovation and offer readers content and an objective perspective they can't find elsewhere.

Over the last five years, we have seen a recognition of the fundamental values of great-quality analog audio, at par with a better understanding of digital, with all the flexibility and convenience of digital hardware and software. Over the last five years, the industry has achieved great progress in the implementation of audio-over-IP and the realization of the transition to networked audio and media infrastructures. We have witnessed a remarkable effort to expand on the possibilities of traditional audio formats, while paving the way to immersive experiences and object-based solutions, from creation to reproduction in large scale sound reinforcement and on the consumer edge.

On a very positive note, the music industry globally is recovering and showing signs of a new era of growth with the consumer's transition to streaming services, which still needs to evolve in quality, but have created a strong deterrent to the destructive practice of commercial-scale piracy, and brought a much needed update to copyright, commercial, and legal practices—paving the way to a fairer avenue to reward musicians and authors.

Since 2013, it has been truly remarkable to witness and document the evolution in all these areas. I believe there remains much work to do in wireless audio and the industry needs to push the convergence of modern media interfaces and networked data protocols from wired to wireless. But in no other area have we seen a greater degree of innovation over the last five years as in voice recognition applications and connected home and mobile devices. And that's precisely the topic of the Market Update featured in this month's edition. In just five years, voice applications have been the engine for the strongest pace of innovation the audio industry has ever seen, fostering the greatest evolution in all fundamental aspects of technology, from microphones and speakers to digital signal processing and audio amplification. We will continue to detail that all-important evolution in 2019.

J. Martins
Editor-in-Chief

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COLUMNISTS

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

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.

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.

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

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An Update Using Modern JFETs

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In 2013, Dimitri Danyuk wrote an article that presented apples-to-apples test results for noise densities for several low-noise discrete, SMT n-JFETs. New JFETs have appeared on the market in the last five years, so he decided to extend his design note from 2013. The tests used the same operating conditions so that designers can make a fair comparison for their applications.

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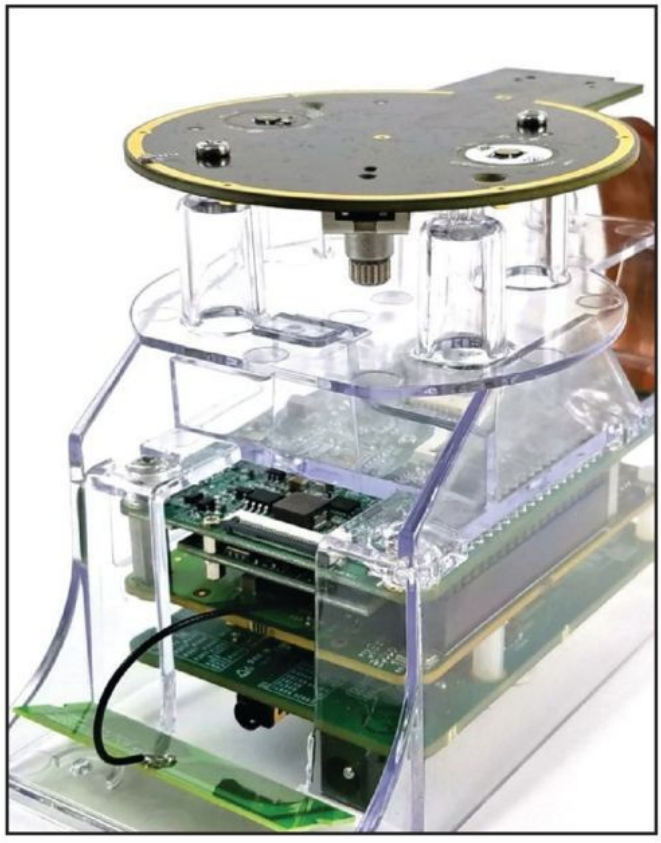

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Soviet Germanium Transistors

This article encourages DIYers to make use of the large quantities of these transistors now available on the surplus market as replacement parts or for building projects from old design handbooks.

By
Scott Dorsey
(United States)

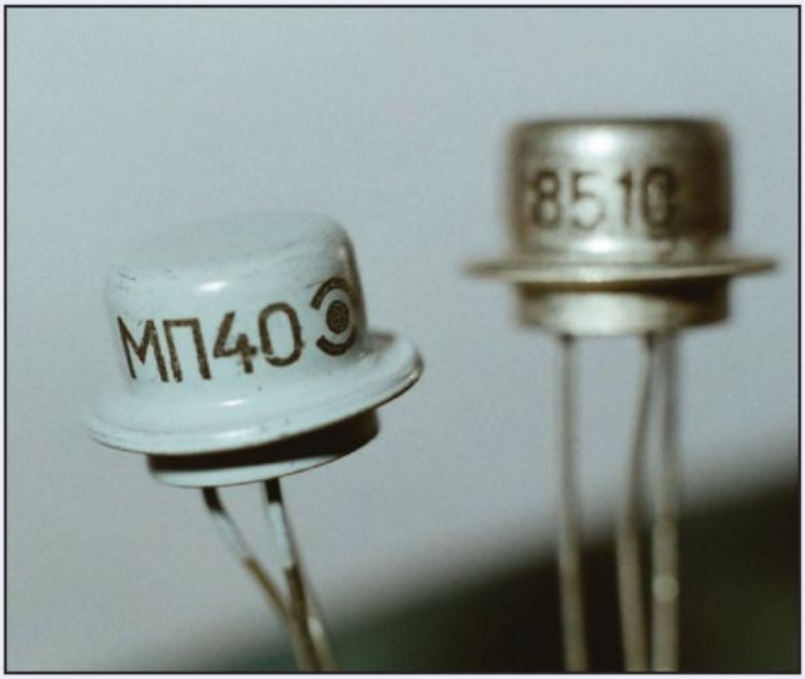


Photo 1: The majority of components in this series on the surplus market display the backward C with-Dot logo from the Novgorod Semiconductor Factory.

Before there were silicon transistors, there were germanium transistors. Germanium has a lower bandgap voltage so germanium transistors turn on at lower voltages than silicon, but they have characteristics that change considerably with temperature, so for the most part they were made obsolete rather quickly in the West.

However, because the Soviet Union was trying to build up industry they were very reluctant to close down any factory making any product even remotely useful. So it appears that factories in Russia continued making standardized germanium parts until sometime after the fall of the Soviet Union.

Large quantities of these parts are now available on the surplus market and are usable as replacement parts or for building projects from old design handbooks. However, a lack of available English-language data has kept these from finding much of a market in the West.

In some ways the later production Russian transistors are of higher quality than germanium parts made in the West, since the major Western semiconductor manufacturers abandoned the process very early while the Russians continued to improve things. So although die cleanliness and process control on Russian transistors made in 1990 may have fallen far behind what you could purchase from Motorola or Toshiba in 1990, they are still far better than that of the last Motorola or Toshiba germanium parts in the 1960s. So these are like having 1950s designs manufactured with 1990s technology.

This article is specifically about one series of transistors, which are especially common—the MP39, the MP40, and the MP41 series (see **Photo 1**). These transistors are "Alloy transistors." This was a process, short-lived in the West, which briefly predated the epitaxial transistor manufacturing process used today. A small sheet of N-type germanium has two small pellets of P-type silicon fused onto either side of it, forming a transistor junction. Because the N and P components were originally separate pieces of metal rather than one monolithic piece, there is some discontinuity at the joint, which results in added resistance and noise.

The alloy process was developed by General Electric in 1951, and by 1960, it was already obsolete and being superseded with the planar process. Typical American-made alloy transistors of the 1950s included the 2N43/44/45 series, the 2N107, the 2N135, the 2N393, and the 2N396. Some original

literature described these as "diffused-junction" transistors but this should not be confused with the planar diffusion process.

Russian companies began manufacturing copies of the Western Electric point contact transistors in 1954. In 1955, they introduced the P1 alloy transistor, and by the 1960s, various Russian factories were making an entire series of transistors with P-numbers such as P16 and P309.

Around 1960 or so, a new transistor numbering was introduced for improved quality components. The MP designation for newer transistors stands for "Modernized P" and they were used for "improved replacements" for the P series devices. So the original P40 transistor got replaced with the more compact and better-sealed MP40. In many ways, some attributes from these have been copied from the West—the case design is a Motorola one although they replaced the expensive kovar alloy with a softer copper alloy. And, the manufacturing appears to be copied from the GE components (see **Photo 2**).

These became the standard commodity transistors in the Soviet Union, and because of centralized planning they did not have the sort of second-source arrangements or the variety of similar components that we had in the West. So the MP39/40/41 series produced by the Svetlana factory in Leningrad and the Novgorod Semiconductor Device Factory became common whenever low-frequency gain was needed (see **Photo 3**). And, as mentioned earlier, the difficulty of keeping up with demand for any components meant that these stayed in production until shortly after the fall of the Soviet Union.

Availability of Germanium Transistors

There are a large number of Internet suppliers from the former Soviet Union and former Soviet Bloc countries selling these transistors. I find that if I purchase a large bag of one type and look at the distribution of alpha that there are a couple holes in the distribution, as if what is being sold online are leftovers from transistors being selected for specific applications. That's fine! They are still perfectly good. There are a few US components importers that are selling new-old stock (NOS) Russian transistors as well, and a Google search will find some of them.

The majority of components in this series on the surplus market display the backward C with-Dot logo from the Novgorod Semiconductor Factory as shown in Photo 1. That factory is still in operation today, having become Planeta Semiconductor after the close of the Soviet era and it was continuing to produce Soviet designs for some time.

The Datasheet

An English translation of the Russian datasheet shown in **Photo 4** can be seen in **Table 1**, and it may look a bit unfamiliar, both because there are some oddities in Russian measurements and because there are inherent properties of germanium transistors, which are assumed by the datasheet author and may not be obvious to people who grew up with modern silicon transistors. First, you'll notice that most of the parameters vary a lot with temperature, from the maximum operating voltages to the beta. This is an important key to designing with these devices—put them on the curve tracer and when you put your

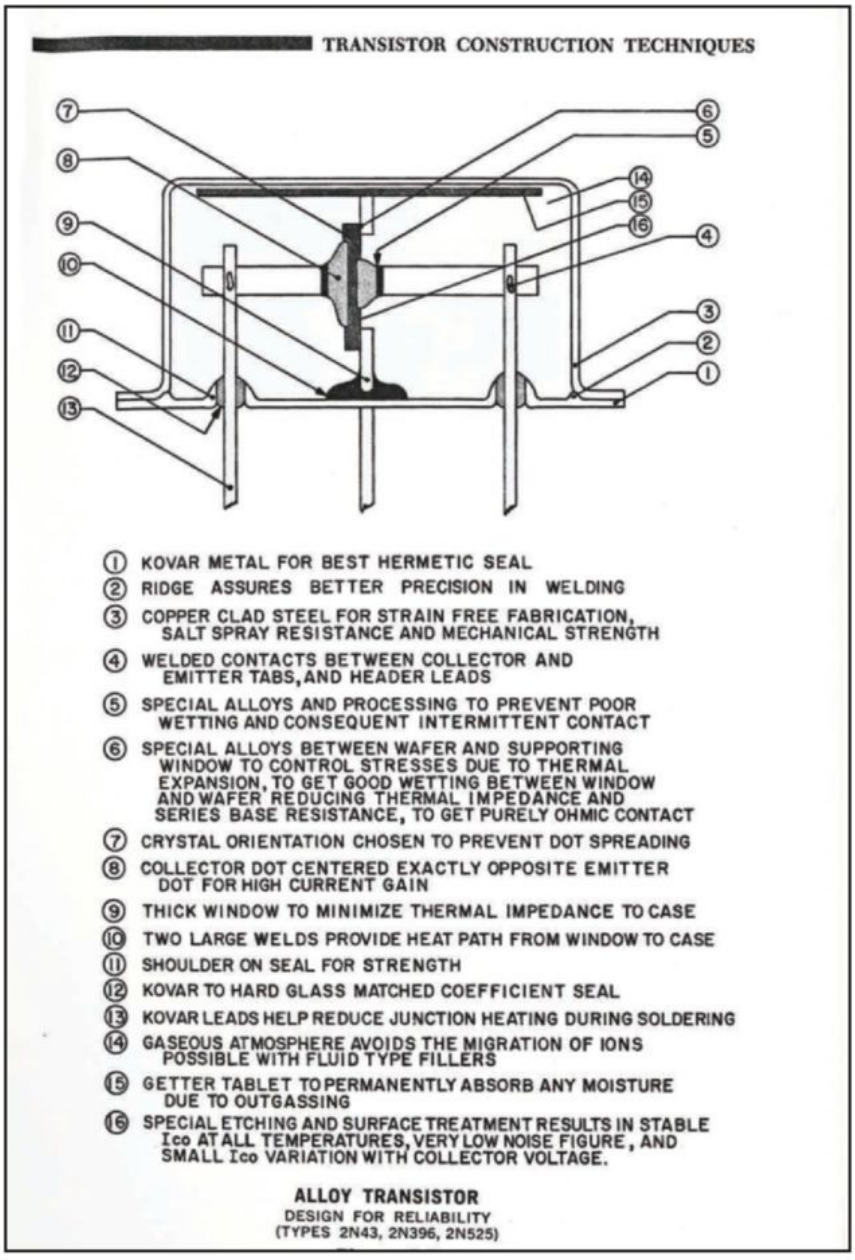


Photo 2: This information came from the General Electric databook.

finger on them and warm them up a degree or two the curves change. All the operating parameters are dramatically affected by temperature, and as you run them, they warm up. It'll take feedback to compensate for the parameter changes.

Beta also varies a lot with collector current, more so than with silicon transistors. If you want high gain, you need to operate them with higher current. The Ft is very low at 0.5 to 1 MHz at 1 mA, though... so bandwidth is limited and that makes feedback tricky! That 60 nF of collector capacitance explains why it is so limited.

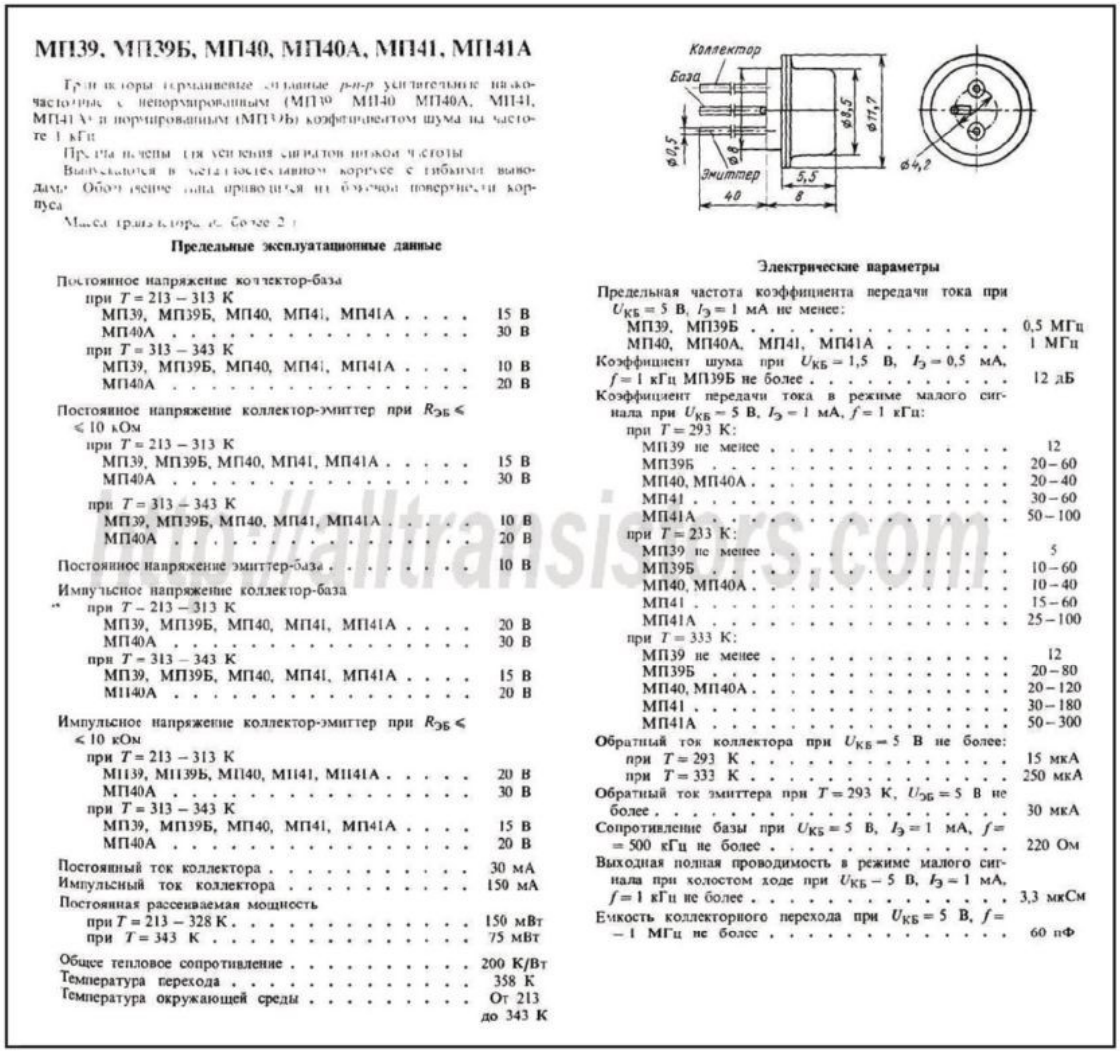
You'll also notice power dissipation isn't as good as you'd expect from a TO-39 package, indicating there is thermal resistance between the die and the package. You'll notice there is a noise coefficient listed... but no reference point! I am not sure that this number is in any way useful except with reference to other Russian transistors.

The total output conductivity is really quite low, meaning the "on-resistance" of the transistor is comparatively high. But most interesting is that low-base resistance—there is a lot of internal leakage in this transistor.

I've tried to translate the datasheet with conventional Western abbreviations for most of the values in order to make this clear, but people who are used to working with modern silicon transistors will find these to be inconsistent with temperature, high in base leakage, and comparatively low gain. But that is just how it was in the days when the transistor was new.

If you actually measure the curves (see **Figure 1**), you will find that some don't match the datasheet but in fact have loops in them! The leading edge and trailing edge of the waveforms are different. It's possible this is just because the capacitive compensation in my curve tracer won't go far enough to compensate for such massive leakage capacitance, but it may also be a function of surface contamination on the transistor.

Back in the 1960s, you could call Sprague Semiconductor and order high-speed switching germanium transistors for computer circuits. They



МП39, МП39Б, МП40, МП40А, МП41, МП41А

Три вида германиевых сплавных *p-n-p* усилительных низкочастотных с непормированным (МП39, МП40, МП40А, МП41, МП41А) и нормированным (МП39Б) коэффициентом шума на частоте 1 кГц.

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Постоянное напряжение коллектор-база	
при $T = 213 - 313$ К	
МП39, МП39Б, МП40, МП41, МП41А	15 В
МП40А	30 В
при $T = 313 - 343$ К	
МП39, МП39Б, МП40, МП41, МП41А	10 В
МП40А	20 В
Постоянное напряжение коллектор-эмиттер при $R_{ЭБ} \leqslant 10$ кОм	
при $T = 213 - 313$ К	
МП39, МП39Б, МП40, МП41, МП41А	15 В
МП40А	30 В
при $T = 313 - 343$ К	
МП39, МП39Б, МП40, МП41, МП41А	10 В
МП40А	20 В
Постоянное напряжение эмиттер-база	10 В
Импульсное напряжение коллектор-база	
при $T = 213 - 313$ К	
МП39, МП39Б, МП40, МП41, МП41А	20 В
МП40А	30 В
при $T = 313 - 343$ К	
МП39, МП39Б, МП40, МП41, МП41А	15 В
МП40А	20 В
Импульсное напряжение коллектор-эмиттер при $R_{ЭБ} \leqslant 10$ кОм	
при $T = 213 - 313$ К	
МП39, МП39Б, МП40, МП41, МП41А	20 В
МП40А	30 В
при $T = 313 - 343$ К	
МП39, МП39Б, МП40, МП41, МП41А	15 В
МП40А	20 В
Постоянный ток коллектора	30 мА
Импульсный ток коллектора	150 мА
Постоянная рассеиваемая мощность	
при $T = 213 - 328$ К	150 мВт
при $T = 343$ К	75 мВт
Общее тепловое сопротивление	200 К/Вт
Температура перехода	358 К
Температура окружающей среды	От 213 до 343 К

Электрические параметры

Предельная частота коэффициента передачи тока при $U_{КБ} = 5$ В, $I_Э = 1$ мА не менее:	
МП39, МП39Б	0,5 МГц
МП40, МП40А, МП41, МП41А	1 МГц
Коэффициент шума при $U_{КБ} = 1,5$ В, $I_Э = 0,5$ мА, $f = 1$ кГц МП39Б не более	12 дБ
Коэффициент передачи тока в режиме малого сигнала при $U_{КБ} = 5$ В, $I_Э = 1$ мА, $f = 1$ кГц:	
при $T = 293$ К:	
МП39 не менее	12
МП39Б	20 – 60
МП40, МП40А	20 – 40
МП41	30 – 60
МП41А	50 – 100
при $T = 233$ К:	
МП39 не менее	5
МП39Б	10 – 60
МП40, МП40А	10 – 40
МП41	15 – 60
МП41А	25 – 100
при $T = 333$ К:	
МП39 не менее	12
МП39Б	20 – 80
МП40, МП40А	20 – 120
МП41	30 – 180
МП41А	50 – 300
Обратный ток коллектора при $U_{КБ} = 5$ В не более:	
при $T = 293$ К	15 мкА
при $T = 333$ К	250 мкА
Обратный ток эмиттера при $T = 293$ К, $U_{ЭБ} = 5$ В не более	30 мкА
Сопротивление базы при $U_{КБ} = 5$ В, $I_Э = 1$ мА, $f = 500$ кГц не более	220 Ом
Выходная полная проводимость в режиме малого сигнала при холостом ходе при $U_{КБ} = 5$ В, $I_Э = 1$ мА, $f = 1$ кГц не более	3,3 мкСм
Емкость коллекторного перехода при $U_{КБ} = 5$ В, $f = 1$ МГц не более	60 пФ

http://alltransistors.com

Photo 3: The MP39/40/41 series produced by the Svetlana factory in Leningrad and the Novgorod Semiconductor Device Factory became common whenever low-frequency gain was needed. (Information courtesy of AllTransistors)

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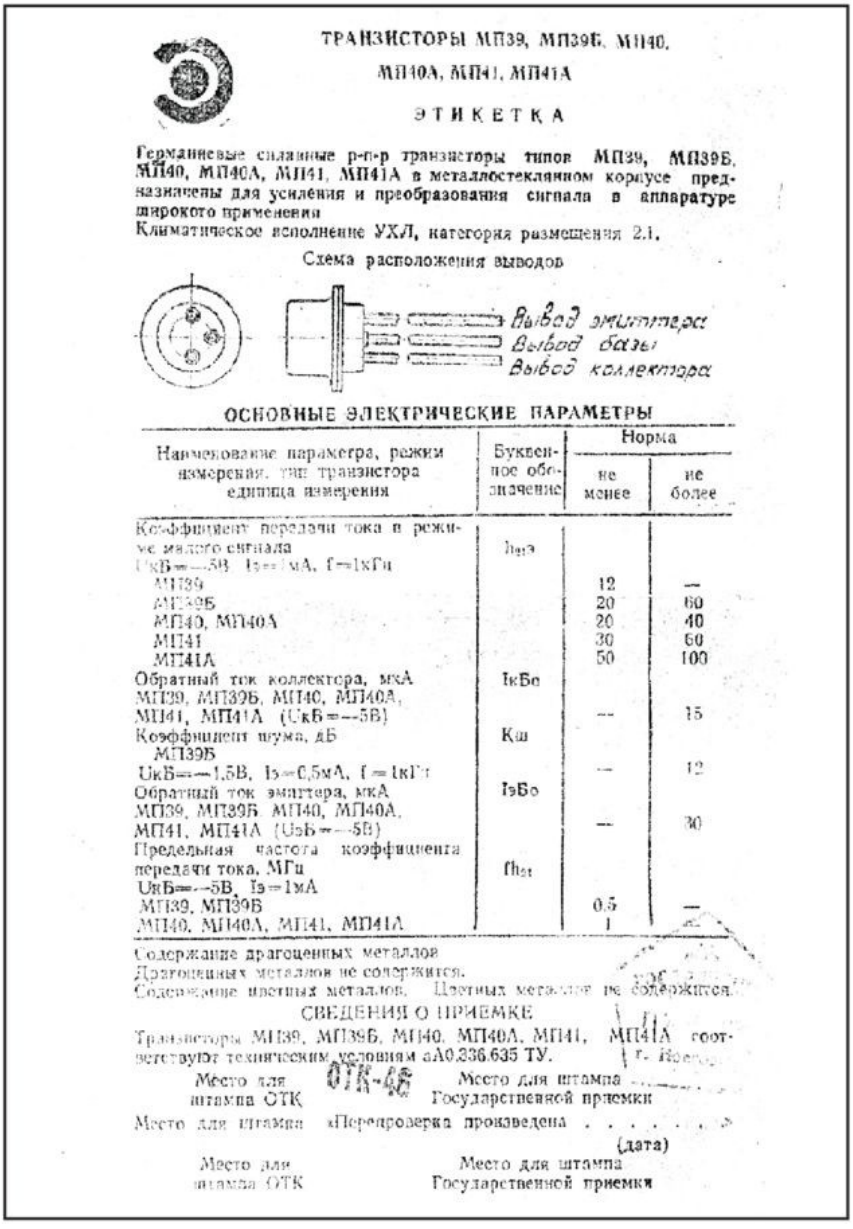
had much more abrupt transitions in the curves and the curves didn't have loops in them. They had much higher transition frequencies and they were even more expensive than regular commodity parts. The 1960s are over and you can't order that stuff any longer, and so you have to make do with what is available and the finest thing about the MP-series parts is that they are available.

Another thing you should be aware of is that the datasheet doesn't really warn you if you put these onto a modern hFE tester or onto a DVM with an hFE setting, you will get spurious values and often they will test as having much higher than they really do, as a consequence of leakage.

The curve tracer will tell the real story, and there are specially-built beta testers for these transistors. Heathkit made a common one that was popular when I was a kid and they turn up in the dollar bins at hamfests because few people today need them.



ТРАНЗИСТОРЫ МП39, МП39Б, МП40, МП40А, МП41, МП41А

ЭТИКЕТКА

Германиевые сплавные p-n-p транзисторы типов МП39, МП39Б, МП40, МП40А, МП41, МП41А в металлостеклянном корпусе предназначены для усиления и преобразования сигнала в аппаратуре широкого применения

Климатическое исполнение УХЛ, категория размещения 2.1.

Схема расположения выводов

ОСНОВНЫЕ ЭЛЕКТРИЧЕСКИЕ ПАРАМЕТРЫ

Наименование параметра, режим измерения, тип транзистора, единица измерения	Буквенное обозначение	Норма не менее	Норма не более
Коэффициент передачи тока в режиме малого сигнала	$h_{21э}$		
UкБ=–5В, Iэ=1мА, f=1кГц			
МП39		12	—
МП39Б		20	60
МП40, МП40А		20	40
МП41		30	60
МП41А		50	100
Обратный ток коллектора, мкА	$I_{кБо}$		
МП39, МП39Б, МП40, МП40А, МП41, МП41А (UкБ=–5В)		—	15
Коэффициент шума, дБ	Кш		
МП39Б			
UкБ=–1,5В, Iэ=0,5мА, f=1кГц		—	12
Обратный ток эмиттера, мкА	$I_{эБо}$		
МП39, МП39Б, МП40, МП40А, МП41, МП41А (UэБ=–5В)		—	30
Предельная частота коэффициента передачи тока, МГц	f_{h21}		
UкБ=–5В, Iэ=1мА			
МП39, МП39Б		0,5	—
МП40, МП40А, МП41, МП41А		1	—

Содержание драгоценных металлов

Драгоценных металлов не содержится.

Содержание цветных металлов. Цветных металлов не содержится.

СВЕДЕНИЯ О ПРИЕМКЕ

Транзисторы МП39, МП39Б, МП40, МП40А, МП41, МП41А соответствуют техническим условиям аА0.336.635 ТУ.

г. Новгород

Место для штампа ОТК ОТК-46 Место для штампа Государственной приемки

Место для штампа «Перепроверка произведена . . . . . »

(дата)

Место для штампа ОТК Место для штампа Государственной приемки

Photo 4: This is a datasheet from Novgorod.

Date Codes

As opposed to the date codes used by American companies, these parts are just marked with year and month. So the transistor marked "MP39BV -- 9005" was manufactured in May of 1990. I haven't managed to find any correlation between quality and manufacturing date.

Many of the transistors have a single digit stamped into the top of the case. Nobody has been able to explain this to me. It may indicate the production line used or the plant, but it definitely does not have anything to do with binning the transistor or the transistor characteristics.

Germanium Transistor Uses

There are plenty of old books with transistor projects in them from the early transistor days and many of the circuits in these will work well with the MP40 series. *The Motorola Semiconductor Circuits Manual*, *RCA Solid-State Hobby Circuits Manual*, Byron Wels' book *Transistor Circuit Guidebook*, Muneef Alwan's *Transistor Techniques*, Rufus P. Turner's *Transistor Circuits*, and the *GE Transistor Manual* provide useful information about older circuit designs and are available on Alibris and at hamfests. These transistors are also perfectly usable as substitutes for repairing older equipment and in new designs (see **Photo 5**).

As I have been testing these components for the past year or so, every time I mention that I have been working with germanium transistors to anyone they invariably ask me if they are usable for guitar pedals. I can't answer that question, because guitar pedals operate so far out of the normal operating envelope of components and they rely on specific characteristics that are not guaranteed on the datasheet and not very well characterized. So all I can say is to try them—they are certainly much cheaper than vintage American and Japanese parts and if they don't work for you, you haven't lost much.

Designing with Germanium Transistors

Designing with these transistors is just like designing with a silicon one, except that all you get are PNP types, the beta is low, and the leakage is high. You may have to fiddle with idle current in order to find the point where the noise is lowest, in an input stage or other noise-sensitive stage. You will want to rely on feedback as much as possible, using more stages instead of higher gain per stage,

in order to keep the devices uniform with changes in temperature. Note that not everyone did that in the early days of transistors because reducing the number of stages kept costs down. We live in a newer age with lower cost components and we know more about their behavior too.

Leakage Bias

One warning about germanium transistor circuits. Because of the high leakage currents inherent in the germanium transistor, a lot of designers in the early era employed "fixed bias," otherwise known as "base bias" or "boneheaded bias." In this circuit, a single fixed resistor provides the current to set the input bias and that single resistor sets the operating current (see **Figure 2**).

There are a number of disadvantages to this approach. For one thing, the operating point changes dramatically with temperature or power supply voltages. Even a tiny supply voltage change will

Translation of Russian Datasheet

Standard-type Tubular PNP Alloy Transistors MP39, MP39B, MP40, MP40A, MP41A with metallized body for amplification and conversion of signals in a wide range of equipment.

Environmental Grade XYL, Performance Category 2.1.

Maximum Usable Values
Continuous Collector-Base Voltage
at T = 231-313 kelvin (-43 to 104°F)
MP39, MP39B, MP40, MP41, MP41A 15 V
MP40A . . . . . .30 V
at T= 313-343 kelvin (104°F to 158°F)
MP39, MP39B, MP40, MP41, MP41A ... 10 V
MP40A... 20 V

Continuous Collector-Emitter Voltage at Rbe <= 10 kΩ
at T = 231-313 kelvin (-43°F to 104°F)
MP39, MP39B, MP40, MP41, MP41A 15 V
MP40A . . . . . .30 V
at T = 313-343 kelvin (104°F to 158°F)
MP39, MP39B, MP40, MP41, MP41A 10 V
MP40A . . . . . .20 V

Continuous Emitter-Base Voltage . . . .10 V

Peak Collector-Base Voltage
at T = 231-313 kelvin (-43°F to 104°F)
MP39, MP39B, MP40, MP41, MP41A 20 V
MP40A . . . . . .30 V
at T = 313-343 kelvin (104°F to 158°F)
MP39, MP39B, MP40, MP41, MP41A 15 V
MP40A . . . . . .20 V

Peak Collector-Emitter Voltage at Rbe ≥ 10 kΩ
at T = 231-313 kelvin (-43°F to 104°F)
MP39, MP39B, MP40, MP41, MP41A 20 V
MP40A . . . . . .30 V
at T = 313-343 kelvin (104°F to 158°F)
MP39, MP39B, MP40, MP41, MP41A 15 V
MP40A . . . . . .20 V

Continuous Collector Current . . . . . .30 mA
Peak Collector Current . . . . . .150 mA
Continuous Power Dissipation
at T = 231-313 kelvin (-43° to 104°F) 150 mW
at T = 313-343 kelvin (104° to 158°F) 75 mW

Similiarity Transition Resistance. . . . .200°KW (Thermal Resistance?)
Transition Temperature . . . . . .358 kelvin (195°F)
Environmental Temperature . . . . . .213 to 343 kelvin (-76°F to 158°F)

Electrical Parameters

Transition Frequency at V_{cb} = 5 V,
I_e = 1 mA maximum:
MP39, MP39B . . . . . .0.5 MHz
MP40, MP40A, MP41, MP41A. . . . . .1 MHz

Noise coefficient at V_{cb} = 1.5 V, I_e = 0.5 mA,
F = 1 kHz MP39B maximum . . . . . .12 dB

Transmission coefficient (beta) in small signal conditions at
V_{cb} = 5 V, I_e = 1 mA, F = 1 kHz:
at T = 293 kelvin (68°F)
MP39 . . . . . .not less than 12
MP39B . . . . . .20 to 60
MP40, MP40AA . . . . . .20 to 40
MP41 . . . . . .30 to 60
MP41A . . . . . .50 to 100
at T = 233 kelvin (-40°F)
MP39 . . . . . .not less than 5
MP39B . . . . . .10 to 60
MP40, MP40AA . . . . . .10 to 40
MP41 . . . . . .15 to 60
MP41A . . . . . .25 to 100
at T = 333 kelvin (140°F)
MP39 . . . . . .not less than 12
MP39B . . . . . .20 to 80
MP40, MP40AA . . . . . .20 to 120
MP41 . . . . . .30 to 180
MP41A . . . . . .50 to 300

Maximum Reverse collector current at V_{cb} = 5 V:
at T = 293 kelvin (68°F) . . . . . .15 μA
at T = 333 kevin (140°F). . . . . .250 μA

Maximum Reverse emitter current
at T = 293 kelvin (68°F), V_{cb} = 5 V . . .30 μA

Maximum Base Resistance at V_{cb} = 5 V,
I_e = 1 mA, F = 500 kHz. . . . . .220 Ω

Total output conductivity in small signal mode, under idle conditions,
V_{cb} = 5 V, I_e = 1 mA, F = 1 kHz maximum 3.3 μS

Maximum Collector capacitance at
V_{cb} = 5 V, F = 1 MHz. . . . . .60 nF

Table 1: This is an English translation of the Russian datasheet.

change Vbe and thus the emitter current will be dramatically affected changing the stage gain.

This is, in fact, a horrible idea and the only advantage of doing this is the reduced parts count over using proper self-bias where a resistive divider provides constant voltage to the base.

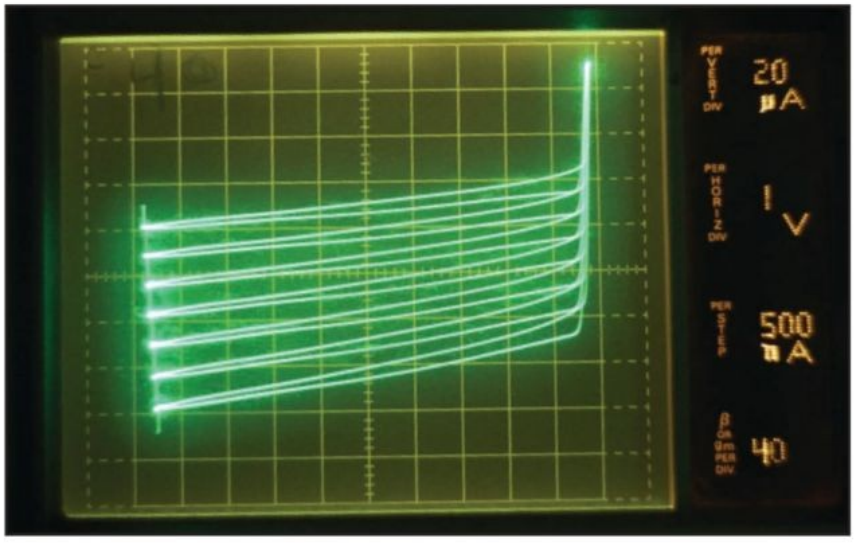
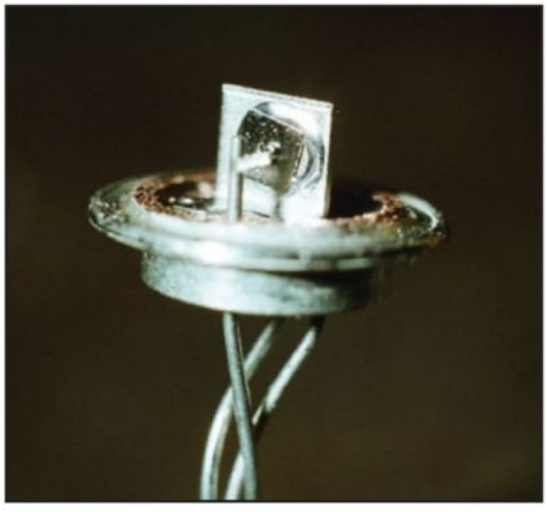


Figure 1: If you measure the curves, you will find that some don't match the datasheet but in fact have loops in them.

Photo 5: Cutaway view of the device



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.

Another workaround you may occasionally see is "collector feedback bias" where a base-to-collector resistor provides a bias voltage that is proportional to the collector voltage (and therefore, inversely proportional to collector current). This provides some negative feedback to stabilize the operating point and it is less horrible than simple fixed bias, but it is still affected considerably by transistor beta.

When you encounter these circuits in the wild, you may find that changing them to modern self-bias greatly improves stability and consistency, and that in the end you spend much less time tearing your hair out. Modifying for self-bias is one of the important steps in converting germanium-based circuits for use with silicon transistors but it is also a very good plan when making any transistor substitutions of any kind.

Transistor Substitution

Transistor substitution for the most part is easy. If you are substituting germanium for germanium or silicon for silicon, there are a few simple rules. First, make sure that the beta is similar. Note that although beta and Hfe aren't exactly the same thing, they are close enough so that if you have a transistor specified as Hfe = 50 to 100 and you have a transistor whose beta is specified in that range, you should be fine.

Second, make sure that the breakdown voltage is high enough. If the Vcbo on the original transistor is specified as 20 V, you can replace it with a transistor with a 25 V Vcbo, no problem, but likely not one with a 10 V Vcbo. On the other hand, if you're on a 6 V supply and not driving any reactive load, that 10 V part might be okay.

Third, make sure that the Ft is in the ballpark. If the Ft is much higher than that of the original part, you may find that the circuit oscillates when it was perfectly stable before. If the Ft is much lower, you may find yourself with poor high frequency response. Try and replace like with like.

After doing the substitution, measure the base-emitter voltage. If it's something crazy high or crazy low, then there is an issue with leakage. It could be the circuit was designed to be biased by leakage, it could be the circuit is being biased by leakage when you don't want it to be. The schematic will likely tell you for sure.

Now, there are some circuits where this method is insufficient, namely pulse circuits and circuits that operate in saturation. That could mean a guitar pedal or a TV vertical sweep circuit. For these circuits it may take some tinkering to make everything happy, likely starting with setting bias.

For switching circuits you may find that these transistors don't turn off completely until the base is pulled down hard. This is typical of most germanium

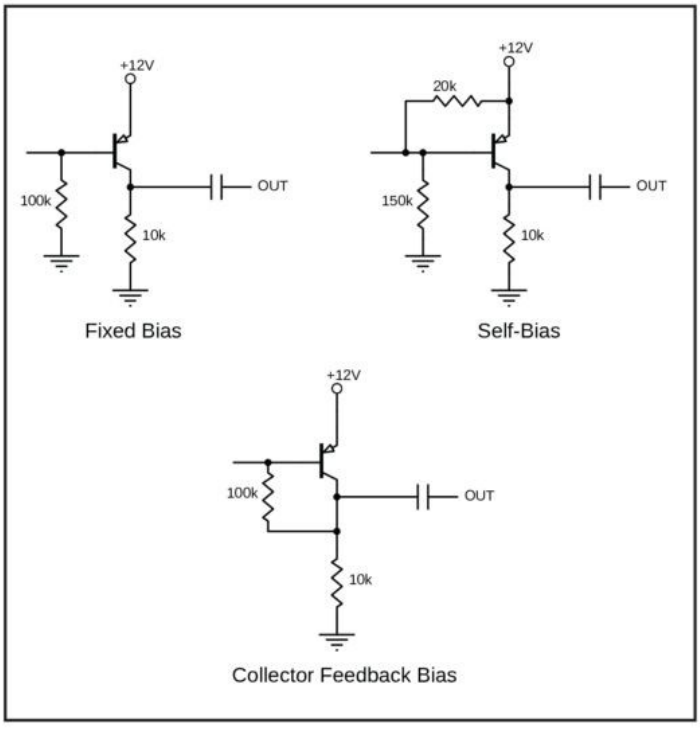


Figure 2: Transistor biasing methods include: the fixed bias, the self-bias, and the collector feedback bias.

parts although Sprague for instance made some transistors designed for computer circuits that were better in that regard. These MP series transistors are not a good choice for substitution of those.

Note that it's possible to substitute silicon transistors for germanium transistors in many circuits. You may need to change bias resistors to get Vbe into the correct ballpark but sometimes you don't even need to do that. However, if you do this, the circuit will sound different. This may or may not be what you want. I have replaced a lot of germanium parts with quiet and linear 2N5087s over the years but just as many times I have had people who wanted to preserve the original sound.

One oddity that the datasheet does not warn you about is that the case is connected to the base, so it cannot be used as an electrostatic shield in most cases. An external shield might be needed in some circuits that had relied on a grounded can for shielding.

Overall Opinion

These are some interesting components, readily available on the surplus market, which are going to have a unique sound and different behavior in audio circuits because of the various curve shapes compared with modern silicon transistors. Their clipping behavior will be softer and cleaner as well as their normal small-signal behavior being different. It is worth trying them in new circuits and old ones.

Author's Note: I would like to thank Andrey Gordenko and Andriy Pshenichniy for the test semiconductors.

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Market Update

Voice Capture and Recognition

In this Market Update, audioXpress explores some of the latest developments in voice technologies and highlights a few companies that are at the forefront of voice development platforms. Due to space restrictions, we could only feature a few companies and products, but this is a topic that we will continue to discuss in future editions.

By
J. Martins
Editor-in-Chief

Voice processing and recognition, leading to the development of voice-based personal assistants is one of the greatest achievements of the decade. As Hooman Kashef, founder and CEO of Libre Wireless Technologies, stated, "Voice interface and control is experiencing rapid growth and fast becoming a primary interface for man-to-machine applications in virtually every market segment."

It took us a long time to get there—from basically the invention of the first computer to the early voice-to-text transcription efforts and the first microprocessor-based voice engines in the 1990s. Google and Apple converted the existing primitive voice recognition engines into useful applications for Voice Search and Interface (2008/2011), introducing cloud computing platforms and artificial intelligence to finally convert speech recognition and basic voice commands into natural language queries, paving the way for true voice interfaces and personal assistants. With Amazon Alexa, the popularity of voice personal assistants exploded, and we can finally say that voice is now mainstream.

Not that there aren't many challenges to solve—from all the other languages that are not English to accents and the creation of truly natural conversation engines, which is the ultimate goal for those assistants. But meanwhile, for the audio industry there is an unique opportunity. The sales of voice-enabled products created a market for amplifiers, DSP, speakers, and microphones as the industry has never seen. Sold in massive quantities, those devices have been propelling many manufacturers to completely new industrial heights, while innovation has been accelerated in areas such as far-field capture, acoustic compensation, voice filtering, and even automatic room correction for smart speakers.

And of course, those developments propelled by voice are now providing exciting developments in speakers and mobile devices, leveraging those sophisticated technologies now available to the consumer electronics industry. The same technologies used to create systems optimized for voice recognition, can also be used to enhance music listening or simply allowing speakers to become part of a smart home network. That makes the investment in new speaker designs much more exciting.

Alango Technologies—New Dimensions for Voice

Based in Israel, Alango Technologies is a company that should be on the map for any manufacturer looking at differentiating voice products and grabbing the gigantic market opportunities that are presented in this space, particularly with hearables. Alango is a technology company focused on voice communications, particularly in the areas of voice quality, intelligibility, and hearing enhancement. It is also a leading developer and licensor of front-end DSP technologies for voice communication and mobile audio.

The company's front-end voice enhancement technologies are used in communication products for hard-of-hearing people, mobile phones, hands-free car kits, Bluetooth headsets, audio-conferencing systems, voice intercom systems, and many other

applications. Digital processing solutions from Alango improve the quality of voice communication and audio experience in a variety of real-life scenarios reducing different types of noise, enhancing speech, cancelling acoustic echo, and dynamically equalizing signal volume and spectrum according to dynamic acoustic conditions.

Alango works with virtually all the leading semiconductor companies on those projects and that's probably the reason why the company's name is not as frequently mentioned, considering the company's DSP software is embedded in many millions of products currently in the field. Its eighth-generation Voice Communication Package (VCP8) development platform is now powering new solutions in the audiology field, while its ongoing collaboration with CEVA, Freescale Semiconductor, and Synopsys, among others, is enabling many exciting implementations of voice in the mainstream market—from automotive to the smart home and connected devices.

Alango's new BeHear Now solution is a personal hearing product that combines the functionality of a customizable hearing aid, a Bluetooth stereo headset, and an assistive listening device.

Since 2016, Alango has been perfecting its own microphone beamforming and array technology for a wide range of applications, allowing projects to address speech recognition enhancement, including far-field speech enhancement. For these purposes, Alango licenses its DSP technologies, including its latest Speech Recognition and Voice Enhancement Packages. The company demonstrated the latest generation of those technologies at CES 2018 and at Mobile World Congress (MWC) 2018, showing a four-microphone platform for a voice-controlled Smart Speaker, and also a six-microphone TV-based linear microphone array.

Voice on Hearables

After 16 years, Alango has recently been focusing on another exciting new area related to hearing enhancement and personalizing sound for hearables. The company provides unique technologies for ambient awareness and personalization of the audio experience and its hearing enhancement solutions are among the most cutting-edge found in the market today, including the ability to control speech reproduction speed in real time.

To show a complete proof-of-concept, Alango has taken the unusual route of creating its own Wear & Hear (www.WearAndHear.com) line of assistive hearing products, starting with BeHear Now—an advanced Bluetooth stereo headset and hearing enhancer, featuring advanced audio processing with sound personalization algorithms and multi-microphone processing that cancels ambient noise and improves intelligibility.

Unique technologies include the company's ListenThrough enhancement of ambient sounds, music, and phone calls; and EasyListen, which provides the ability to dynamically modify or even slow down incoming speech in real time without distortion to increase comprehension.

Designed to look like standard wireless Bluetooth earphones, the BeHear solution supports full spectrum reproduction for live or Bluetooth-streamed music, while enabling users to stay aware of their surroundings, passing through important sounds. The solution also enables users to adjust the level of transparency, without compromising the music quality (ListenThrough). An integrated mobile app includes a self-administered hearing test that automatically generates a personalized profile for each user, adapting to a variety of environments. The user profiles can be fine-tuned in real time and in real usage scenarios to achieve the best results.

Most impressive and truly unique to the industry is the integration of Alango's unique EasyListen processing that improves the intelligibility of "fast-talk" and foreign language. The algorithm is able to analyze incoming speech in order to detect voice and pauses, and automatically compensate the processing delay, slowing down for the most critical words and sentences, while quickly catching up to the real time signal whenever possible. Useful for assistive listening, the real-time processing algorithm can also be applied in voice translation applications.

www.alango.com

Dr. Alexander Goldin, CEO and founder of Alango Technologies. Alango perfected unique audio processing algorithms, including Sound Effect Normalization to improve speech intelligibility in movies and TV programs by de-emphasizing background sounds and music.

Libre Wireless Technologies—From Wireless to Voice

Based in Bangalore, India, Libre Wireless Technologies has been gradually and consistently evolving into one of the leading development platform solution providers for voice recognition applications. With an early start in the areas of Wi-Fi and Bluetooth audio, wireless and media streaming software, providing SDK platforms, electronic modules and RF/antenna design solutions for wireless audio applications and particularly for portable speakers, Libre Wireless' expertise was key in enabling many big-name manufacturers to enter the smart speaker market, while its aggressive program of partnerships, allowed the company to be at the center of the market transition to voice digital assistants and voice-enabled solutions.

Since 2015, Libre Wireless' development solutions for WiFi and Bluetooth audio have powered countless products from leading CE brands on cost-sensitive applications. In 2016, the company evolved its wireless audio connectivity solutions to support voice recognition applications and, in 2017, it demonstrated its LibreSync Platform with support for Amazon Alexa and the Alexa Connected Speaker APIs, placing Libre Wireless at the center of this important technology transition.

Libre's wireless mic-to-cloud voice and media streaming platform solutions enabled complete integration of Alexa and the Alexa Connected Speaker APIs and was once again key in enabling several consumer brands and ODMs to easily incorporate and leverage Alexa voice service together with multiroom audio control via Alexa.

Libre Wireless Technologies' offices in Bangalore, India, and an earlier photo of Libre's engineering team, with Hooman Kashef at the center.

Libre Wireless' MAVID platform enables turn-key and fully customizable, end-to-end solutions—all integrated into a single miniaturized device.

The company's Alexa voice control features fully leveraged Libre's underlying multiroom synchronization technology combined with new advanced cloud-based services, referred to as Libre Cloud Services, which connect Audio/IoT end points to Amazon Alexa, enabling these new features. The company's platforms have been proven to provide competitive advantages in terms of cost, size, power, multi-protocol, and multi-function features.

Its position was further reinforced with the introduction at the end of 2017 of the company's new Multiprotocol Audio, Voice, IoT Device (MAVID) solution—an entirely new embedded voice-enabled multiprotocol wireless connectivity and IoT platform. This ultra-low power and incredibly small device incorporates the complete hardware and software solution for embedded voice, IoT, audio, and wireless connectivity. MAVID not only enables OEMs and ODMs to build unique voice-enabled products but also allows them to create entirely new applications previously not yet realizable, including very small battery-powered, portable, and wearable products for next-generation voice interface products. A true added value in a fast-moving and highly competitive landscape.

The unique and fully integrated miniaturized hardware/software MAVID platform provides performance optimization and wireless coexistence management for a wide range of features, including extensive host MCU features, 802.11ac dual-band Wi-Fi, Bluetooth audio streaming, Bluetooth Low Energy (BLE), and a far-field voice processing interface, including Automatic Echo Cancellation/beam forming/Wake Word recognition, and wake-on-voice features along with integrated and embedded memory. MAVID also offers options for adding Zigbee, Thread, LoRa, DECT/ULE, and LTE connectivity for broader interoperability and multiprotocol applications.

This extensive hardware integration now supports multiple voice services and media streaming applications, all running on a very fast and power efficient real-time operating system.

Libre Wireless also recently introduced its mic-to-cloud LS5BV LibreSync voice module with integrated far-field voice interface/control based on DSP Group's SmartVoice Technology and HDClear SoC. Once again, this has created a powerful platform with fully integrated and optimized far-field voice interface capabilities as well as extensive music streaming and multiroom audio features

for OEM and ODM manufacturers. The module uses DSP Group's sophisticated algorithms for detecting and processing far-field voice, including advanced noise reduction, echo cancellation, music barge-in, and audio processing features.

The compact LS5BV module is designed to support from one to three mics, offering flexibility for various applications that can be optimized for cost, front facing 180°, or circular 360° applications. The LS5BV offers a clean interface to off-module mics, enabling the product designer ultimate flexibility on mic placement and product integration while providing a simple low-cost interface in a fully integrated mic-to-cloud voice solution.

DSP Group describes the resulting solution as an incredibly sophisticated yet fully integrated hardware and software solution, while Libre's Hooman Kashef praised DSP Group's "high-integration and high-efficiency and low-power operation of voice processing algorithms."

www.librewireless.com

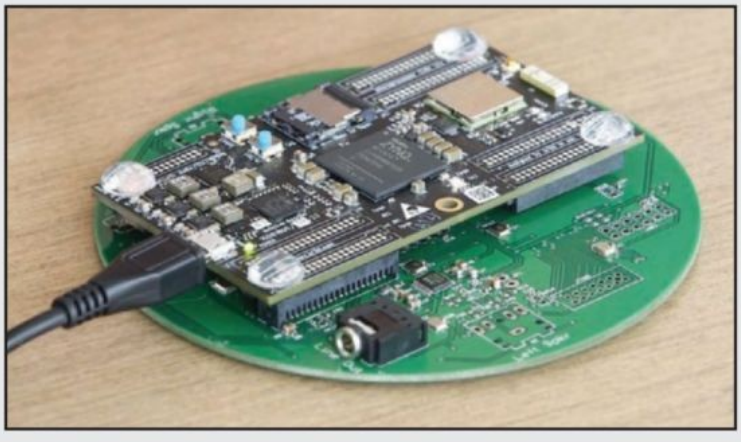
In 2018, Libre Wireless Technologies helped Audio Design Experts (ADX) launch the new RIVA Concert and RIVA Stadium speakers, the latest models from RIVA Audio, now with support for Amazon Alexa.

Aaware and Infineon

Aaware is a global supplier of sound capture technology for far-field, voice-enabled digital products. The company's Sound Capture Platform was designed to provide product development teams with higher performance sound capture integrating algorithmic acceleration technology. Aaware's Acoustically Aware sound capture algorithms enable product development teams to create robust far-field, voice-to-machine communications functionalities, helping to solve the challenges of canceling noise, echo, and reverb, which is critical for cloud-based digital assistants, while featuring the ability to recognize more than one wake word/phrase.

The Aaware platform separates source speech (wake word) and follow-on speech from interfering noise with ultra-low distortion, enabling the system to more easily integrate with third-party speech and natural language engines. In addition, source localization data can be forwarded to downstream applications, improving the performance of sophisticated multi-sensor artificial intelligence (AI) applications.

Aaware's Acoustically Aware sound capture algorithms, paired with TrulyHandsfree wake-word detection technology from Sensory, allow systems to adapt to differing noise interference without requiring calibration for different environments or integration of the reference signal.



Aaware's Sound Capture Platform was designed in cooperation with Avnet and Xilinx to provide product development teams with higher performance, scale, and flexibility. The platform has a concentric circular array of 13 microphones that may be configured in different combinations, allowing for product performance tuning. It separates source speech (wake word) and enables follow-on speech from interfering noise with ultra-low distortion, allowing the system to more easily integrate with third-party speech and natural language engines.

Apart from collaborating with Sensory in the development of voice solutions, Aaware has partnered with Xilinx to power edge acceleration for voice-enabled AI applications.

Recently, Infineon Technologies, and Aaware announced a partnership to launch a next-generation voice development platform using its Acoustically Aware technology combined with Infineon's digital microelectro-mechanical systems (MEMS) microphone technology. The new voice development platform is based on the Zynq Ultrascale+ MPSoC device and features Infineon's most sensitive XENSIV Digital MEMS microphones. As Infineon stated when the new platform was announced, "We are impressed with Aaware's voice capture technology and complete single chip SoC implementation for noise, echo, reverb-cancellation and wake word, and we are intrigued with their vision for advanced sound capture and neural network support."

www.infineon.com | www.aaware.com

Frontier Silicon Multi-Ecosystem Voice Control for Smart Speakers

Frontier Silicon is a leading supplier of technology solutions for digital terrestrial radio, smart home, and smart audio products. The British company—part of Frontier Smart Technologies Group, which also encompasses Frontier Microsystems—is now an established development partner and technology provider for many audio manufacturers with solutions for digital and Internet radio, Wi-Fi-enabled wireless speakers, soundbars, and more recently, voice recognition solutions.

From silicon through software to production-ready platform designs, Frontier Silicon has supported brands such as Bose, harman kardon, Onkyo, Panasonic, Philips, Pioneer, Pure, Sony, Yamaha, and many more on these projects.

The company's Smart Audio platform is integrated in a reference platform, Minuet, creating a complete toolkit for brands and manufacturers to develop smart audio devices with voice assistants (e.g., Amazon Alexa or Google Assistant) quickly and cost-effectively. Since 2016, Frontier Silicon has been one of a small number of solution providers chosen by Google to work on this technology.

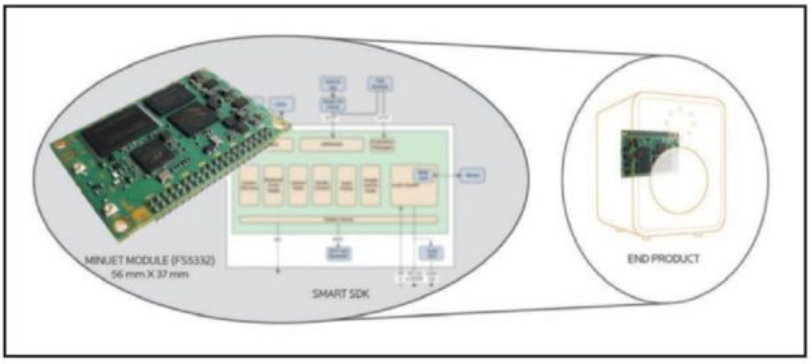
Minuet has enabled brands and manufacturers to develop wireless streaming and was expanded to support voice interfaces on smart speakers, access Internet services, and control a wide range of smart home devices.

Frontier's Minuet module includes the entire design set needed to build an audio product, including datasheets, schematics, gerbers, mechanical drawings, bill of materials (BOM), application guidelines, simulation tools, iOS/Android reference apps, and libraries. It comes pre-certified for CE, FCC, Wi-Fi Alliance, and Bluetooth and also offers Apple AirPlay, Spotify Connect, Bluetooth, and Internet radio support. Minuet incorporates all the elements required for a state-of-the-art audio device, including diversity antennas, a dual-core ARM processor, and multiple audio inputs and outputs. The complete engineering BOM for the Minuet speaker toolkit is as little as US $30.

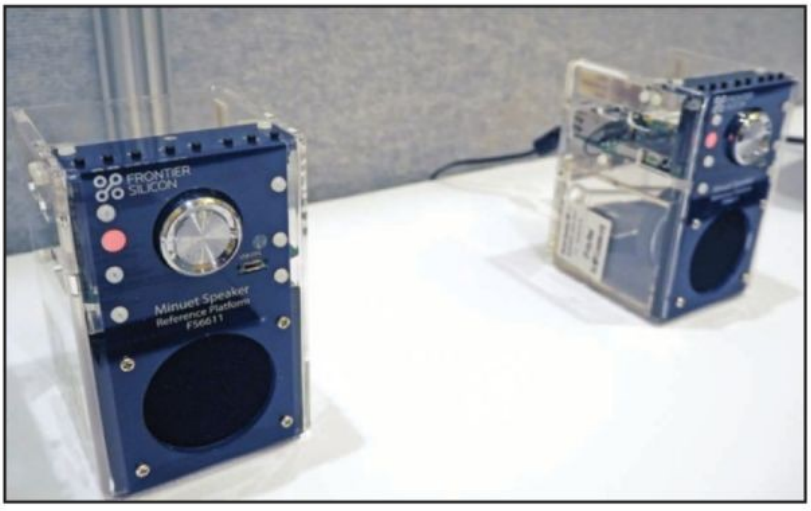
Recently, Frontier announced a significant enhancement to its Minuet Smart Audio platform and SmartSDK, to enable smart speakers, soundbars, and AVRs to be simultaneously controlled by voice-enabled devices from Google, Amazon, and Apple—helping manufacturers to differentiate, and not have to compete head on, with first-party devices. Frontier calls it its "WorksWith" reference design.

Manufacturers and brands will be able to position their devices as being controllable via any of the major voice assistant ecosystems, launching new products that are able to support "Works with Google Assistant," "Works with Alexa," and can be voice-controlled via an Apple HomePod. In addition, the platform also added support for multiroom applications with Google Chromecast built-in and Apple AirPlay 2.

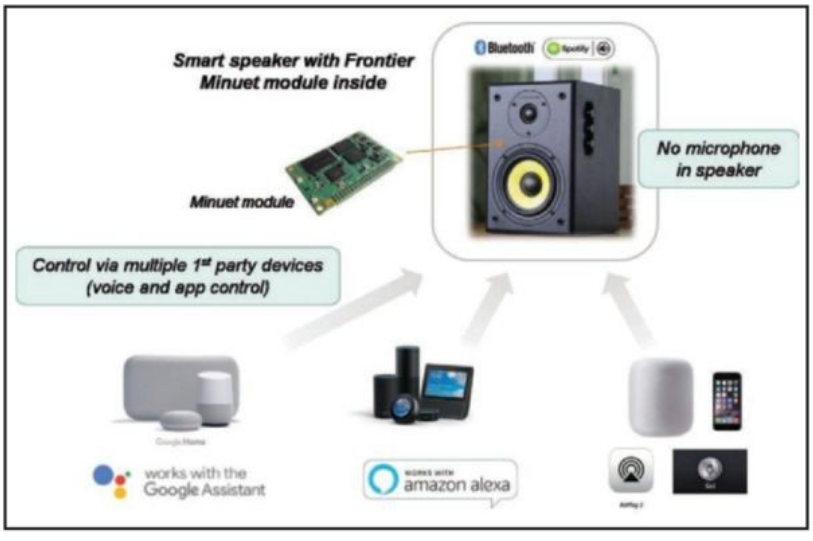
www.frontiersmart.com



Frontier Silicon's Minuet speaker toolkit is a design platform enabling audio manufacturers to develop wireless speakers incorporating multiple voice assistants, which means faster product development and mass market pricing.



The Minuet speaker toolkit includes the entire design set needed to build an audio product.



Frontier's new "WorksWith" reference design supports Google Assistant, Amazon's Alexa Voice Services, and Apple's Siri.

Qualcomm Smart Audio Platform and Speaker Development Kits

Providing technologies and complete development solutions for manufacturers based on Qualcomm's key platforms has been a focus of the San Diego, CA, company for the last two years. Following the first demonstrations at CES 2018, Qualcomm introduced a series of new platforms, system-on-a-chip (SoC), and development kits targeting new advanced audio applications and smart speakers, including the support of key wireless technologies, voice, and cloud ecosystems. Solutions include its Smart Audio Platform optimized for smart speakers, and a six-microphone far-field voice platform qualified for Amazon Alexa Voice Service (AVS).

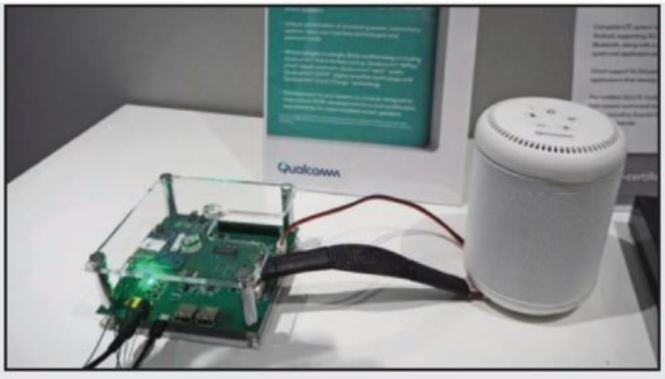
Originally announced in June 2017, Qualcomm's Smart Audio Platform was designed to help manufacturers accelerate the development and commercialization of smart and networked speakers with support for Alibaba AI Voice, Amazon Alexa, Baidu DuerOS, Google Assistant, and Microsoft Cortana.

The flexible solution offers two Qualcomm SoC options based on APQ8009 and APQ8017, with a range of software configurations that are designed to enable OEMs to create truly optimized smart speakers across various product tiers and categories. The integrated platform brings together a unique combination of processing capability, connectivity options, voice user interfaces, and premium audio technologies to help meet increasing consumer demand for comprehensively featured, highly intuitive smart speakers.

Qualcomm's Smart Audio Platform is engineered to bring together a unique combination of processing power, connectivity options, voice user interfaces, and proprietary audio technologies engineered for high-resolution audio, networked audio, echo cancellation, beamforming, noise suppression, and barge-in wake-word detection. With support for both Linux and Android Things operating systems, as well as voice services from major cloud providers, the Smart Audio Platform offers manufacturers a variety of choices to help them participate in the growing smart speaker opportunity.

Built on the Smart Audio Platform, the development kit features a Wi-Fi certified System-on-Module (SoM) that integrates the key system components, helping to reduce development time and overall design cost. The kit also includes schematics and design files to support easier customization and differentiation in the manufacturers' products. Manufacturers can use a single design for different SKUs with varying operating systems and capabilities, which helps to improve cost efficiency and reduce development time.

Qualcomm Smart Audio Platform is qualified by Amazon for the Alexa Voice Service (AVS) and the company offers a reference platform with all the hardware and software building blocks required for a smart speaker.



Qualcomm's smart speaker development kits are engineered to help developers and audio manufacturers simplify the development of smart speaker products, at a range of price points, thereby helping them take advantage of the commercial opportunities presented by this rapidly growing ecosystem.

Additionally, the development kit offers a reference design for smart speaker devices. This reference design features a six-microphone array and speaker housing to support far-field voice interface and high-performance wake-word detection—including barge-in capabilities supported by echo cancellation technology, pre-tuned to the hardware. The speaker housing is designed for a rich 360° sound experience.

More recently, Qualcomm introduced highly integrated development kits based on QCA4020 and QCA4024 SoCs, designed for all sort for connected applications in smart cities, toys, home control and automation, appliances, networking, and all types of home entertainment devices. The kits provide interoperability among diverse devices including popular wireless standards, protocols and communication frameworks, and ease connectivity to various cloud and application services.

www.qualcomm.com

Some of the products in the market that use Qualcomm's Smart Audio Platform include the Cleer Space Smart Home Speaker launched at CES 2018.

Synaptics Full Voice Portfolio

In June 2017, Synaptics acquired Conexant Systems in an effort to accelerate its efforts in the expanding market of far-field voice solutions. Conexant was already a leading provider of audio and voice technology solutions, and Synaptics had meanwhile acquired the video and audio processing division of Marvell Technology Group, which also included key IP on advanced audio processing. In many of the earlier far-field voice demonstrations at shows, the Conexant four-mic development kit became a familiar sight, while the company's voice processing solutions have been featured in more than 60 different products, from home appliances to security devices and set-top boxes (STBs), prior to the acquisition by Synaptics. This, naturally, placed Synaptics front and center in all the development projects related to voice recognition technologies and human-machine interfaces.

Currently, Synaptics combines the full portfolio of hardware and software solutions for audio and voice applications, with a significant IP portfolio in DSP, analog and mixed signal technology, and embedded software to enrich and expand audio and voice capabilities of products across every vertical platform.

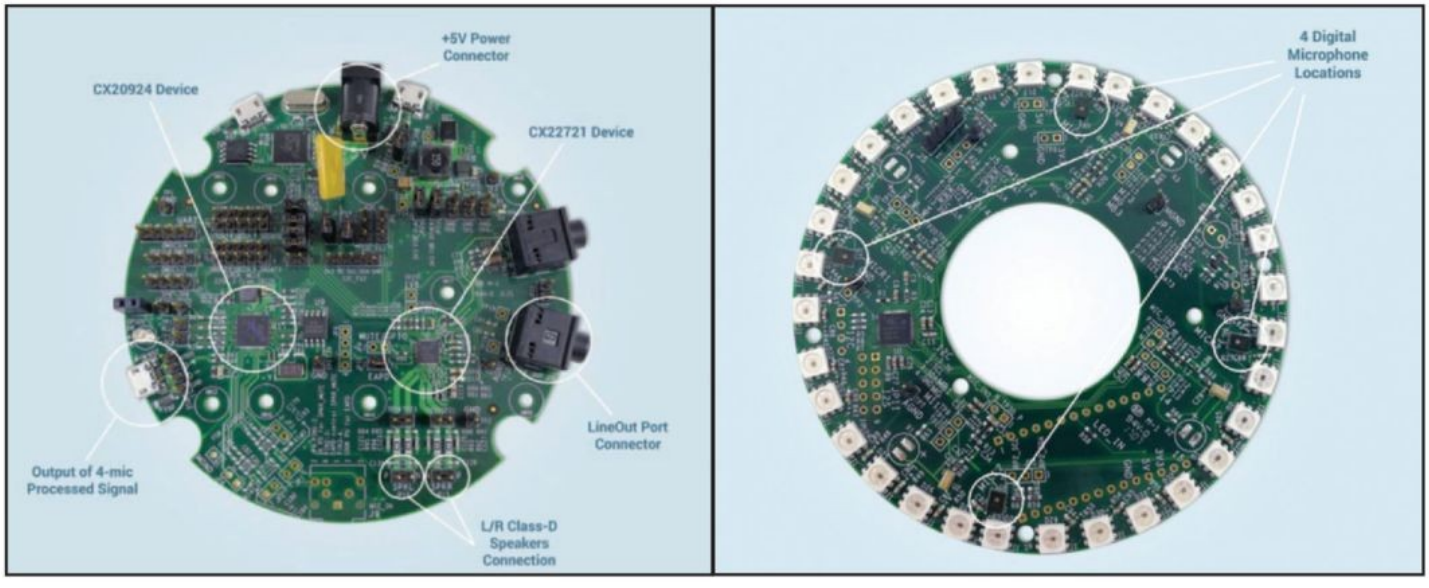
Currently, we can find Synaptics AudioSmart far-field voice platform in many smart speakers on the market, and the California company is working on new efforts—such as the project with Baidu in China for its new DuerOS Mobile Accessory (DMA), a voice-based AI platform that the company is implementing in Bluetooth speakers, headphones, wearables, and automotive accessories.

The DMA Bluetooth device solution is the first of its kind in China and will allow tens of millions of products to instantly become voice-activated AI devices. Synaptics also collaborated with Baidu in multiple far-field, voice-enabled devices, including soundbars and smart speakers available in the Chinese market.

Synaptics' low-power far-field AudioSmart product portfolio was developed to support electronics manufacturers' efforts designing voice-enabled consumer and enterprise products, providing a cost-effective and fast time-to-market advantage. Synaptics' key AudioSmart features include full-duplex acoustic echo cancellation (AEC) barge-in, and the ability to detect the user's voice omnidirectionally, even if noise sources are coming from the same direction as the user. AudioSmart separates speech signals from background noises and only processes the human speech, providing superior recognition accuracy, while Synaptics development solutions are currently able to support all major Automatic Speech Recognition (ASR) services.

With the AudioSmart platform, Synaptics is able to offer a complete far-field voice development platform with leading performance for speech recognition systems, using two-mic and four-mic advanced voice processing, working with all current voice assistants and supported with reference designs, drivers, and tuning tools.

Key to these types of projects is Synaptics' CX22721 stereo class-D amplifier platform, designed to enable advanced in-air playback functionality for systems that use Synaptics Voice Processors, and featuring advanced Power Management and SpeakerShield real-time protection features, as well as support for digital-to-analog conversion with sampling frequencies up to 192 kHz. Dedicated hardware DSP capabilities enable a high-quality audio experience, especially as part of a closed-loop system with a



The Synaptics AudioSmart 4-Mic development kit features the company's CX22721 stereo class-D amplifier platform, optimized for voice processing, and the CX20924 far-field voice input processor system-on-a-chip (SoC).

Synaptics Voice Processor. This includes a seven-band/channel hardware EQ and a single-band DRC (with a single knee), which, independent of any host-based effects, can be combined to optimize speaker response and increase device loudness while simultaneously minimizing distortion. A programmable high-pass filter on the Class-D output removes DC offset from the I²S audio channels, protecting the speakers from overheating.

Already in 2018, Synaptics made an important announcement, regarding the company's second-generation AudioSmart digital headset SoC solution for USB Type-C headphones. Again, the combined key technologies developed by Conexant and embedded in this solution are powering multiple products from smartphone makers, making Synaptics a global leader in USB-based headset technology, and related implementations of hybrid active noise cancellation, smart voice pick-up, and noise suppression and high-resolution codecs in mobile audio solutions.

Another key focus for Synaptics is the growing trend of bringing far-field voice to STBs and TVs. Recently, Synaptics announced two new AudioSmart Far-Field Voice solutions for both emerging and legacy STBs integrated into its VideoSmart VS-550 multimedia processors (SoC). For the millions of existing STBs already deployed, this allows users to simply connect a Synaptics-enabled standalone accessory via USB to instantly add a voice assistant to their television services.

As Synaptics recognizes, consumer demand for smart speakers to control music and home automation is rapidly growing, and smart home users now expect to be able to also use their voice to control their TV experience without having to hunt for a custom remote control. The company is able to combine its comprehensive solution portfolio of market-leading voice and video solutions to provide a low-cost and fast to market solution for this market.

Synaptics also offers a solution for ODMs to develop a simple to use plug-in USB accessory that takes advantage of USB ports common in STBs, as a lower-cost alternative to a standalone voice-assistant device.

www.synaptics.com

A joint project with NXP, Arrow Electronics, Synaptics, and Amazon, this Development Kit for Amazon Alexa Voice Service (AVS) combines the Synaptics AudioSmart 2-Mic and NXP Pico i.MX7Dual SoC, enabling manufacturers to quickly and cost-effectively integrate Alexa into a fully functioning prototype and then take that work into design for manufacturing and production.

Skyworks and Avnera Corp.

Skyworks Solutions recently announced it has signed a definitive agreement to purchase Avnera Corp., the developer of Analog System on Chips (ASoCs) and one of the main technology providers for audio development in the areas of analog/mixed signal audio, voice, and speech processing engines. Avnera solutions are used by many reputable consumer electronics brands for all wireless audio applications as well as Bluetooth, Lightning, and USB-C product development.

SKYWORKS®
BREAKTHROUGH SIMPLICITY

The acquisition of Avnera, a private fabless semiconductor supplier, augments Skyworks' leadership in wireless connectivity by adding ultra-low power analog circuits to enable smart interfaces via acoustic signal processing, sensors, and integrated software. As Skyworks details, the acquisition expands its activities in synergistic, high-growth applications, extending the company's reach. Immediate target applications include AI speakers/microphones, virtual assistants, intelligent gaming controllers and vehicle in-dash systems as well as wired/wireless headsets.

At the heart of Avnera's system solutions are market-specific ASoCs that combine breakthrough audio/voice analog circuits, highly efficient power management, and custom hardware accelerators for smart acoustic signal processing/AI features. Key Avnera customers include Audeze, Harman/JBL, Panasonic, Philips, Pioneer, Polk, Samsung, Sennheiser, Sony, Skullcandy, Vizio, and Yamaha.

www.skyworksinc.com

XMOS—Voice at the Cutting Edge

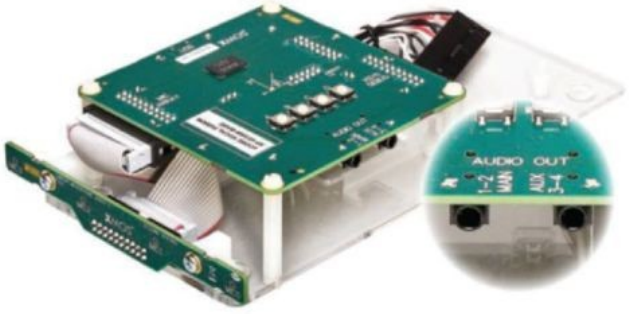
XMOS is a leading provider of audio and digital signal processing (DSP) platforms and audio connectivity solutions that currently power numerous consumer, studio, and broadcast audio solutions from hundreds of vendors, including high-resolution audio, multichannel, and networked audio platforms. Since 2015, the British company has evolved its scope to become a leading provider of voice interface solutions. And, its xCORE family of Multicore Microcontrollers is delivering comprehensive Voice User Interface (VUI) controller solutions for all kinds of devices. Backed by some of the largest names in high-tech venture capital, in 2016, XMOS acquired Setem Technologies, another British company and a pioneer on algorithms that enable separation of voice or conversation signals within a crowded audio environment to achieve optimized input into speech recognition systems. The two companies had been collaborating on next-generation voice interfaces and decided to combine the companies and are now a leading vendor in the fast-growing application area.

Based on Setem's 3S Platform, XMOS was able to leverage a patented mathematical representation of the far-field soundscape to improve the quality of speech recognition for all applications. The software platform basically solves the "Cocktail Party Problem" by applying Advanced Blind Signal Source Separation technology to focus on a specific speech source and mitigate extraneous background noise or interference, working with a minimum of two or three microphones.

From its early xCORE-Voice Array Microphone, XMOS quickly expanded its range of voice development platforms based on its VocalFusion technology, which is able to aggregate multiple audio streams from microphone arrays, convert the analog Pulse Density Modulation (PDM) microphone signal to a digital PCM stream to which DSP can be applied, and pass the optimized audio stream to remote or local voice recognition engines.

With support for circular or linear arrays, with 360° and 180° far-field voice capture in excess of 5 m distance, the VocalFusion devices apply sophisticated DSP on the captured signal—including mono/stereo Acoustic Echo Cancellation (AEC), dereverberation, and noise suppression, to address the problems of distance, isolation, removal of noise, and the establishment of direction of voice capture. DSP is also used to implement barge-in functionality, allowing the device to understand speech and commands as the product outputs audio streams.

The xCORE VocalFusion Speaker kit provides direct interfacing to four Pulse Density Modulation (PDM) microphones in a circular array, with a choice of USB or I²S interfaces.

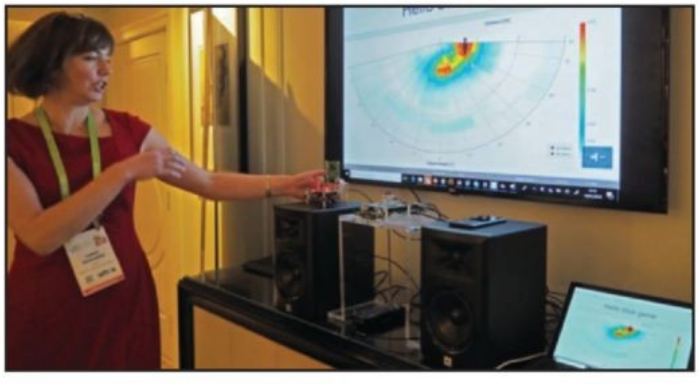


XMOS VocalFusion Stereo Dev Kit for Amazon AVS was the first linear mic array solution qualified by Amazon for far-field performance that supports stereo acoustic echo cancellation.

In 2017, XMOS partnered with Infineon Technologies to deliver a completely new building block for voice recognition, featuring a combination of Infineon's BGT60TR24B 60 GHz 2Tx/4Rx radar IC with accompanying antenna, 70 dB SNR silicon microphone sensors, based on Infineon's dual backplate MEMS technology, and an audio processor from XMOS. In combination, the solution provides far-field voice capture by audio beamforming combined with radar target presence detection. This provides optimal sound recognition and flawless execution for digital voice assistance across a broad range of voice-controlled devices, ideal not only for smart speakers, but also for smart TVs and set-top boxes (STBs), among many other voice-controlled applications.

The XMOS audio processor processes the signal data from an array of microphones, adjusting the angular and distance data from each microphone to produce a beam at an angle identified by the radar data. An illuminated LED indicates presence detection, from which sound is taken. The combination of Infineon radar and XMOS beamformer enables the microphone to precisely target a specific target source, even as the target moves around a room of talking people and inanimate sounds. The solution has also been implemented with XMOS VocalSorcery, a Blind Source Signal Separation technology that can spatially identify individual speakers or conversations within a crowded noisy audio environment.

A more recent development was XMOS' new VocalFusion Stereo Dev Kit (XK-VF3500-L33), the world's first stereo AEC far-field linear microphone array solution—based on the XVF3500 voice processor, which delivers two-channel full stereo-AEC. The processor interfaces directly to four PDM microphones in a linear array with 33.33 mm



XMOS VocalSorcery is a Blind Source Signal Separation technology that can spatially identify individual speakers or conversations within a crowded noisy audio environment to achieve optimized voice capture and input into speech recognition systems.

inter-mic spacing. Uniquely, the solution also supports configurable AEC latency, where the AEC reference signals can be accurately calibrated, and the latency adjusted to enable after-market far-field voice accessories for existing products.

This new voice processor development kit is designed for developers working in the growing smart TV, soundbar, STB, and digital media adapter market segments—all of which require true stereo-AEC support to deliver compelling "across the room" voice-interface solutions.

These and other solutions have been expanded by the XMOS engineering team. Anyone who has seen the company's demonstrations of stereo-AEC voice capture and blind sound source separation understands why the British company is leading the market in advanced embedded voice and audio solutions for edge-of-room applications.

www.xmos.com

Sensory—The Wake Word Pioneer

Sensory was the originator of wake words for personal assistants and is by far the most adopted solution for voice recognition applications. The Silicon Valley, CA-based company recently raised the performance bar for always-listening wake word and speech recognition with significant upgrades to the embedded Artificial Intelligence (AI) in its sixth generation of TrulyHandsfree, boosting the technology's already industry-leading wake word performance and accuracy by more than 65%. The new solution introduces upgraded turn-key voice interface solutions for devices of all shapes, sizes, and power requirements.

Additionally, the updated TrulyHandsfree SDK boasts improved deep-neural network training that allows for even better near- and far-field speech recognition performance in all room conditions.

Version 6.0 improves performance and word recognition accuracy relative to the last two generations of TrulyHandsfree by reducing wake word false positives by more than 65%. This reduction is due to Sensory's new high-resolution speech feature front-end that sees a higher resolution digitized representation of the speech audio, combined with the introduction of on-device wake word post-qualification about wake word events to better discriminate against false positives.

In addition to enhanced wake word performance, Sensory's new high-resolution speech feature front-end has improved upon what was already the benchmark for embedded speech recognizers. These front-end enhancements in accuracy also enable TrulyHandsfree to support multiple wake words, like "Okay Google" "Alexa," "Hey Cortana," "Hey Siri," and "Xiaodu, Xiaodu" in a single implementation with great performance. This enables device makers to create new products with a user-friendly voice interface that works with more than one digital assistant technology.

Sensory also upgraded the machine learning within TrulyHandsfree to take advantage of high-resolution audio information for deep-neural net training. This improved training allows the algorithms to anticipate a variety of factors associated with wake word performance, including understanding how one person, or a population of people, may pronounce a wake word. It also takes into consideration acoustic challenges (e.g., various room configurations, device placement, room size, reverb, and echo). This ensures superior always-listening speech recognition performance regardless of where the device is placed in a room.

The update also improves privacy and data security. TrulyHandsfree 6.0 does all processing on device, keeping voice data completely safe by never storing it or sending it to the cloud. Additionally, TrulyHandsfree 6.0 includes a layer of voice biometrics recognition in the voice interface for user authentication and security. TrulyHandsfree's embedded high-resolution voice enrollment and speaker verification (SV) technology allows users to enroll their voice and their own custom wake word or passphrase, restricting unauthorized users from accessing the voice user interface. Even if an unauthorized person learns the custom wake word or passphrase, Sensory's voice biometrics technology will recognize that it's not the enrolled user speaking and not authenticate them.

Specifically tuned to provide an ideal voice barge-in experience with TrulyHandsfree, Sensory now offers an upgraded AEC solution that supports single-mic input systems with mono or stereo sound sources. This technology allows users to interrupt their devices by saying the wake word while the device is in the middle of playing voice prompts, music, or other sounds, at any volume level.

TrulyHandsfree's AEC technology is also scalable and can be tuned to optimally balance acoustic echo cancellation performance requirements and MIPS/power restrictions. For ultra-low power devices that have limited battery capacity such as wearables, Sensory offers its Low Power Sound Detector (LPSD) hardware component for DSPs and smart microphones. LPSD reduces low-power configurations of TrulyHandsfree wake words to operate at an average battery draw of under 1mA.

Sensory offers off-the-shelf wake word models for all major IoT cloud services, including Amazon AVS, Apple Siri, the Google Assistant, and Microsoft Cortana, as well as wake word models for third-party devices that support cloud AI systems from Baidu, Alibaba, or Tencent. Sensory can also combine multiple wake words into one solution.

Sensory's TrulyHandsfree supports US English, UK English, Arabic, Dutch, French, German, Italian, Japanese, Korean, Mandarin, Portuguese, Russian, Spanish, Swedish, and Turkish. The TrulyHandsfree SDK is available for Android, iOS, Linux, QNX, and Windows. Sensory provides developer support for cloud service interfaces on Linux, Android, iOS, and Windows as well as support for dozens of proprietary DSPs, microcontrollers, smart microphones, and other low-power embedded devices.

Additionally, ultra-low-power deeply embedded ports of TrulyHandsfree are available for leading DSP/MCU IP cores from ARM, Cadence, CEVA, NXP, Synopsys and Verisilicon, as well as for integrated circuits (ICs) from Ambiq Micro, Analog Devices, Cirrus Logic, DSP Group, Fortemedia, Intel, Knowles, Microchip (Microsemi), NXP, Qualcomm, QuickLogic, Realtek, Synaptics, STMicroelectronics, Texas Instruments, Yamaha, and XMOS.

www.sensory.com

Voice Assistants

The Essential Next-Generation Consumer Interface

Amazon Echo's success has paved the way for a new smart speaker category, increasing shipments of home audio devices by 50%.

By
Simon Bryant
(Director of Research, Futuresource Consulting)

Apple Homepod's share remains small in volume, but its relatively high average selling price (ASP) means its share of spending is more than three times higher than its volume share. Proprietary devices continue to represent 19 out of every 20 speakers sold, while third-party devices from the Sonos, Harman, Sony, and many more legacy audio brands compete within that one in 20 device share. This is leading many to pursue platform-agnostic, "works with" and other strategies.

It's been three short years since Amazon launched its Echo speaker system. During that time, sales have exceeded 42 million units and Echo's success has paved the way for a new smart speaker category, increasing shipments of home audio devices by 50%. On top of that, the category has achieved almost $3 billion in retail revenues. And let's not forget Echo's role in the rise of voice assistants (VAs), which are pervading a wide range of devices. Futuresource Consulting's upcoming voice assistant report forecasts more than five billion devices will be in use in less than five years, creating a solid platform for voice-enabled content and services.

Focusing on this year, 16 million smart speakers were sold in the first half of 2018, doubling the figure achieved in the first half of 2017. The innovative feature-sets built into VAs, coupled with low price points, continue to whet consumer appetites for the latest tech. This is particularly the case in markets such as the UK, the US, and, increasingly, China, which are all avid adopters of the latest gadgets and represent nine out of every 10 smart speaker units sold, despite the fact smart speakers are now available in many more countries. These three markets represent around 40% of the wider consumer electronics (CE) market, giving an indication of the market potential of smart speakers as the rest of the world plays catch up with the traditionally large CE countries.

Privacy Concerns and Regional Differences

Privacy concerns remain a barrier to adoption, due to the "always listening" nature of smart speakers and wake-word VAs. This appears to be particularly problematic in Germany and Eastern Europe. End-user research carried out by Futuresource supports the assumption that privacy is a key inhibitor to consumer uptake, but Futuresource expects these concerns to diminish over time as the value and convenience trade-offs become increasingly evident to consumers.

According to the 2018 edition of Futuresource Consulting's "Smart Home Devices & Appliances" consumer survey, 38% of non-adopters of smart speakers "can't see a use for smart home devices," with a third citing privacy concerns. VA vendors must continue to build compelling use cases and, perhaps more importantly, address consumers' fears of having devices in their homes that eavesdrop on their conversations. Apple's focus on tackling similar issues at the Apple Worldwide Developers Conference (WWDC) was evidence of a wider industry focus.

While smart speakers already come in a variety of shapes and configurations (with screens, Zigbee,

better audio, and cameras), consumers are generally choosing the most basic of feature sets, with Dot and Home Mini representing almost two-thirds of Amazon's and Google's volumes. Although cost will be a factor, this suggests far-field access to the assistant is the primary driver of demand.

Smart Speakers: A Threat from Within?

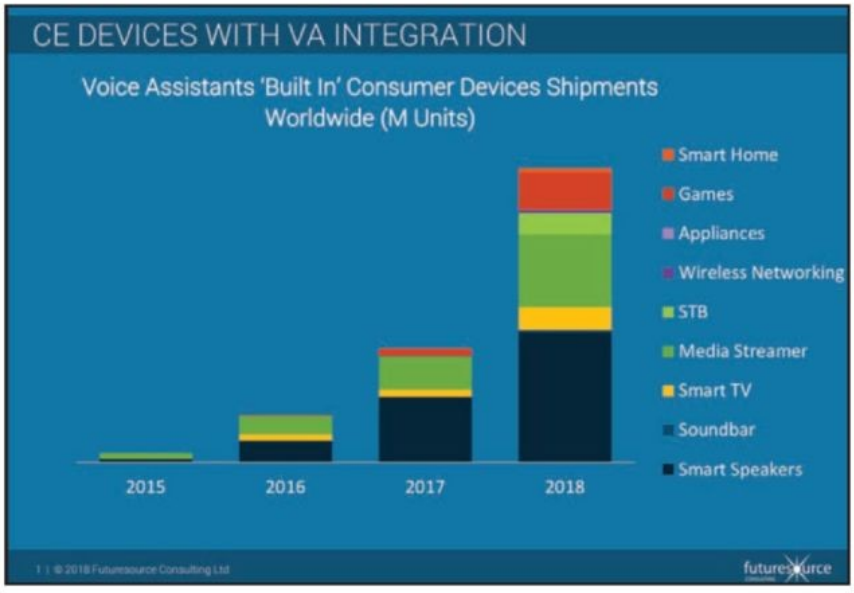
A longer-term threat to the smart speaker market comes from its integration into a variety of other devices, both inside the home and beyond. Industry attention is shifting from audio and smart home to the battle for the living room, particularly the TV user interface. 2018 will see a growing number of soundbars, smart TVs, media streamers, consoles, and set-top boxes (STBs) incorporating far-field microphones to wake up Google Assistant and Alexa. This year, while 53 million smart speakers are forecast to sell, a further 65 million TV centric devices supporting Alexa, Google Assistant, Siri, and Microsoft's Cortana (Xbox) will also ship. Three million of these will support far-field wake-word microphones. Moreover, support for far field is growing at the expense of push-to-talk buttons on remote controls, as consumers seek the same hands-free access to VAs offered by smart speakers.

The Living Room and Beyond

The battle for the living room is likely to be the most hotly contested VA battle in the smart home, and the TV presents a set of unique challenges, as well as significant opportunities. From changing volume, to switching the set on and off, to requesting a movie, this could ultimately play a role in controlling the wider smart home. As a result, VAs should mutually benefit Amazon's, Google's, and Apple's existing ambitions in over-the-top (OTT) media services and TV video connectivity, assets which include Android TV, Fire TV, Prime Video (Amazon), Chromecast, iTunes, and Apple TV.

Amazon's growing influence in the entertainment space in recent years supports its bid to become the lynchpin in the living room entertainment experience, through both its own branded services and through controlling third-party apps. Its boldness and risk-taking in both consumer technology and the media space are further emphasized by its recent Premier League rights deal in the UK, where there are now more than seven million Prime members. This was an expected step, but unusual for Amazon given it is a UK-only deal.

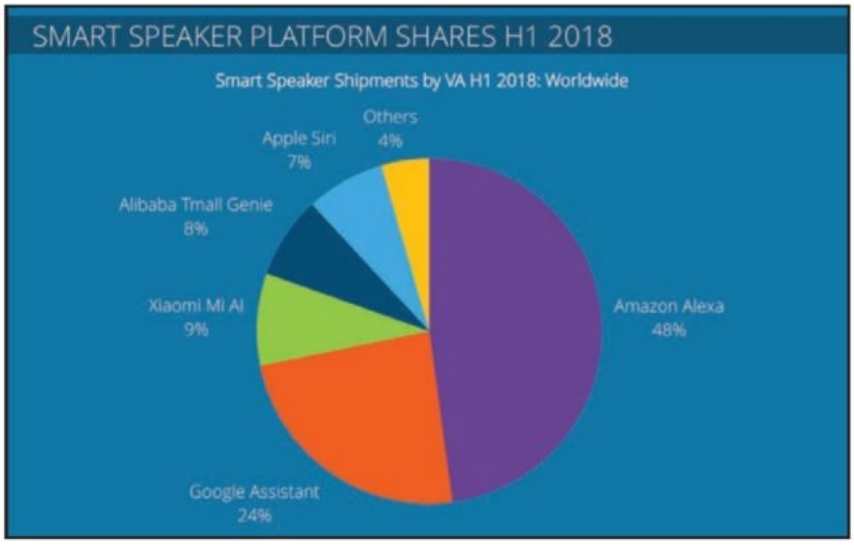
As well as the Fire TV Cube media streamer with support for Alexa, Amazon is now also powering smart TVs with Fire TV built-in. Amazon is the number one vendor within the media streamer market worldwide, with more than 20% shipment share last year. Despite only officially being available in just a handful of countries, it has overtaken its main rivals—Apple and Google. However, the media streamer market in comparison to smart TV is relatively small, with 53 million streamers expected to ship globally this year compared with 155 million smart TVs.



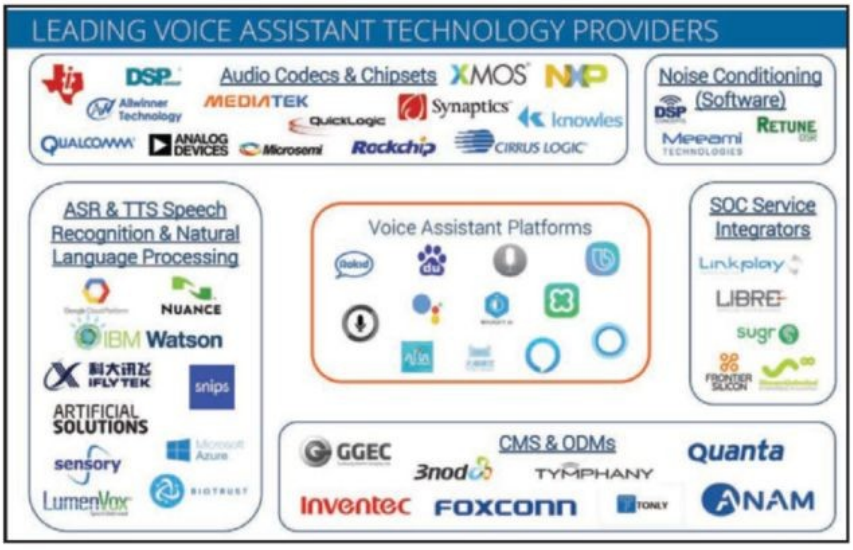
Following the popularity of smart speakers, other consumer electronic categories are now also implementing voice assistants. (Image courtesy of Futuresource Consulting)

The B2B Angle

While the trajectory of VA technology is currently being propelled by the consumer segment, the industry is exploring how it could transfer across to B2B verticals and applications. Opportunities are rife in meeting rooms, operating theaters, factory floors, classrooms, and much more. Many of the value points so highly regarded by consumers are likely to resonate in B2B applications, though perhaps with more challenges to overcome for widespread



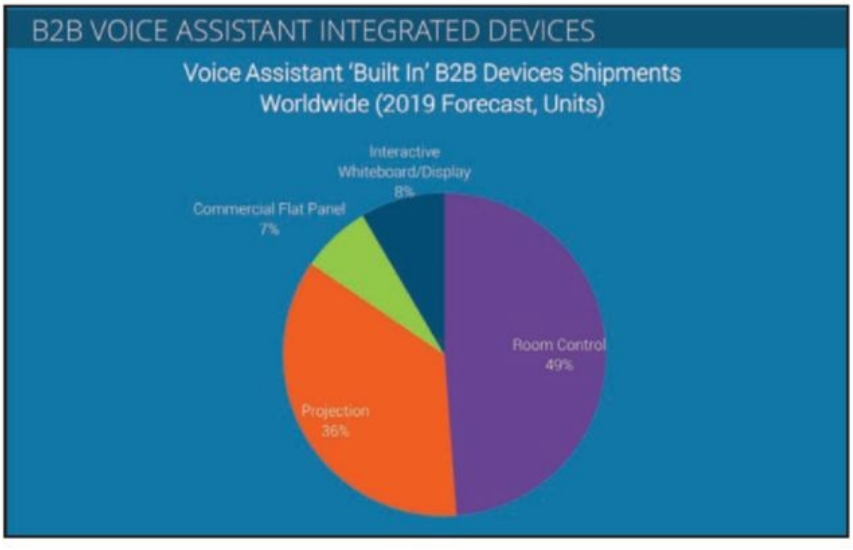
As growth begins to shift to markets outside North America and the UK, platform share will move increasingly toward Google Assistant and Asian platforms, overshadowing Alexa as Amazon's geographic reach remains limited.



These are the leading voice assist technology providers.

adoption. Some companies have already produced interesting proof-of-concept solutions to attract interest and understand customer requirements, with IBM and Amazon coming to the forefront as pioneers in this space.

Despite a slow start, VAs will eventually prevail in B2B. By 2022, 82 million voice-enabled B2B devices will be shipped. By comparison, the consumer device market in the home will represent more than 450 million "built-in" units shipped, while mobile devices, wearables, and cars will add a further 1.9 billion "built-in" devices shipped, amounting to more than five billion VA-installed devices in use by 2022—predominantly smartphones (supporting Google Assistant, Siri, and Bixby), but also in cars on the road, B2B devices, and the home CE devices (i.e., speakers, TVs, etc.).



B2B applications for voice assistants are rife in meeting rooms, operating theaters, factory floors, classrooms, and much more.

Then, Now, and Next

While 2017 was the year consumers and OEMs started to familiarize themselves with VAs and explore the potential, 2018 is the year when VA devices are establishing themselves, growing rapidly, with the focus of attention shifting from getting the devices into people's homes to extending their usage and controlling household entertainment (e.g., music and TV, or integrating into the home's infrastructure, controlling lighting, security and heating).

Wireless speakers will remain the key gateway device for introducing VAs to consumers, with CE vendors (e.g., Sony, Panasonic, and LG) launching me-too devices, while premium brands (e.g., Bang & Olufsen and Technics) launch their own speakers with the emphasis on unique design. The visual display capability offered by smart displays will offer a compelling method for delivering voice-activated content and will establish itself as a new form factor in the wireless speaker market.

Beyond smart speakers and even CE, voice is poised to be the next-generation primary consumer interface and is now being incorporated into many different devices, used for a variety of applications, and is stretching its legs beyond the consumer market.

Looking to mobile, it does have the potential to be the main carrier of VAs, but ultimately the touchscreen will compete with VAs as the primary user interface. Moving forward, wireless headphones and wearables have the potential to unleash VA usage in mobiles as wake-word technology and push-to-talk buttons penetrate these categories. Futuresource forecasts show that 24 million headphones will support VAs in 2018, a figure that's forecast to rise to 154 million by 2022.

Developments in areas such as AI, with natural language processing, machine learning, facial recognition, and emotion engines, coupled with improvements in computing and network infrastructure and multi-modal control (e.g., gesture and voice), will allow VAs to become more proactive, conversational, and human in their interactions. Instead of just answering questions and following commands, the VAs of the future will make recommendations, inform our state of mind, and assist with decision-making. This dialog will be based on our habits, likes, aspirations, and goals, accumulated from the vast stores of user information and behavioral data that has been gathered about us. VAs are beginning to open up a world of convenience, immediacy, and cohesion that we've never before experienced. For more information, visit www.futuresource-consulting.com.

Turning Up the Innovation on Smart Speaker Designs

By
Benwei Xin
(Business Development Manager, Texas Instruments)

Wenchau Albert Lo
(System Engineer, Personal Electronics, Texas Instruments)

There is increased competition in the smart speaker industry to create designs that provide enhanced functionality and ease of use. Smart speaker features now include interactive voice response, artificial intelligence, smart home control, and multimedia playback. To succeed, designs must distinguish themselves in terms of overall system cost, battery runtime, thermal considerations, voice recognition error rate, and premium audio quality.

Overall System Costs

Reducing system cost is a priority for the entire smart home industry. In the smart speaker space, the greatest opportunity for designers to potentially reduce system cost is in the selection of the audio amplifier. Instead of trading audio performance for cost savings, Texas Instruments' (TI) audio team developed solutions to reduce the system cost around the amplifier while improving performance.

For audio systems under 10 W per channel, for example, TI's electromagnetic interference suppression technology enables the use of ferrite beads instead of inductors. Given that the average cost of four inductors is $0.16 and the average cost of four ferrite beads is $0.04, designers could save an average of $0.12 in a two-channel system with this feature alone.

Battery Runtime and Thermal Considerations

Small battery size makes smart speakers smaller, lighter, and portable. Higher efficiency and better thermal considerations help reduce battery size while retaining battery runtime.

However, there is a constant power draw from the microphones and system-on-a-chip (SoC), which are listening for the next instruction. Thus, designers must look for other areas where they can decrease standby power and thermal dissipation.

One of the best places to start is the audio amplifier because it has the highest standby power draw and highest thermal dissipation apart from the SoC. To reduce standby power, audio amplifiers with multiple modulation modes can change the duty cycle of the output pulse width modulation (PWM) to control the current flow from the inductor to the speaker, reducing idle current up to 50%. This will reduce the overall power draw and improve thermal dissipation, allowing for longer battery runtime without increasing battery size.

Acoustic Echo Cancellation

Accurate speech recognition is a defining feature of a smart speaker, affecting the performance of the virtual assistant (VA). One of the challenges is that the microphones pick up all types of background noise in the environment, including other appliances, people talking, and even the music the smart speaker is playing. To filter out background noise and recognize user voice commands, smart speakers should have more than one microphone and employ complex techniques for accurate speech recognition.

One of those techniques is acoustic echo cancellation (AEC), which takes the digital signal of the music being played and subtracts that signal from the input of the microphone. AEC digitally cancels all background noise, including the music. To better filter out background noise, audio amplifiers specifically designed for AEC have a post-processed SDOUT signal. As shown in **Figure 1**,

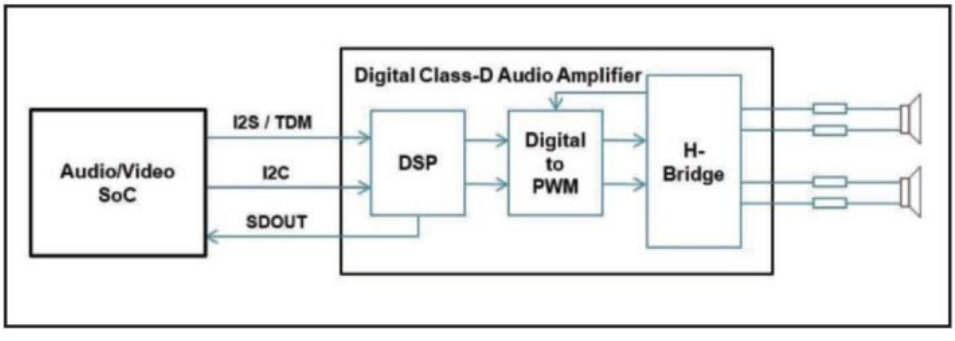


Figure 1: This is a block diagram for an acoustic echo cancellation (AEC) system.

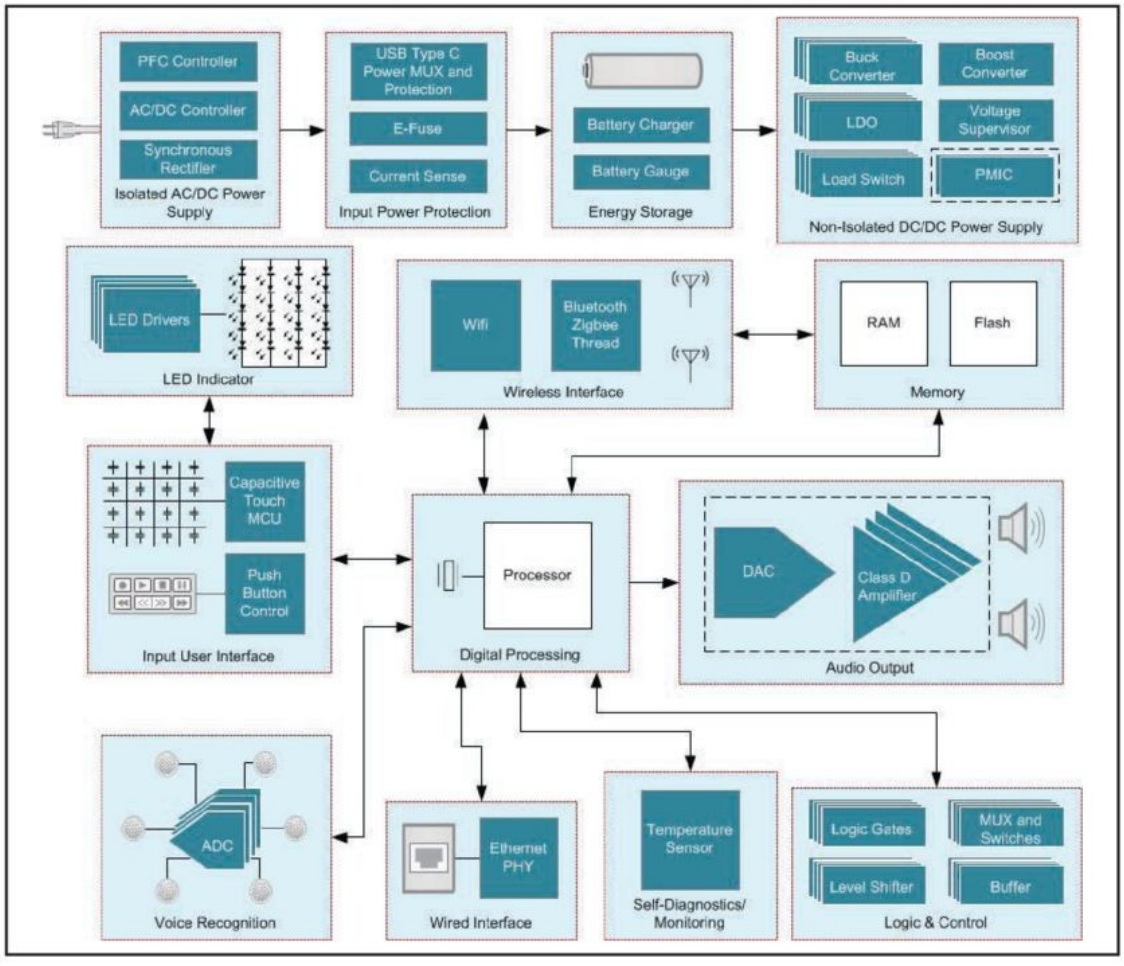


Figure 2: Here is Texas Instruments' smart speaker system block diagram.

the SDOUT is fed to the SoC in a way that filters out the music signal the microphones are receiving.

Premium Audio Quality

Smart speaker designs are small and attractive. But with a smaller speaker size, there is less audio output power. Amplifiers with robust speaker protection can achieve bigger sound from a speaker without damaging it. By using algorithms and models, speaker protection amplifiers analyze the incoming music signal, then deliver peak output power up to five times the nominal speaker rating, resulting in an output that is two to three times louder, without changing the speaker. This means that even a small smart speaker can sound bigger than it looks.

Resources

See these TI reference designs for smart speaker systems at www.ti.com/smartspeaker.

Less than 90 mW Ultra-Low Standby Power Auxless AC/DC Power Supply Reference Design 12 mm × 12 mm, 5-Rail Power Sequencing for Application Processors Reference Design

Human Machine Interface with LED Animation Reference Design

MSP432 MCU with MSP430 Capacitive Touch MCU, Haptics, and LCD Reference Design

PCM1864-Based Circular Microphone Board (CMB) Reference Design

Ultra-Mobile, Low Power nHD Display Reference Design with TI DLP Technology for Low-Cost Processors

What's Next?

Achieving high-fidelity sound is one of the smart speaker market's top priorities, as is the convergence of interactive voice response with traditional speakers. Although most Bluetooth speakers on the market today have no speech recognition or interactive voice-response functionality, it will become a de facto requirement.

Battery-powered smart speakers will need to meet the same power demands as line-powered speakers. This will be the next opportunity for smart speaker designs.

Over the past 10 years, as speaker design trends have shifted from larger residential applications to smaller, portable personal applications, TI has provided the technology and systems expertise to support these designs. **Figure 2** shows TI's smart speaker system block diagram.

To view an interactive version of this block diagram, or to browse more reference designs, application notes and other resources addressing smart speaker design challenges, visit www.ti.com/smartspeaker.

Voice Capture Challenges

The Threshold of the Golden Age of Voice

By
Mike Klasco
(Menlo Scientific, Ltd.)

Audio product development, and audio magazines for that matter, have traditionally been all about music reproduction, but recently voice has taken center stage. Audio signal processing has been divided into two separate camps—voice and music processing. I think we can credit the iPhone for multi-tasking telephony and music and making it mainstream.

As Bluetooth protocols with enhanced sound quality along with stereo brought us wireless headphones paired to smartphones, which added fuel to the fire, we are reaching full ignition with voice assistance emerging in everything from smart speakers to soundbars, smart wall outlets, and now smart TVs.

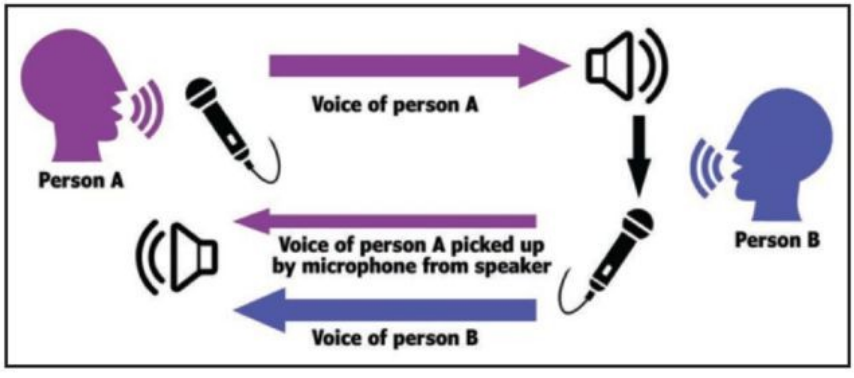
Voice processing has been with us since the start of Bell Labs, which was initially a telephony research department at AT&T's Western Electric set up in 1907. While Bell Labs has undertaken a wide range of research, voice was its primary focus from encryption vocoders to compressor/limiters and uplink and down link noise filtering to transducer refinements.

The radio broadcast industry has always been obsessed with extracting the maximum range from their transmitters—as advertising rates are calculated by the listener coverage area. In the 1950s, the CBS Volumax was the first effective broadcast automatic gain control (AGC) but its original purpose was to eliminate the signal level watchman—a technician whose only job was to prevent transmitter over-modulation. This overload would blow up the expensive transmitter tubes not to mention over-modulation would incur the wrath of the Federal Communications Commission (FCC). It did not take long for radio stations to also realize the dynamically compressed signal traveled to the fringe reception areas more clearly.

Decades later, it was multi-band limiters such Orban's Optimod and the Aphex Compeller that ruled the broadcast waves. In the 1970s, the back rooms of intelligence agencies were using sophisticated forensic voice processors such as those from Rockwell (now Rockwell Collins) to extract intelligible speech from noisy signals.

Today, you don't have to look behind secret doors to find some of the most sophisticated

From telephones to smartphones with voice recognition, the evolution of voice processing technologies is now the result of more than a century of research.



Acoustic Echo Cancelation (AEC)—the echo effect most often occurs when a side's B microphone picks up a signal from side A and rebroadcasts it back at site A, usually resulting in a feedback loop. An AEC processor monitors incoming signal on side B, and then removes it from the outgoing signal, taking the room environment into account. (Original image courtesy: http://lakeofsoft.com/vc/a_acousticechocancellation_AEC.html)

The Dolby Conference Phone with a Dolby voice-enabled service provides participants the benefit of voice placement: each speaker's voice is presented from a distinct location, making it easier to follow the conversation.

voice processing—it is in your pocket—in your smartphone. More than 1.6 billion cell phones are sold each year and that kind of number is hard to beat in the consumer electronics industry. Smartphones are big on signal processing, voice for telephony communications, and also for music playback, gaming, and movies.

Voice processing is the unsung hero in mobile audio devices—significantly enhancing voice call quality, intelligibility, and speech recognition (command) accuracy. Voice processing enables natural full-duplex communication in speakerphones, suppressing echo and acoustic feedback, and beamforming the mics for clear speech conversations in noisy places.

The vendors of these voice processing technologies operate in a fiercely competitive market, which drives continuous product design improvements. Further innovative R&D efforts will no doubt lead to better, faster, less expensive solutions that will benefit mobile device users in years to come.

Some of the processing takes place in the voice co-processor/codec (e.g., from Qualcomm and Cirrus Logic), but the processing smarts are distributed around a bit. Some processing, such as active noise cancelation (ANC), is starting to appear in the USB-C audio signal chain. And, most smartphones drive the receiver (earpiece) and speaker (speakerphone) with a smart amp, which monitors excursion and voice coil heat and limits operation to the safe operating area (SOA) of the speaker system.

The bottom line is smart amps enable manufacturers to get more sound out of tiny speakers crammed into shrink-wrapped enclosures without distortion, death, and destruction. Most of the leading semiconductor amp chip vendors such as NXP, Qualcomm, Texas Instruments (TI), and Maxim now offer smart amps.

Early Smartphones

Early mobile phone voice signal processing research was conducted by Motorola, Nokia, Ericsson, and the other pioneering mobile phone companies. Apple's Newton digital personal assistant, soon followed by Apple's development of iTunes (with iTunes initial launch in a Motorola mobile phone in 2004), and all this coalesced with the iPhone 2G, which debuted in 2007.

As smartphone computing power continued to ramp up, so did the voice processing in the audio hub/co-processor. Third-party algorithm specialists emerged, with many now absorbed by bigger fish. SoftMax (now Qualcomm's Fluence), Acoustic-Tech (now Cirrus Logic's SoundClear), Audience (merged into Knowles Intelligent Microphones), Convergent (now merged in Synaptics), among others, all fought each other for smartphone, tablet, and laptop design wins. A more recent entry (with some design wins in Samsung phones) is DSP Group's HDClear. Today, even the least sophisticated phone has processing power beyond the dreams of anyone just a few decades ago.

Voice Challenges

Let's state the obvious, while there are endless app and personal assistant functions, a mobile phone, at its core purpose, is voice communication and as such, sound clarity is the highest priority. The goal is to provide clearly intelligible speech in noisy environments. With the uplink (what you say) and the downlink (what the receiving party hears), users expect to communicate without intrusive noise. Every day users are taking calls while walking down a busy street in NYC, driving in a car with the window open and the radio blaring, at a windy beach, or in the din of a crowded restaurant). All these examples are challenging environments for even the most sophisticated state-of-the-art signal-processing algorithms.

Aside from voice processing for the obsequious smartphone, we have Bluetooth earphones and headphones. In 2017, the dollar value of Bluetooth earphones and headphones exceeded wired versions. There are two types of DSP cores embedded in the chipset or system-on-a-chip (SoC). For cost-effective

implementations the algorithms options available are in the read-only memory (ROM) embedded into the Bluetooth. The product development team at the headphone brand will pay to enable the features they want.

The more sophisticated and flexible path is to select the flash memory version of the Bluetooth IC, where special algorithm features can be selected and programed into the flash memory to run on the on-board DSP Bluetooth processor section of the Bluetooth IC.

Smart codecs combine the D/A codec with a DSP core and this approach is used for most digital wired ANC headphones, but more recently it is also used in the USB-C signal chain. Headphones that have a direct connection to the USB-C jack have yet to get traction with the brands.

Another approach to signal processing is a dedicated chip in the form of an application specific integrated circuit (ASIC), which can be interfaced with most of the preceding circuits.

Moving out of consumer audio electronics, the same types of voice signal processing can be found in desk speakerphones and conference phones and so forth.

Voice Co-Processors

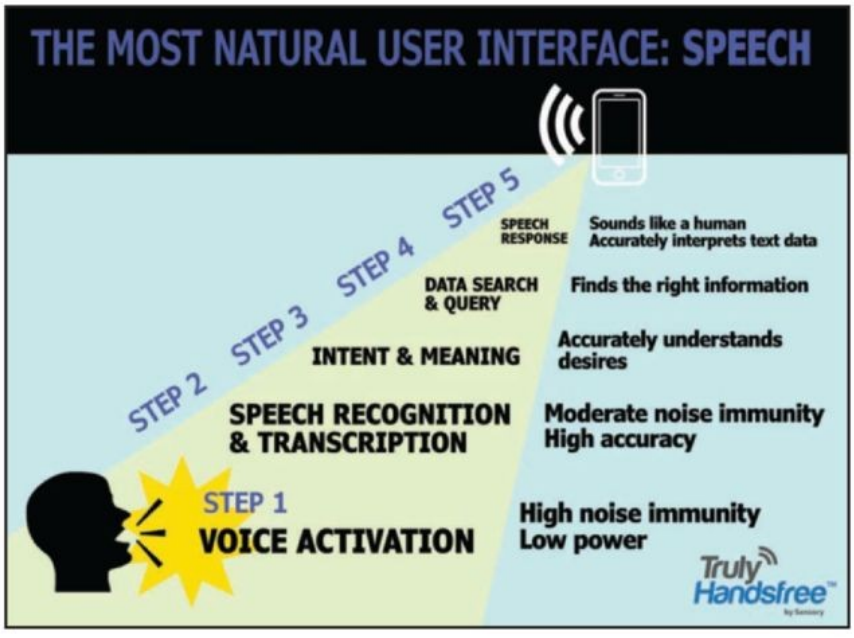
A co-processor's function is to assist the primary processor with specific tasks—increasing performance or freeing up processing power that can be used for something else.

In desktop and laptop computers, dedicated graphics chips or graphics processing units (GPUs) co-processors take the heavy demands of visual processing away from the central processing unit (CPU). Also, sound cards can be considered a class of audio co-processors that boast higher audio quality codecs, multichannel audio playback, and real-time audio DSP effects. Now, most smartphones use dedicated integrated circuit (IC) voice co-processors that are a type of ASIC. In the most sophisticated adaptive processors, intelligent speech tracking and adaptive processing analyze ongoing speech patterns and environmental conditions in real time. The result is a self-adjusting signal-dependent algorithm that emphasizes voice quality—whether in a quiet room or the noisiest vehicle, using techniques such as automatic level control and automatic volume control.

Noise suppression processing contends with crowd noise, audience babble, and similarly variable and unpredictable complex ambient noise that can significantly degrade speech intelligibility. Only the most sophisticated voice processors can effectively separate and filter this type of random noise from the speech signal. Mobile audio products tackle wind noise suppression through acoustics and signal processing. Yet signal processing is not a complete solution (e.g., when the mic is saturated when overloaded by the wind) and the signal output from the mic is distorted, then no amount of signal processing will be able to extract what was not cleanly captured.

The first line of defense from wind noise is acoustic filtering and is achieved through the use of mesh or non-woven wind screens, which is the same materials used in professional stage microphones. These materials are designed to be acoustically transparent while blocking the wind's force before it can overdrive the microphone's diaphragm. Once the microphone diaphragm is hit by too much wind pressure—too much is the acoustic overload point (AOP)—then the signal processing cannot do much to help. Many filter membrane materials also have high (IP) values, which means they act as barriers to prevent dust and moisture from entering the microphone.

Once past the wind screen's acoustic filtering, any remaining noise component of the wind turbulence can be attenuated through signal processing. Wind is not a constant noise. Between the wind's changing speed and direction, the user's unpredictable path, and the device's aerodynamics, we receive "non-stationary" noise. This means the noise's direction, amplitude, envelope shape, and duration may all



Sensory's TrulyHandsfree voice control technology is the leading solution for voice interfaces and offers multiple phrase technology that recognizes, analyzes, and responds to dozens of keywords. It consistently recognizes phrases even when embedded in sentences and surrounded by noise.

The
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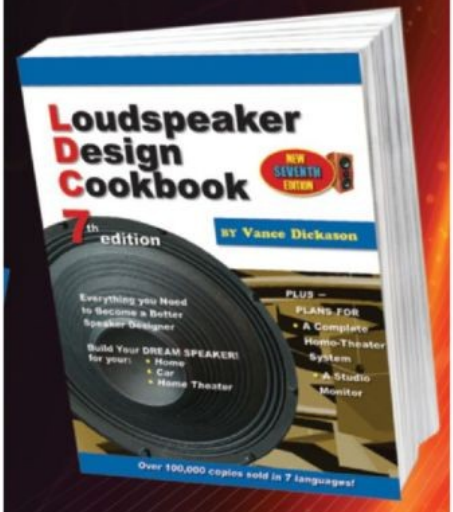

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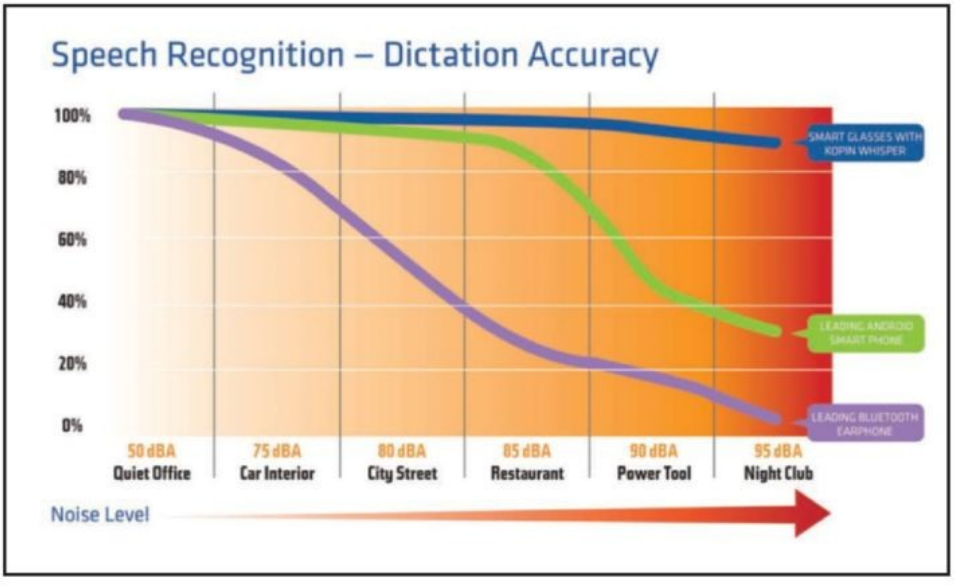
vary in a random fashion. Contrast this with stationary noise (e.g., a motor at constant speed and load when the user is not moving relative to the noise source). This situational noise is more easily suppressed with less computationally intensive signal processing.

Another type of noise that adaptive noise filtering must contend with comes from heating ventilation air conditioning (HVAC) systems. While air ventilation noise may not be that noticeable to those using a conference phone in a meeting room, for those on the receiving side of the call this type of noise can make the conversation fatiguing and unpleasant. Our hearing uses both ears to help us "tune out" ambient noise, but when the noise and the voice communication emanate from a single speaker source, our natural psychoacoustic processing does not work.

In the case of a party or similar environment where there are many simultaneous conversations (crowd babble), we experience the "cocktail party" effect. This lets us use our binaural hearing to zoom in on a single conversation that is not necessarily louder than the next. Dolby Labs has introduced a conference phone that actually enables up to three people to talk from the source room and (assuming both sides have this system) be reproduced through separate loudspeakers spaced in a circle, which reproduces the "cocktail party" effect.

Acoustic echo cancelers (AEC) are signal processing methods used to improve voice quality either by preventing echo from being created or removing it after it is already present. Consider a full-duplex speakerphone—when you talk, your voice will be reproduced at the far location through a speaker. If the microphone gain at the far location is too high (above unity gain), then your voice from the loudspeaker will be picked up by the microphone and sent back to you, resulting in an echo. There is always some degree of latency in voice communications systems, so the echo is further delayed, which can make conversation impossible. When conditions are right (wrong, actually!), the microphone



The Whisper chip from Kopin doubles the accuracy of automatic speech recognition in noisy environments and dramatically improves far-field voice quality in smartphones and wearables. The chart compares the performance of smart glasses equipped with the Whisper chip against the automatic speech recognition (ASR) and noise cancellation technologies found in two popular devices—a leading Bluetooth earphone and a leading smartphone. While the Whisper chip's performance remains consistent as noise levels increase, the earphone's performance begins degrading at 75 dB (the amount of noise associated with a car interior or dishwasher) while the smartphone's ASR performance starts to drop at approximately 85 dB (or the amount of noise associated with restaurant).

will pick up your echoed voice and send it back to the other location and the feedback loop begins.

Echo suppressors work by detecting a voice signal traveling in one direction on a circuit, and then inserting loss in the other direction. Usually, the echo suppressor at the circuit's far-end adds this loss when it detects a voice coming from the circuit's near-end. This added loss prevents the speaker from hearing his own voice. Echo cancellation involves first recognizing the originally transmitted signal that re-appears, with some delay, in the transmitted or received signal. Once the echo is recognized, it can be removed by subtracting it from the transmitted or received signal.

This technique is digitally implemented using an algorithm running on digital signal processor. AEC enables natural full-duplex conversations—two singers at distant locations can sing a duet without interference created by the send/receive signal paths.

The Near Future is Far Field—The Challenge of Voice Recognition

After decades of voice recognition development, with voice assistants (e.g., Apple's Siri and Microsoft's Cortana) introduced on smartphones and PCs, Voice Command Appliances (e.g., Google Home, Amazon Echo, and similar devices) have energized the smart home market. Finally, the accuracy of near-field voice assistance is robust. Near-field use cases include headphones even without a boom, talking to a laptop or smartphone, smart speakers that are nearby, and other use cases where the mic(s) are less than 6' away from the talker. Voice recognition processing is still a challenge when your voice commands have to compete with ambient noise (e.g., the TV or radio) but also when more than 7' or 8" from the source accuracy results start getting marginal. So, at the moment, sitting on the couch watching TV with the sound blasting, and changing channels or the volume through far-field voice command, remains elusive.

And no exploration of voice assistants and voice command is complete without touching on Sensory wake words used to trigger the service. Sensory's voice technologies are omnipresent in consumer electronics applications including mobile phones, automotive, wearables, toys, IoT, and home electronics. Sensory's TrulyHandsfree voice control is the standard for smartphone' ultra-low power "always listening" voice control—embedded in more than a billion consumer products.

Sensory has developed a suite of voice models for Alexa that are part of Sensory's TrulyHandsfree speech recognition engine. Now offering multiple phrase technology that recognizes, analyzes, and responds to dozens of key words, it consistently recognizes phrases even when embedded in sentences and surrounded by (modest levels of) noise. Sensory technology is now part of Qualcomm's FluentSoft SDK and for Android environments with other platforms.

Integrated TrulyHandsfree enables products to wake up and respond when the trigger word is called, so no button pressing, or manual manipulation is needed. This very low-power technology is designed to prevent false triggers during normal room noise and conversations.

One promising development to improve the implementation of voice recognition is the Whisper voice extraction processing that was originally developed by Kopin (www.kopin.com) for communication in the extremely high noise environments of the F35 fighter cockpit helmets. This pre-processor (roughly) doubles the accuracy of automatic speech recognition while improving far-field voice quality in smartphones, wearables, and voice command appliances. The algorithm can also run on some of the Bluetooth chips that accommodate flash memory. I had the opportunity to check this out myself and was impressed. I expect to see this embedded in a lot of products by 2019.

All this technology is the tip of the iceberg in voice processing and the tipping point for the use of voice interfaces in the smart home. For decades, there has been a struggle to make voice command and smart homes viable in real life. At the 2018 CEDIA show, voice assistance was embedded in most upcoming products being introduced at the show. The technology now has a toehold and it is only up from here.

Linear Tube Audio's ZOTL Ultralinear Power Amplifier

Photo 1: The LTA Ultralinear is the latest ZOTL amplifier design by David Berning, utilizing his Zero-hysteresis output transformerless audio design and his unique power supply.

This month, two of our esteemed authors joined forces to review the ZOTL Ultralinear Power Amplifier from Linear Tube Audio (LTA), which has been making tube amplifiers in its Washington, DC-area workshop since 2015. LTA was founded with the goal of sharing the experience of ultimate, immersive audio reproduction. To this end, its reference amps use the unique ZOTL topology created by celebrated audio designer David Berning.

By
Oliver Masciarotte
(United States)

ZOTL Ultralinear
Linear Tube Audio
7316 Carroll Ave.
Takoma Park, MD 20912
www.lineartubeaudio.com
Price : $6,800

After Stuart Yaniger finished running Linear Tube Audio's ZOTL Ultralinear Power Amplifier "through the ringer," he boxed it up and sent it diagonally northwest with the parcel arriving in good order. When I received the package, I was slightly surprised by the weight, or rather the lack thereof. That's the thing about hollow state, I always expect it to have an excess of iron (see **Photo 1**).

ZOTL Background Information

The Ultralinear is physically light, just under 15 lb, yet it isn't a lightweight in terms of the parameters that I value—extended frequency response, excellent published distortion performance, solid power transfer at the low end and "...improved amplitude and phase accuracy at frequency extremes" as David Berning's 1997 foundational "ZOTL" patent states. Harvey "Gizmo" Rosenberg, the manufacturer and pundit who coined the term ZOTL, cited David Berning's (zero) output transformerless designs as a "...high expression of the outer edge of the thermionic arts." Bonus ZOTL features include relatively low output impedance, auto-bias, extended tube life and protection against shorts, opens, and prolonged overloads. Not to mention, the tubes used are not hard-to-find expensive exotics.

LTA's Mark Schneider is the guy responsible for the Ultralinear, and has managed to carry the ZOTL banner farther than most. Schneider told me that his Ultralinear is designed specifically for high-efficiency speakers. He suggested that, with speakers of 90 dB sensitivity or higher, he would steer a prospect toward the $6,800 Ultralinear over his slightly lower priced $6,400 ZOTL40. "The (Ultralinear's) balance of triode and pentode is 'more pleasing to listen to,' per David (Berning), than triodes (alone), which are more forward." In comparison, his ZOTL40 performs best with low-efficiency speakers or when you are using more than a few watts. Schneider asserts that, "The detail of the Ultralinear surpasses the ZOTL40 at low power levels... (and) is the most detailed of all of our offerings."

Despite the OTL moniker, ZOTL amps are not conventional OTL designs. Lacking an output transformer, the traditional OTL approach requires a raft of paralleled output devices to match the high voltage, low current operation of tubes to the low voltage, high current

requirements of your loudspeakers. ZOTLs employ a high-frequency carrier, very similar to amplitude modulation, as an impedance converter to both eliminate the transformer and optimally match output impedance to the load.

In the case of the ZOTL Ultralinear, the carrier frequency is 250 kHz and the output impedance is rated at 2 Ω, though Stuart Yaniger informed me via e-mail that, "...output impedance (of the Ultralinear) is high enough that you'll have frequency response variations, which we laughingly call 'tube sound.'" As he'll probably mention, a 6 Ω load turned out to be optimal for power and distortion.

For my testing, I used a 16 Ω speaker, along with a nominal 6 Ω model...more on that in a bit. A traditional OTL design is "...not sufficiently low in output impedance under large signal conditions to properly control back EMF from many loudspeakers, and careful matching of amplifier to speaker is required," according to Berning's patent.

Audiophile lore states that linear power supplies are good, yet again design heterodoxy wins over ho-hum in the ZOTL Ultralinear. In keeping with the switcher at the center of the amp, the Berning-designed power subsystem is also a switcher, with attitude. Schneider contrasts bottom feeder, ultra-cheap switching PSUs, with their minimally sized and rated parts, to the next step up; a linear supply.

"Big transformers, low noise regulators, lots of caps for storage," Schneider opined, "...these nearly always sound better than switchers in audio. Along comes David, whose day job for decades was developing switching supply technology for a government agency. He designed a unique type of switcher optimized for audio. It is better than any linear supply, has less heat, universal voltage and power factor correction. However, it is the most difficult and labor intensive part of the amplifier to build, and a big part of the cost. David taught us that the power supply is half of the amplifier circuit. The performance of the amp is completely interwoven with the performance and quality of the power supply. Not all of the ZOTL amps use David's switching supply, but all the highest quality ones do." Berning's 1970s vintage Audionics BA-150 power amp and TF-10 pre both had switching supplies, so this concept is nothing new to the designer.

Speaking of turning theory into practice, I asked Schneider what the manufacturing difficulties were that he overcame. He told me Berning's amplifiers are difficult to fabricate since his technology is unique. You can't just sashay over to your computer and put in an order for most of the major inductive parts. "Transformers, impedance converters, filters, chokes, etc. all have to be handmade. There are many steps, it's complicated and unforgiving if you make the slightest error. So our challenge was to enable non-engineers to make these special parts. My background in manufacturing provided me with the processes (we implemented) to ensure consistent build quality. Our technicians are now very experienced in these processes, but it took a while. The major cost reduction is the cost of labor, which is the largest cost component."

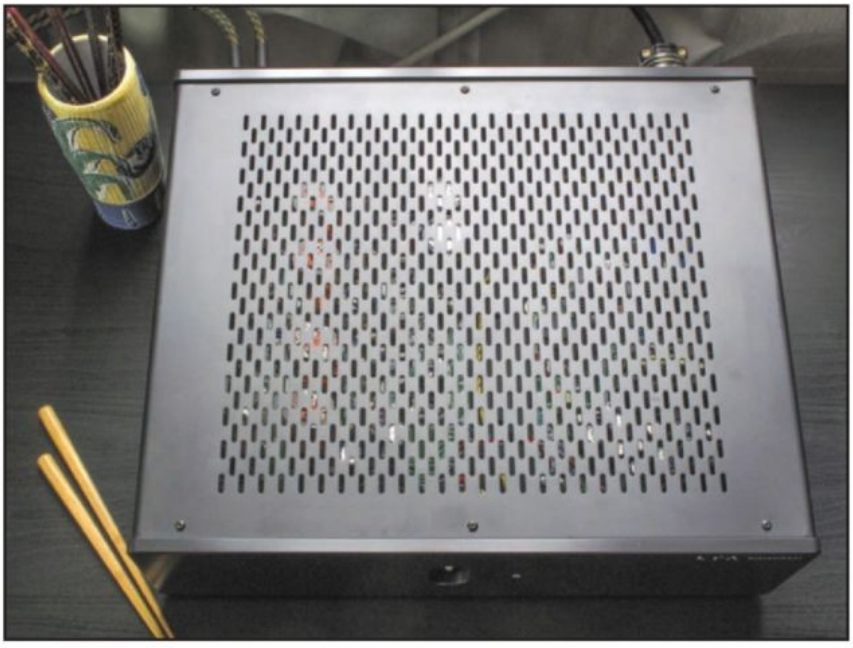


Photo 2: Fern and Roby Audio designed the casework, featuring front and rear panels milled from slabs of aluminum, Cardas RCAs, WBT copper binding posts, front panel industrial toggle switch, and a stereo mono switch with all markings laser engraved.

Initial Impressions

Let's talk about the result of his and his team's efforts...The ZOTL Ultralinear is clad in an unpretentious, almost matte black exterior. The case is bent and perforated sheet aluminum, designed and manufactured by the mischievous and equally renegade elves at Fern and Roby Audio (see **Photo 2**). The machined aluminum front panel presents a Zen cleanliness; only a white LED pilot light, and centrally located, inset bat-handled power toggle interrupts the dark satin plane. The back panel is also equally spartan, with an IEC power receptacle, a set of WBT copper binding posts, Cardas RCA inputs, and a mono/stereo switch (see **Photo 3**).

In the same manner, the shipping container is unassuming—simple rigid foam slabs cradling the contents. Frankly, I find the bling so prevalent nowadays in audiophilia more than a bit offensive. LTA bucks that trend, with its Spinal Tap "blacker than black" casework. LTA's gear uses dollars where they count, not on overspec'd unobtanium components and billet heatsinks requiring a day's output from bauxite mines "Down Under." Rather, money goes toward thoughtful design and carefully

Photo 3: The back panel is also equally spartan, with an IEC power receptacle, a set of WBT copper binding posts, Cardas RCA inputs, and a mono/stereo switch.

selected, appropriate parts. Schneider mentioned they do use copper "...as much as possible. Copper lead resistors, copper lead capacitors, pure copper wire, no steel leads." Steel, according to Berning, is undesirable for audio.

The front and rear panel are each CNC milled from a slab of aluminum. That low-luster black livery is a special anodizing that results in a flat finish that is extremely hardwearing. All exterior lettering is laser engraved which, according to Schneider, "...is more durable than silkscreened paint." The ZOTL Ultralinear is an all American product. Though individual components are sourced from around the world, Schneider stresses that all sub-assemblies are made "...in America by Americans, in Maryland, in a Takoma Park suburb of Washington, DC."

Earlier, I mentioned Berning's patent. It alludes to, at some length, an amplifier that can reproduce frequencies down to DC without overheating. Though the Ultralinear isn't DC coupled, it does go quite low, an octave below what's expected. Hollow state usually has issues with very low frequencies, so I asked Schneider about what ZOTL brings to the table for very low frequency (VLF) performance. As expected, the biggest barrier to VLF is the output transformer found in traditional designs. Remove that obstacle and the result is a deep diving low end.

His ZOTL products are a half of a decibel down at 8 Hz. "An output transformer might have to be the size of a coffee table to go down that far. The other main reason that it's perfectly linear at VLF is that the amplification is done at the carrier frequency, where it is perfectly linear. All devices get nonlinear at their boundary condition ...bass frequencies are considered a boundary condition, as they represent the lowest 0.5% of the audible audio range. So, even solid-state devices are going to be nonlinear at VLF. The ZOTL does all the heavy lifting at 250 kHz."

Inside the Unit

There are only four tubes in the entire unit, a pair of 17JN6 Compactron beam tetrodes for the impedance converters, with an input section of commonplace 12AT7 and 12AX7s. The Ultralinear develops 20 W per channel into 8 Ω of Class AB push pull power, with double that in mono mode. I'm a fan of thoughtfully applied feedback, rather than the "Iron Fist of Stability" school. In the case of the Ultralinear, the feedback is local to the power tube, as opposed to being tied back to the first stage as

Switch It Up

David W. Berning's 1997 patent number 5,612,646 opens by discussing the downside of conventional hollow-state designs, then digs into the patentable innovations. His formulation "... includes a push-pull pair of vacuum tubes operating in a linear amplification mode coupled through a pair of DC-DC switching power converters to an external load impedance."

The heart of the ZOTL design is the switched mode impedance converter. A ZOTL output transformerless amplifier impedance matching circuit has, "...at least one switching power converter bridge having opposite terminals connected between a fixed source of voltage and the load. Two bridges and two power supplies are used for bi-polar or push-pull applications. The returns for the power supplies are connected to the return for the load. The other two terminals of a given bridge are connected across the primary winding of a conversion transformer. The four switching transistors that make up the legs of the bridge are alternately switched on and off at a high rate of speed so that the voltage difference between the fixed voltage and the load voltage is impressed across the primary of the conversion transformer. This voltage difference is a relatively slowly changing voltage, and it is alternately impressed across the converter transformer in one polarity and then the other polarity at a rate determined by the gate drive for the transistors. One gate drive transformer can be used for driving all eight gates of a bi-polar two-bridge system. A gate drive frequency of 250 kHz is practical for desired amplifier bandwidths of 100 kHz assuming simple filtering is used to remove switching noise from the load. The full-wave nature of the circuit and the fact that it uses square wave switching leads to the bulk of the switching noise being high-order harmonics of 500 kHz, which can be effectively filtered with a second order L-C filter."

The patent is available on the interwebs, and makes for a fun and geeky read.

it is in conventional designs. Schneider conjectures that the Ultralinear's audible detail is a result of that design choice (see **Photo 4**).

The automatic bias scheme in the Ultralinear removes the requirement that the driver tubes be matched. One portion of the auto-bias is similar to a sample and hold circuit. It samples the current sum of the two tubes through their common cathode resistor and looks for minimum current. Using this information, the B+ on the driver stage is adjusted with a servo to force the current sum to within an acceptable range. The other portion involves monitoring DC offset. Since the output is direct coupled, the DC offset is balanced between the two power tubes. Any error, any offset, is applied back to the input so it is nulled. If it cannot, a signal is sent to the power supply to shut down.

The Sound

As with the only other OTL design I have subjective knowledge of—Atma-Sphere's S-30 Mk. 3.3—the ZOTL Ultralinear, frankly, makes most other tube designs in its price range sound broken. Dark or hooded, a bit flabby in the low end, distortion disguised as pretty—those are traits I don't consider worth the euphonic excitement that thermionic noise and inductive hysteresis lends. When I first powered it up and started to listen, I was disappointed into thinking that this was sounding like a budget hollow state. A vague sort of lo-fi sound was what I was hearing when compared to my reference amp, a Class A, solid-state Lindell AMPX. However, in very short order, the sound jelled into a lovely clear and wide presentation. Not gratuitous, mind you, but fluid and involving.

Remember the RCA inputs I mentioned earlier? My rig is balanced, from DAC to amp, straight through with nothing in between except wire. For this review, I wanted to keep all participants on as even a playing field as possible. I don't own RCA-to-RCA interconnects, except the type that came bundled with analog TVs back in the bad old days. So, I reached out to the folks at Wireworld, who were kind enough to lend me a set of excellent, half-meter Gold Eclipse 8 interconnects. Since my reference amp is fed by a much longer 3 m of Soundstring's balanced GEN II Beta 2-22S cabling, I felt that the differences in wiring wouldn't significantly skew the sound quality in favor of a particular party.

The Ultralinear is profoundly pleasing and smooth, engaging the listener in a special way. I had the pleasure of spending considerable time with my review unit, switching between two impressive, high-quality open baffle speakers—the Black Monolith (a.k.a. Spatial Audio's $3,995 M3 Triode Master), and Pure Audio Project's $4,790 Trio15 Heil. The M3 Triode Master is a 16 Ω model, while the big white Trio15 Heils I had ranged from 3.5 to 8 Ω across the audio passband. I mention this because uniform load impedance is critical for best performance, especially with the ZOTL. Subjectively, the LTA Ultralinear is more gelled, in that thermionic glue sense, than my reference Class A solid-state amp, a 40 W AMPX ($1,600). The lower two octaves are subjectively leaner, though the ZOTL delivers flat response down below 20 Hz. The upper bass is noticeably less prominent compared to my reference. The midrange is more restrained, less "in your face." The Ultralinear offers a midrange appropriately scaled relative to the frequency extremes. Being about 10′ away from rather efficient speakers meant I didn't desire a tremendous amount of power and, with a high-impedance oscilloscope probe hanging off one channel, I didn't get even close to clipping before I exited my loudness comfort zone.

Overall, I think the Ultralinear is less honest in a strict sense, but also less analytical. I would think that relaxed quality would pair nicely with speakers that have a more aggressive high end, such as those from Joseph Audio, Von Schweikert Audio, or KRK, though I didn't have access to any of those during my review period. While listening to everything from Peter Gabriel's modernist, Glassy instrumental "San Jacinto" (*Real World* 1982) to the State Academic Symphony Orchestra of Russia Evgeny Svetlanov's rendition of Pyotr Ilyich Tchaikovsky's "The Sleeping Beauty," Op. 66 (International Classical Artist 2018), every little nuance of each recording was there, without popping out to draw away attention to the flow.

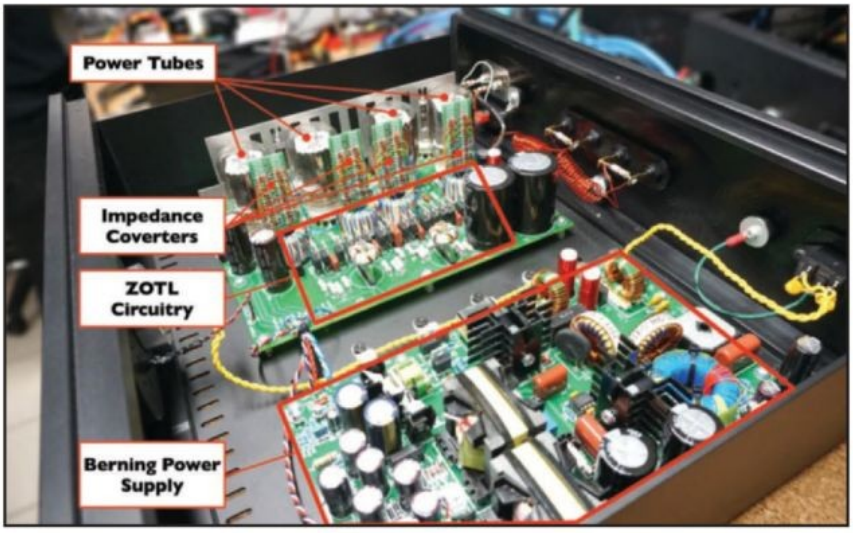


Photo 4: The LTA Ultralinear outputs 20 W per channel using 17JN6 tubes, a Compactron-style TV sweep tube that can't be easily used in traditional transformer-coupled amplifier designs. The power supply is an advanced switching supply optimized for audio that only David Berning deploys. All the benefits of ZOTL technology, including three times the tube life, one-third the heat, a super wide frequency response of 8 Hz to 60 kHz, and an under 2 Ω output. There is also a sophisticated speaker protection system to prevent damage to the amplifier and speakers, if there ever was a tube issue.

About the Author
Oliver A. Masciarotte has spent more 30 years immersed in the tech space, working on facilitation, optimization, marketing, and product development for clients worldwide. As an author and speaker, he enjoys informing folks about technological best practices. More information is available at seneschal.net and othermunday.com.

Getting a Second Opinion

As I often do, I invited a fellow enthusiast over to the house to gauge his reactions. Using my exaSound e22 MkII DAC fed from Amarra 4 as the level-matched source and the Trio15 Heils, we used TIDAL streaming so we could bring up songs as they came to mind. A few examples...George Winston's stark solo piano was a good benchmark for my friend Michael Newman. Listening to George Winston's "Joy" from *December* (44.1 kHz; Windham Hill 1990): "With the AMPX, the hammers hitting the strings seem more distinct, the room and reverb is more easily perceived. The LTA's transients are looser, they seem to lack precision. I'm missing the style of George's playing somehow."

Another oldie but goody we streamed was Cat Stevens' "Longer Boats" from *Tea For the Tillerman* (96 kHz; A&M 2000, first MQA unfolding in Amarra). Michael opined that the Ultralinear exhibited, "...a nice personality, and works to this album's advantage. It masks some of the flaws in the recording, especially the overdubbing." Thinking about the significant shifts in timbre, I had to think the variations heard between my AMPX and the LTA amp were largely caused by the "frequency response variations" that Yaniger mentioned, which pointed to the load as the culprit. So, I switched to Spatial's more controlled impedance M3 Triode Master and, as expected, the voicing shifts evaporated. Moral of the story: choose your speaker complement with care, as is the case with any hollow-state amp.

More Comparisons

I was curious how LTA products differ or depart, design-wise, from Berning's own models, especially the ZH-230. I knew that the addition of Stillpoints and a fancy PCB add to the cost, but what are the fundamental differences and why did LTA go that route? Schneider reminded me that the electronics are pure David Berning. "The ZOTL40 and the UL are actually the ZH230 electronics adapted to EL-34s in the case of the ZOTL40 and to 17JN6s in the case of the UL," Schneider told me via e-mail. "Everything David learned over the 5 to 7 years the ZH-230 was in production was baked into both the 40 and the UL. The protection circuitry is improved from the ZH-230. The ZH-230 used 33JN6s, the same family as the 17JN6, but it was strictly triode mode."

Schneider also said they do use "fancy" PCB material in some of their products, but only where it makes a positive audible difference. As to externals that the customer can see, Schneider revealed that they tried a "...less expensive case initially, and now we use similar cases that David uses. Regarding the Stillpoints, let's just say David says he puts them on because people think they need them. Whether it makes any difference is still a topic for discussion. Our whole approach is to bring David's designs to market at a much lower cost. (He) goes through every one of our amplifiers before they are shipped to make sure he is happy with the sound and the workmanship." Schneider added that customers can always add their own preferred aftermarket vibration control to the product.

My "Battle of the Baffles" resulted in some interesting contrasts, which I'll discuss in my upcoming review of the Trio15 Heil, but I found the combination of ZOTL Ultralinear and M3 Triode Master was a magical match. With the Trio15 Heils, basic EQ largely evened out the variations. To my ears, a broad parametric lift at 11.9 kHz, +3 dB and a Q of 1, subjectively brought in the snap my friend Michael had been looking for, while a +4.5 dB first-order shelf starting at 98 Hz simulated the low end weight the AMPX provided. I heard far better sound from these relatively budget pairings, no exaggeration, than I have from systems costing ten times as much. Though the ZOTL UL can't be all things to all audiophiles, it may just be the last amp most of us would ever need. Rugged and no fuss, with a weight that won't herniate your back and a price that won't have your significant other chasing you with a rolling pin, Linear Tube Audio's ZOTL Ultralinear sets the bar for rich, sublime musical goodness.

Music Resources

Partial list of tracks in heavy rotation during this review include:
Les Amazones d'Afrique's Dombolo—Listen and You'll See: 10 Years of Society of Sound (B&W/Real World 2018) Rockin' muscular music that veers away from the tinkly, guitar-centric norm

Mitski Miyawaki—*Puberty 2* (Dead Oceans 2016) At times tuneful, with intrusions of jagged grrrl power, Mitsky crafts a lyrical, sly, compelling document

Sons Of Kemet—*Your Queen Is A Reptile* (Impulse! 2018) U.K. jazzers deliver a big soul sound

JD McPherson—*Undivided Heart & Soul* (New West Records 2017) A toe-tapping HRA excursion into rock, pop and occasional rockabilly that keeps me coming back for repeat visits

Anbessa Orchestra—*Negestat* (Bandcamp 2018) Heavy, horn-driven North African rock from this New York-based septet with Budos Band influences

Childish Gambino—*Awaken, My Love!* (Glassnote 2016) Donald Glover's alter ego displays the complex, street-wise musical outlet that's one juicy slice of his expansive artistic talent

Kacey Musgraves—*Same Trailer Different Park* (Mercury Nashville 2013) Snarky, sentimental lyrics combined with poppy hooks and Musgraves' distinctive "Every Gal" delivery makes this modern country platter a classic

The Linear Tube Audio's ZOTL Ultralinear Power Amplifier

Measurements

By
Stuart Yaniger
(United States)

Over the years, I've designed, built, and measured hundreds of tube amplifiers, so I have (I think) reasonable expectations for the genre. And likewise, prejudices based on experience! The amps designed by David Berning always challenge those prejudices, and the Linear Tube Audio (LTA) Ultralinear is no exception.

Two things stand out right from the start. First, this amp is rather lightweight. If not for the fairly robust chassis, it could be swung around by a child, in great contrast to typical tube power amps. Second, the vast majority of tube amps have output transformers with multiple secondary taps for different impedance loudspeakers, the idea being to optimize the impedance seen by the output stage. In a sense, this is a bit futile because most speakers have impedance that varies greatly with frequency, so choosing the best tap requires some experimentation (and usually trade-offs). The LTA amp has but a single set of speaker binding posts. At least this makes the measurement regime simpler!

A few notes before diving into the measurements. The measurements were performed prior to Oliver Masciarotte's listening sessions and the results were not disclosed to him in advance. All measurements were taken using an Audio Precision APx525 analyzer on generous loan from the kind folks at Audio Precision (AP).

The Measurements

First, let's summarize the basics. All measurements for this table were taken at 1 W (2.83 V into 8 Ω and 2 V into 4 Ω) and 1 kHz. Total harmonic distortion plus noise (THD+N) was measured at a 22 kHz bandwidth, unweighted—because of the carrier frequencies characteristic of this amplifier, the distortion measurements at greater bandwidths were significantly higher due to ultrasonic noise (see **Table 1**).

Frequency response was measured with both a 4 Ω resistor and a mini-monitor loudspeaker and is shown in **Figure 1**. Because of the relatively high source impedance, there's broadband frequency response alterations of 1 dB or so with the speaker,

Load	Channel	Gain	THD+N
8 Ω	L	21.7 dB	0.09%
8 Ω	R	21.4 dB	0.07%
4 Ω	L	20.6 dB	0.15%
4 Ω	R	20.1 dB	0.07%

Table 1: All measurements for this table were taken at 1 W (2.83 V into 8 Ω and 2 V into 4 Ω) and 1 kHz.

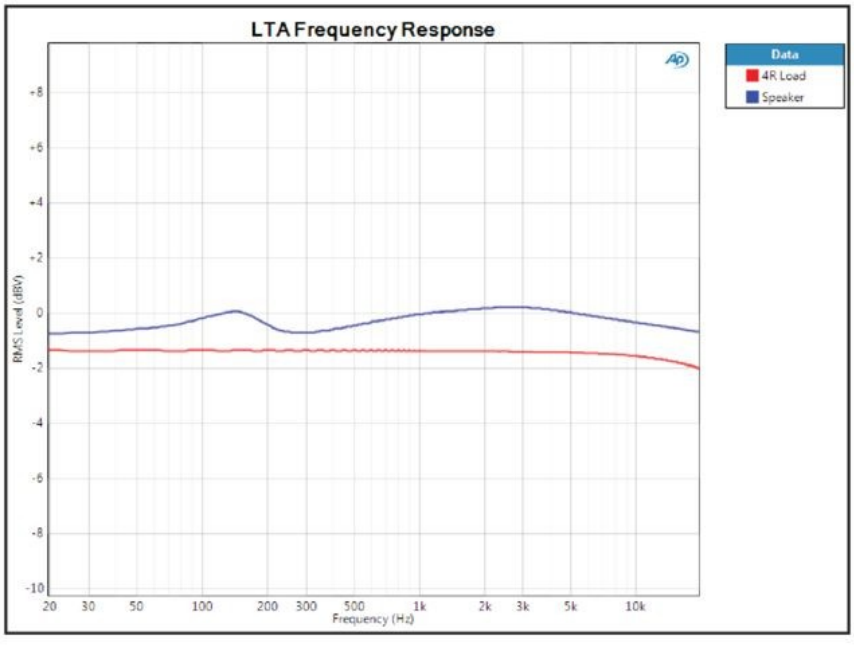


Figure 1: Frequency response was measured with both a 4 Ω resistor and a mini-monitor loudspeaker.

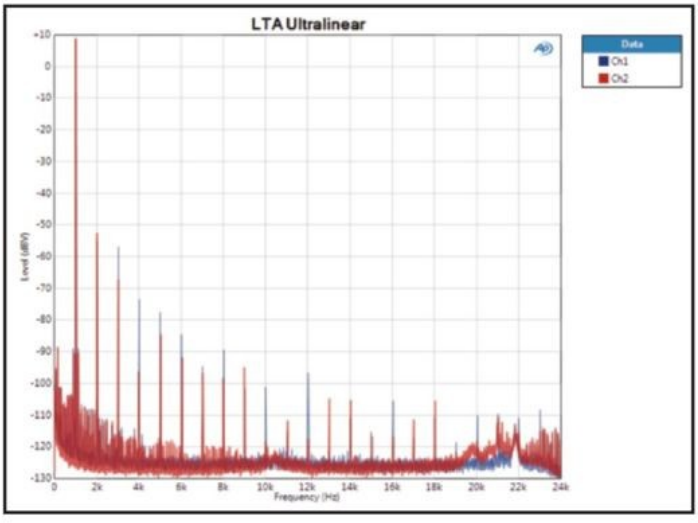


Figure 2: This is the distortion spectrum at 1 W into 8 Ω at 1 kHz.

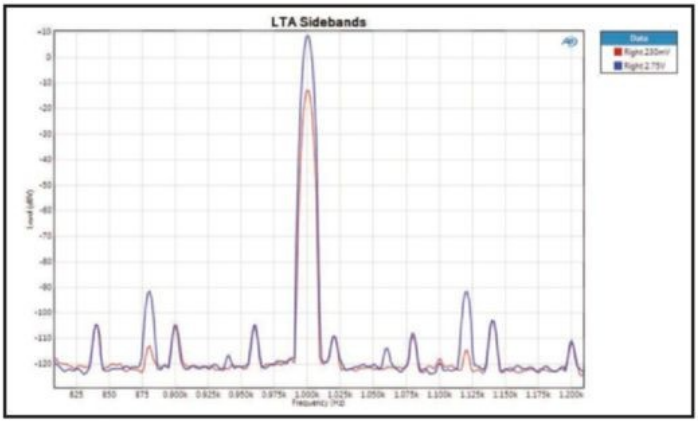


Figure 3: Here is an enlarged view around the fundamental frequency, with measurements at two different voltage levels (230 mV and 2.75 V out).

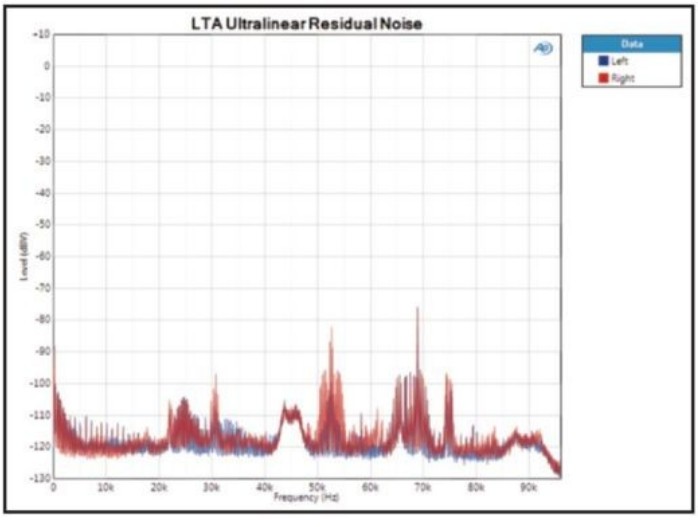


Figure 4: This graph shows the noise spectrum with no signal applied.

so there will certainly be a sonic difference between this amp and a low-source impedance solid-state amp in direct comparison.

The distortion spectrum at 1 W into 8 Ω at 1 kHz is shown in **Figure 2**. Second and third order are, as expected, the highest, but there's a spray of harmonics reaching well into the teens. In a log frequency plot (not shown), you can also see power supply noise at 60 Hz and its harmonics, with the largest spike at 120 Hz, about -96 dB below the fundamental. Interestingly, the 1 kHz fundamental has sidebands at 120 Hz above and below it and the harmonics appear to show the same behavior. This is often

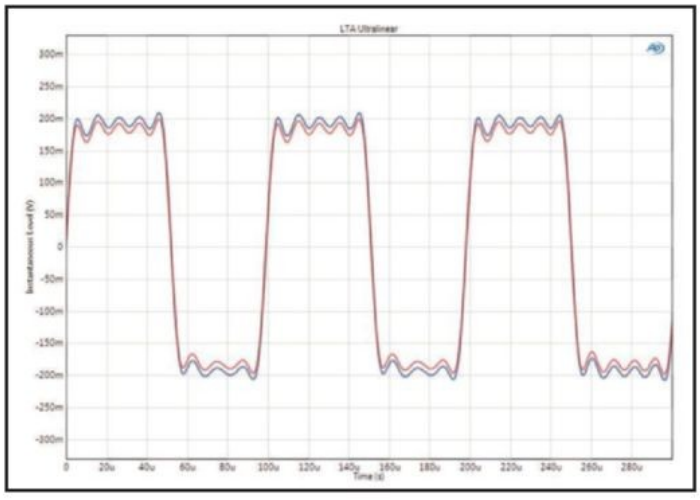


Figure 5: This is the amplifier's response to a 10 kHz square wave.

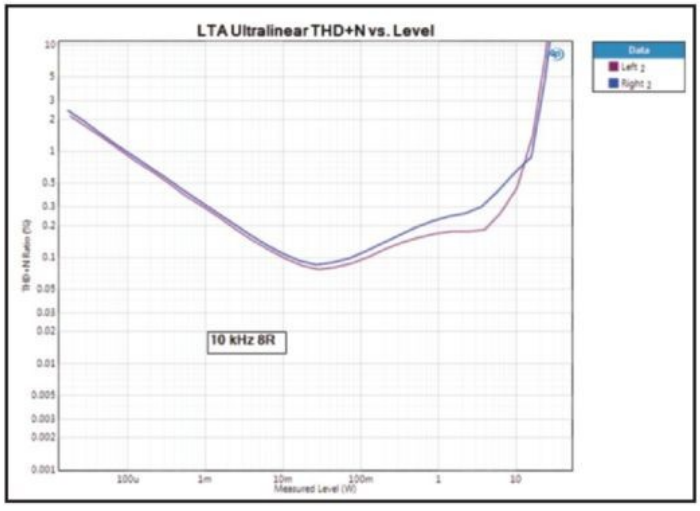


Figure 6: In this figure and the next plot of distortion vs. power, you can see that the distortion remains about equally low at 10 kHz and 20 Hz.

About the Author

Stuart Yaniger has been designing and building audio equipment for nearly half a century, and currently works as a technical director for a large industrial company. His professional research interests have spanned theoretical physics, electronics, chemistry, spectroscopy, aerospace, biology, and sensory science. One day, he will figure out what he would like to be when he grows up.

seen in conventional amps near the overload point when their power supplies are under-designed, but in this case, the main power supplies are switchers and the amp is only putting out 1 W. **Figure 3** shows an enlarged view around the fundamental frequency, with measurements at two different voltage levels (230 mV and 2.75 V out). The sidebands scale with the tone (i.e., when the tone amplitude is increased, the sidebands do as well). So there seems to be some sort of coupling issue between the supplies and a full wave supply used somewhere in the circuit—no schematic was provided, so I can't really speculate any further as to the cause. But Figure 3 does demonstrate that the coupling is some sort of signal dependent intermodulation and not a constant.

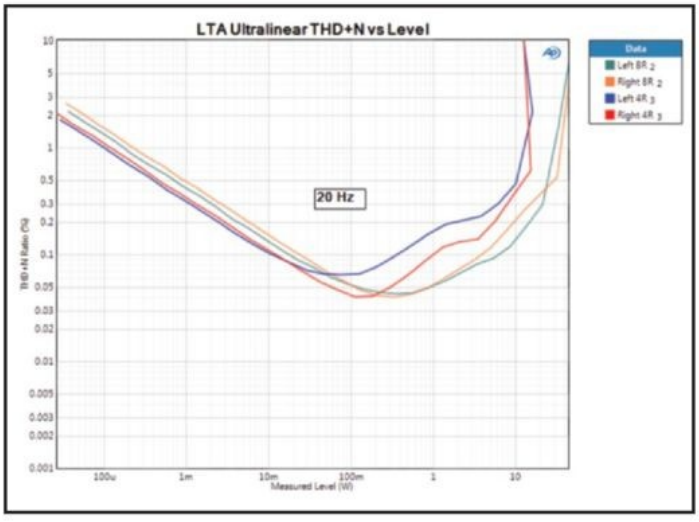


Figure 7: The amp is distinctly less happy with a lower impedance load, but the distortion is still low (for a low feedback tube amp).

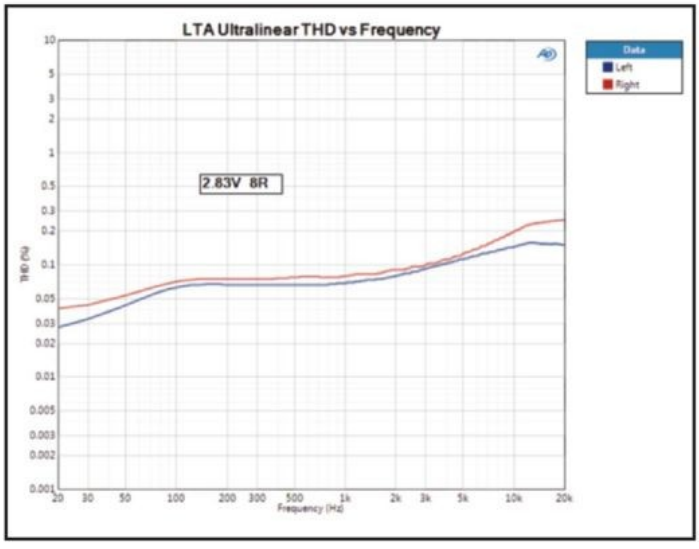


Figure 8: At 1 W output, THD+N is shown as a function of frequency.

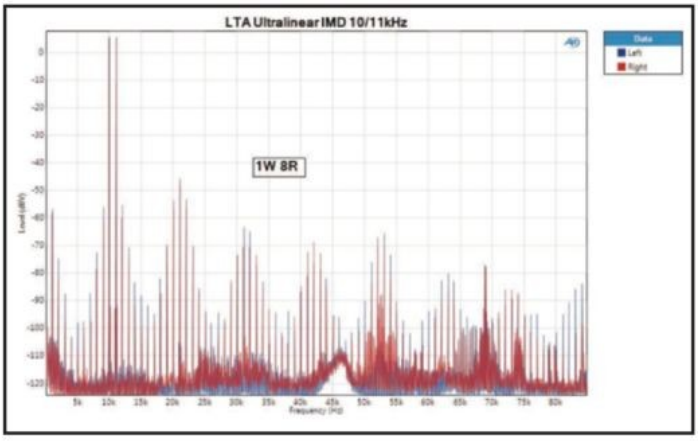


Figure 9: The intermodulation behavior is shown when the amp is fed a pair of tones at 10 kHz and 11 kHz, at 1 W output into 8 Ω.

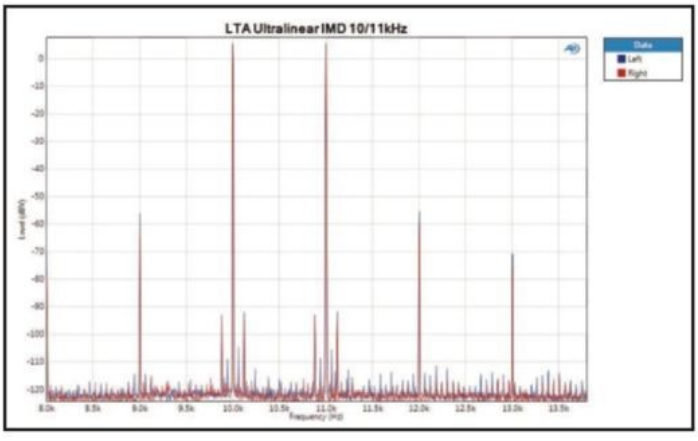


Figure 10: Every intermodulation tone has the 120 Hz sidebands as before, making the spectrum rather messy.

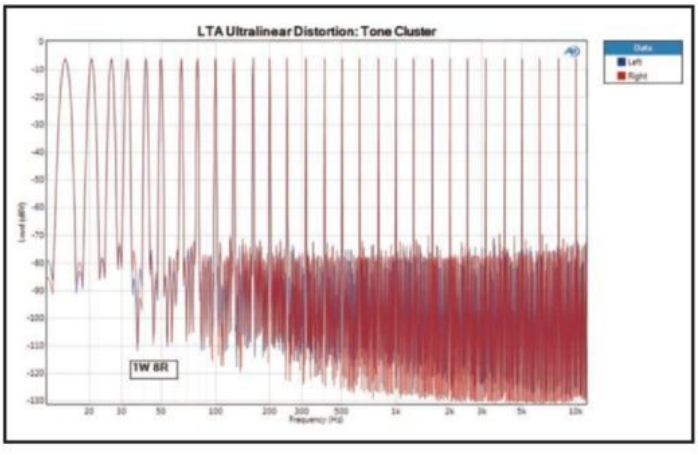


Figure 11: Here is an illustration of the consequence of this odd distortion behavior, which is the spectrum of a 31 tone cluster, again at 1 W into 8 Ω.

Note also that each harmonic of the fundamental tone also shows these sidebands—this has some interesting consequences, which we will shortly see.

Figure 4 shows the noise spectrum with no signal applied. It's easy to see why bandwidth limiting greatly reduced the THD+N; the "N" part is relatively high, with the biggest peaks being at 52 kHz and 69 kHz. Not likely high enough in amplitude to be any danger to your tweeters, but your cat might find your speakers to be curious objects while the amp is on.

Figure 5 shows the response to a 10 kHz square wave. Because of sampling rate limitations, there's ripple on the top and bottom, which can be ignored. What can't be ignored is the excellent rise and fall time. The elimination of the output transformer removes a major limitation on high frequency extension. We would also expect it to remove a major limitation on the low end as well. **Figure 6** and **Figure 7** bear this out. Conventional tube amps with output transformers get a bit antsy at bass frequencies, but in these plots of distortion versus power, you can see the distortion remaining more or less equally low at 10 kHz and 20 Hz. The amp does runs out of steam a little earlier at 20 Hz, about 15 W into 8 Ω. You can also see from Figure 7 that the amp is distinctly less happy with a lower impedance load, but the distortion is still low (for a low feedback tube amp) right up until overload.

At 1 W output, **Figure 8** shows THD+N as a function of frequency. Again, the lack of a conventional output transformer keeps bass distortion low. The rise at higher frequencies is likely due to a reduction in open loop gain and consequently, a reduction in feedback. Nonetheless, it is still relatively low, again showing a possible advantage of Berning's impedance conversion over conventional output transformers.

Things start looking a bit worse when we feed the amp more complex signals. **Figure 9** shows the intermodulation behavior when the amp is fed a pair of tones at 10 kHz and 11 kHz, at 1 W output into 8 Ω. Not only are there relatively high intermodulation products, but **Figure 10** shows that every intermodulation tone has the 120 Hz sidebands as before, making the spectrum rather messy. The left channel also shows sidebands at 60 Hz above and below each tone. This will only get worse at higher powers and less ideal loads (e.g., actual loudspeakers). An illustration of the consequence of this odd distortion behavior is shown in **Figure 11** shows the spectrum of a 31 tone cluster, again at 1 W into 8 Ω. Ideally, one would see nothing between the tones down to the noise floor. Clearly, the amp's behavior is not ideal! The errors are, as before, those of addition, not omission.

Wrap Up

The measurements are certainly a mixed bag. The single tone distortion is relatively low for the genre and the bandwidth is excellent, showing the advantage of the virtual output transformer over a hunk of iron. But the odd sideband behavior and high intermodulation suggest that with complex musical signals, the amplifier could very well audibly color the sound. At its price, despite some really clever and innovative circuit design, it is for my tastes more a work of performance art than of performance.

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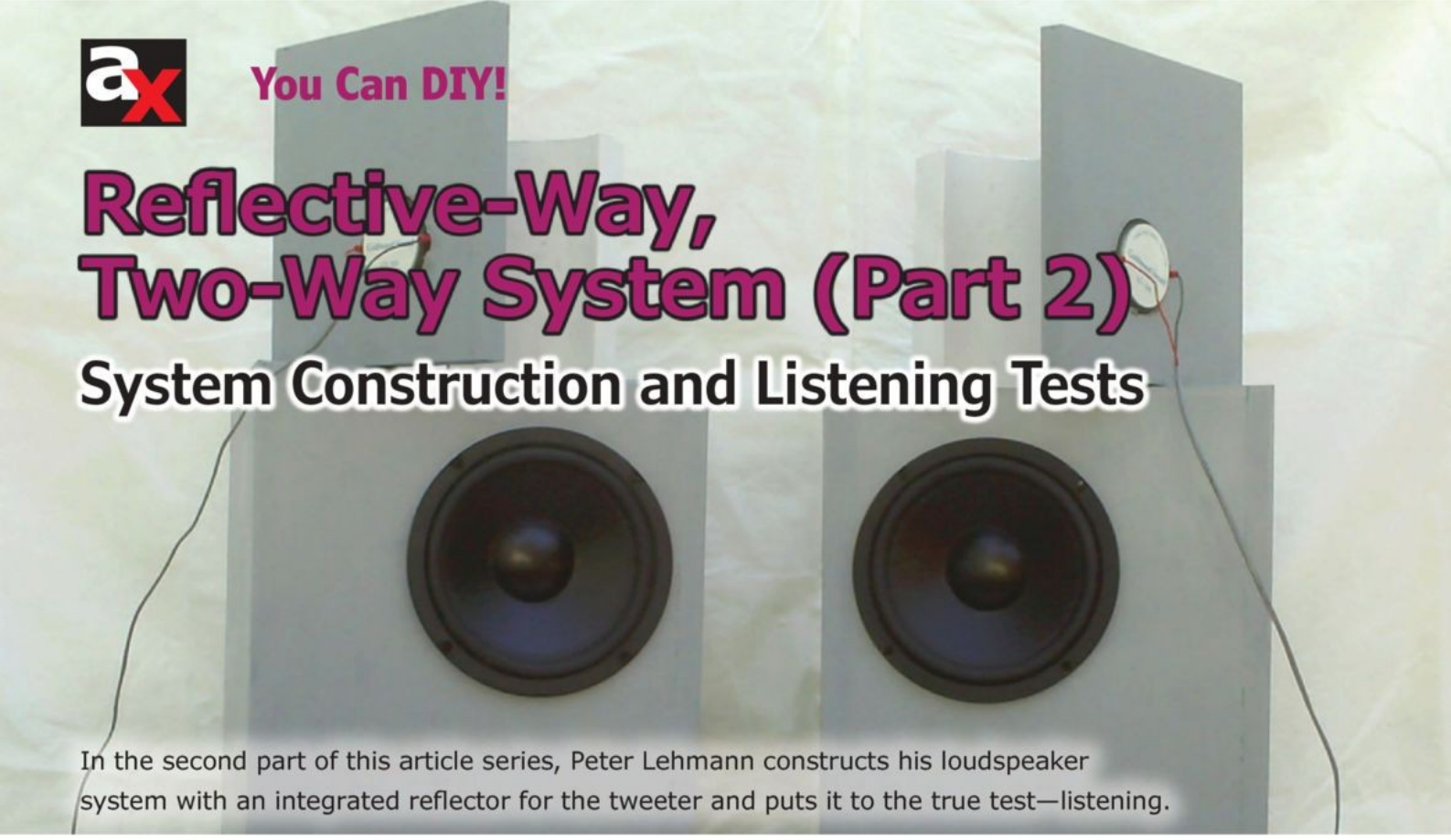
Reflective-Way, Two-Way System (Part 2)

System Construction and Listening Tests



In the second part of this article series, Peter Lehmann constructs his loudspeaker system with an integrated reflector for the tweeter and puts it to the true test—listening.

By
Peter Lehmann
(United States)

In the second part of this article, I am going to start with the construction and theoretical reflectivity of the nearby reflector.

The reflector design shown in **Figure 1a** is constructed from plaster of Paris (POP) "spiked" with a relatively smaller amount of Portland cement. Making it from POP is a good choice as doing so is fairly low-tech and a plaster surface has good high-frequency reflectivity. The Portland cement was added to the plaster to make the reflector less susceptible to damage, however, I discovered that adding too much cement caused the mixture to set very quickly and not accurately conform to the mold's shape. The ratio by volume of POP to Portland cement that I settled on was 10:1.

The radius of the reflector's concavity shown in Figure 1a, 3-7/16", is that of the equivalent larger radius circle that approximates the gentler curvature of the ellipse discussed in Part 1 of this article (*audioXpress*, November 2018). An equation for calculating this radius R_{LG} has been previously given. The concavity of the POP reflector encompasses 90°. The height of the POP reflector is made equal to twice the height of the center of the circular opening for the tweeter in the flat baffle design shown in **Figure 1b**, above the bottom edge of the baffle.

Ideally, reflectivity loss of the POP reflector should not be more than 5 dB. This follows from the fact that sensitivity (1 W/1 m) of the Newark No. 55-1190 woofer and the Goldwood GT-525 tweeter equal, respectively, 87 dB and 92 dB. Too bright of reproduction can be much more easily compensated for than reproduction that has too much bass.

From a simplified method of analysis of reflectivity losses by Jens Rindel (see Resources), components of the total loss ΔL are:

$$\Delta L = \Delta La + \Delta Lm + \Delta Lk + \Delta Ls$$

where the source of the losses are:

ΔLa: distance
ΔLm: absorption (material)
ΔLk: curvature
ΔLs: reflector size

ΔLa is the loss that occurs when the summation of the distances from a sound source to a reflective surface, and from the reflective surface to a receiver (listener) is substantially greater than the distance from the sound source directly to the receiver. This does not apply to a listener at a normal distance of not less than about 6' from the tweeter/reflector of my loudspeaker system as the POP reflector is at a relatively very short distance from the tweeter. In the loss calculation here, ΔLa equals 0 dB.

From a table of random-incidence absorption coefficients (α) titled "Lightweight constructions

and linings," the average α for plasterboard in the frequency range of 2 kHz to 8 kHz equals 0.05. This value of α in the table is actually for two layers of 13 mm plasterboard secured in a very specific manner and painted. The best that I could do was to hope that absorption of my POP reflector would not substantially differ from this measurement.

The loss in the reflected sound wave caused by absorption of the reflector in decibels can be calculated with the equation:

$$\Delta Lm = I_R/I_I = 10\log_{10}(1 - \alpha)$$

where I_I and I_R equal one-sided intensity of respectively incident and reflected sound. From this equation given that α equals 0.05 it follows that ΔLm equals −0.22 dB.

Solving loss due to curvature, ΔLK, the equation given by Rindel for this loss can be simplified based on the fact that the distance from the tweeter to the POP reflector is much less than the distance from the tweeter to the listener. The simplified equation is:

$$\Delta L_K = -10\log_{10}\left|\frac{1+2\alpha 1}{R\cos\theta}\right|$$

where:

a1 = distance from the tweeter to the point of reflection
R = radius of curvature RLG
θ = angle of sound ray with respect to the radius from Cr to the reflection point

For **Figure 2** (originally shown as Figure 7 in Part 1 of this article, *audioXpress* November 2018), the distance of the reflection point from the radiating center of the tweeter and angle of reflection of each of the six sound rays shown is different for each of the six rays. Calculating ΔL_K for each of the six rays and taking the average, the average value of ΔL_K equals −2.5 dB. The maximum difference between loss for each of the six rays is that between rays 4 and 6 equal to 3.4 dB.

Rindel's equation for loss due to size ΔL_S is simplified in the same way that his equation for ΔL_K above has been. The simplified equation is:

$$F_G = \frac{C\alpha_1}{S\cos\theta}$$

where:

Frequency > F_G implies that loss due to size ΔL_S equals 0 dB

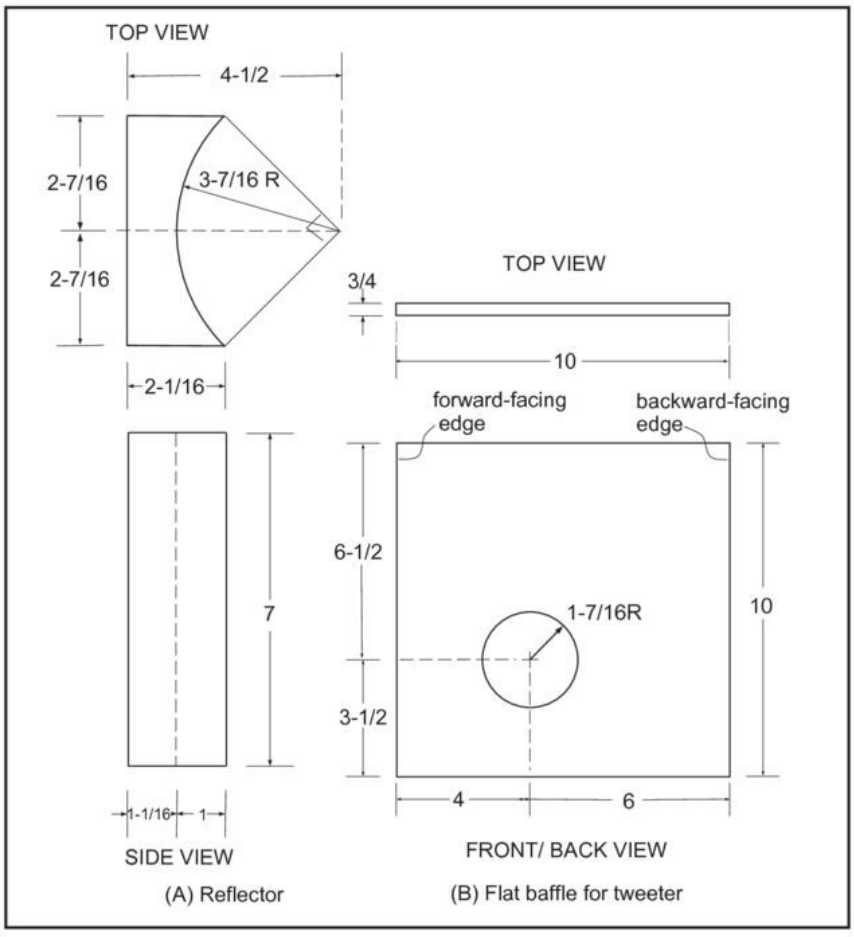


Figure 1: All dimensions are shown in inches. The nearby reflector (a) is molded from plaster of Paris and is identical for both the left and the right loudspeaker systems. The baffle board (b) of the tweeter is a square panel of 3/4" medium density fiberboard (MDF). The tweeter attached to the side of the baffle shown at (b) makes this the assembly for the left channel system.

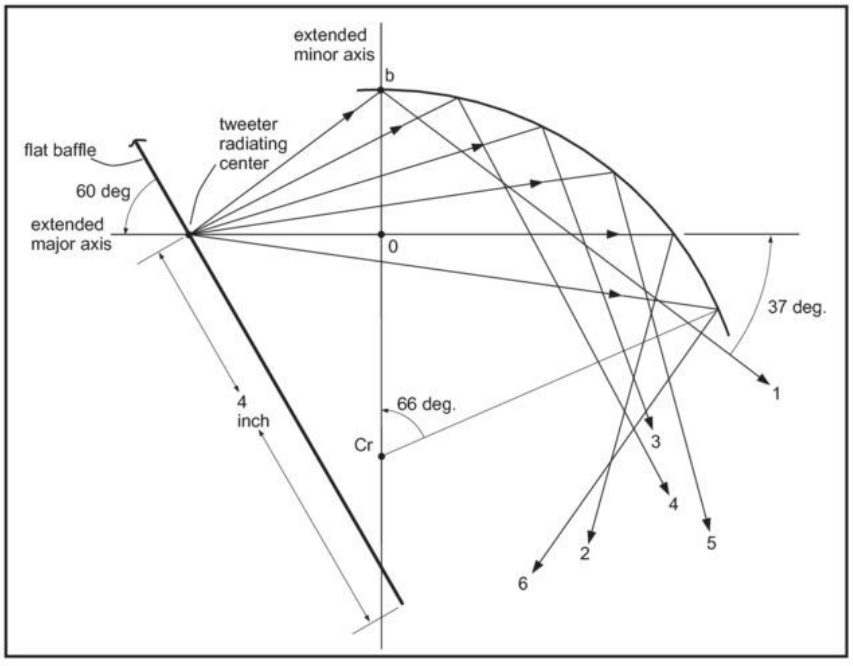


Figure 2: A reflector of circular curvature that closely approximates the gentler curvature of the ellipse shown in Figure 4 from Part 1 of the article (*audioXpress*, November 2018) is substituted for the elliptical reflector shown in Figure 6 from Part 1 of the article. The substitution results in none of the reflected rays at a very shallow angle to the major axis and the baffle for the tweeter at a 60° angle to the major axis can be 4" long below the major axis without obstructing reflections off of the nearby reflector.

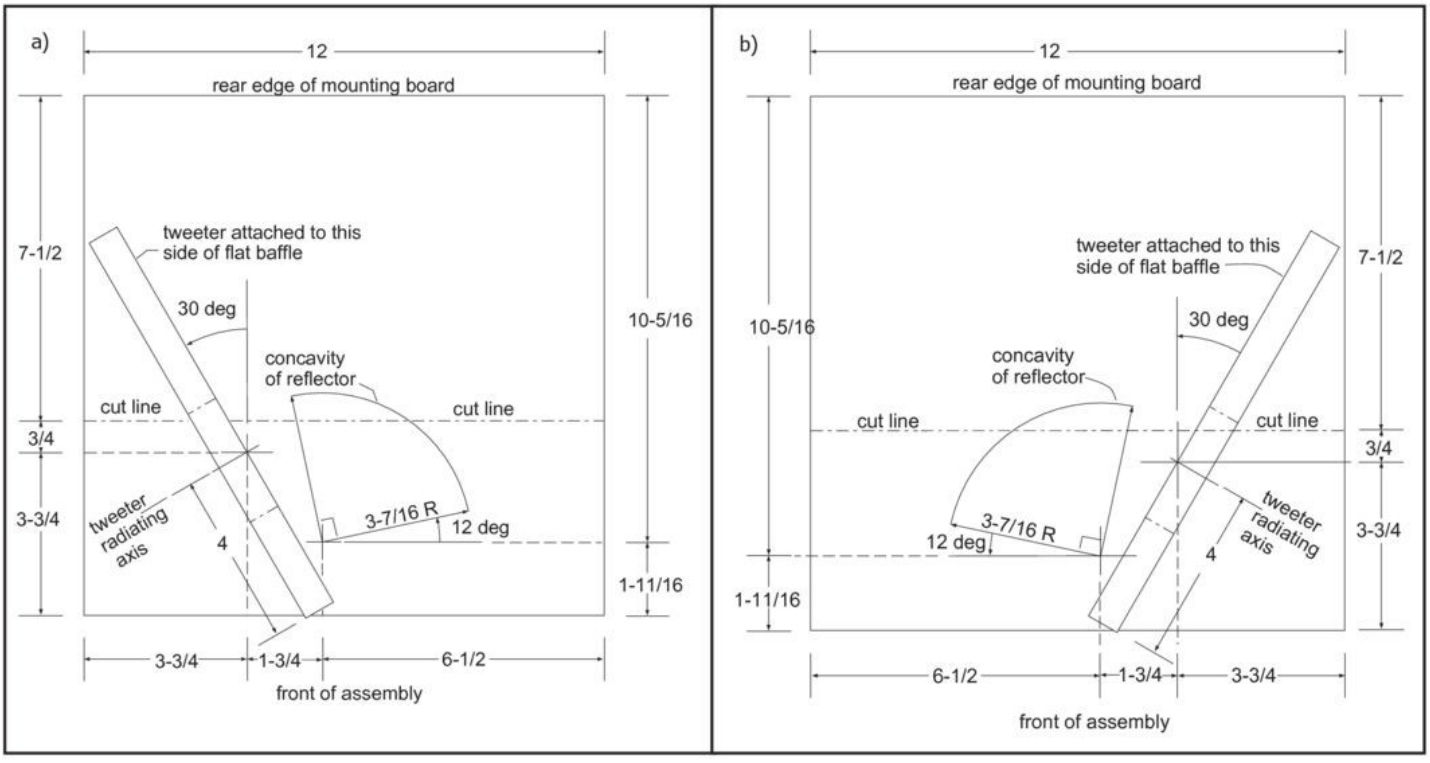


Figure 3a: Shown with dimensions in inches is the left channel assembly of baffle for the tweeter and POP reflector on a 12" square mounting board of 3/4" MDF. After tracing the layout of the figure on the mounting board and preliminarily attaching of the baffle and reflector to the board, depth of the mounting board is reduced to 7-1/2". Figure 3b: The arrangement of the baffle and reflector of the right channel system on a mounting board is the mirror image of that arrangement for the left channel system. Here the baffle for the tweeter is on the right-hand side.

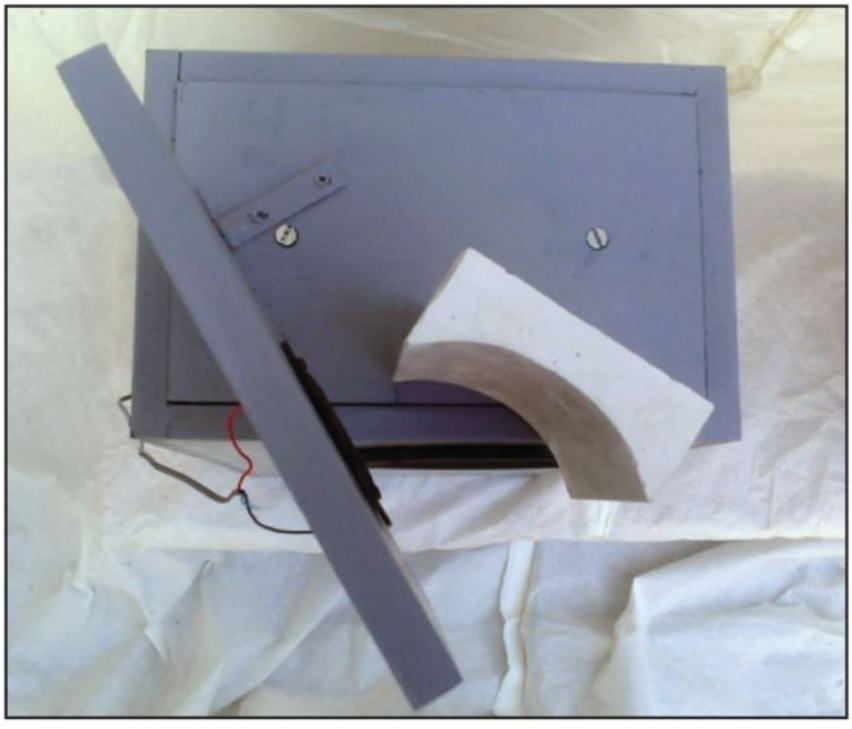


Photo 1: The assembly of the baffle for the left channel tweeter and POP reflector on a mounting board following Figure 3a is shown attached to the supporting cabinet. To avoid diffraction by the top panel of the cabinet, as shown here, the center of the tweeter is positioned directly above the front edge of the cabinet.

Frequency < F_G implies loss at the rate of 20 dB/decade as a function of the ratio of Frequency/F_G.
C = speed of sound
S = surface area of the reflective cylindrical concavity of the POP reflector
θ = angle of sound ray with respect to the radius intersecting the reflection point

As was done in calculating loss due to curvature ΔL_K, an average value of F_G must be found as the angle of incidence and distance a1 of the six rays shown in Figure 2 differ. For rays 1 and 4 found in Figure 2, F_G equals 1.3 kHz. For all of the other rays from that figure, F_G is very nearly equal to 2 kHz. The average value of F_G equals 1.7 kHz. Given that the crossover frequency of my loudspeaker system equals 2 kHz and the high-pass filter for the tweeter is second order, I settled on loss due to size ΔL_S equal to about –1 dB.

Finally totaling the losses:

$\Delta L = \Delta La + \Delta Lm + \Delta Lk + \Delta Ls$
$\Delta L = 0\ dB - 0.2\ dB - 2.5\ dB - 1\ dB = -3.7\ dB$

As sensitivity of the tweeter is 5 dB greater than that of the woofer, this calculation indicates that the

assembly design shown in Figure 9 from Part 1 of this article (*audioXpress*, November 2018) is acceptable.

Flat Baffle for the Tweeter

Making the flat baffle for the tweeter shown in Figure 1b to be 10″ square results in a baffle step frequency equal to 456 Hz. The crossover frequency is about 2 octaves above the step frequency, thus, this size baffle essentially limits radiation by the tweeter to into half-space only.

The opening for the tweeter made 3.5″ above the base of the flat baffle as shown in Figure 1b is a compromise in preventing diffraction by the top of the cabinet for the woofer that the flat baffle is

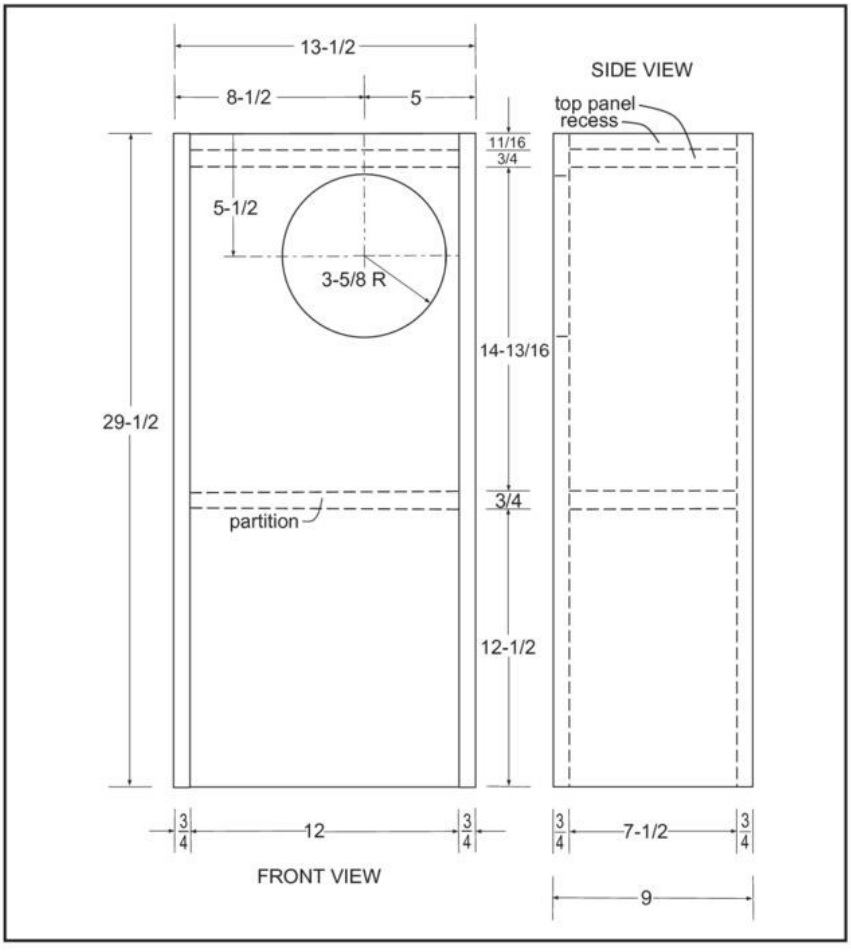


Figure 4: All the dimensions are inches. For the tower of my left channel loudspeaker system, the opening for mounting the woofer is offset to the right-hand side. The opening for the woofer of my right channel system is the mirror image of the left channel system shown here.

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Photo 2: With my stereo pair of loudspeaker systems correctly positioned laterally, the woofer and the tweeter of each are offset respectively to the inside and outside of each tower. In the pair that I built (shown here), the passive crossover network of each of the pair is installed in the enclosed volume beneath the closed-box for the woofer.

mounted on and not making the vertical distance between the radiation axes of the woofer and tweeter not excessively greater than one wavelength of the crossover frequency of 2 kHz.

The opening for mounting the tweeter made 4" in from the forward edge of the baffle is done because the tweeter/reflector assembly is attached to the loudspeaker cabinet with the radiation center of the tweeter horizontally directly at the top front edge of the cabinet. This is done to minimize diffraction by the top surface of the cabinet. Additionally, as shown at Figure 2, the width of the baffle obliquely forward of the major axis cannot be much greater than 4" without diffracting some of the sound from the tweeter reflected off of the nearby POP reflector.

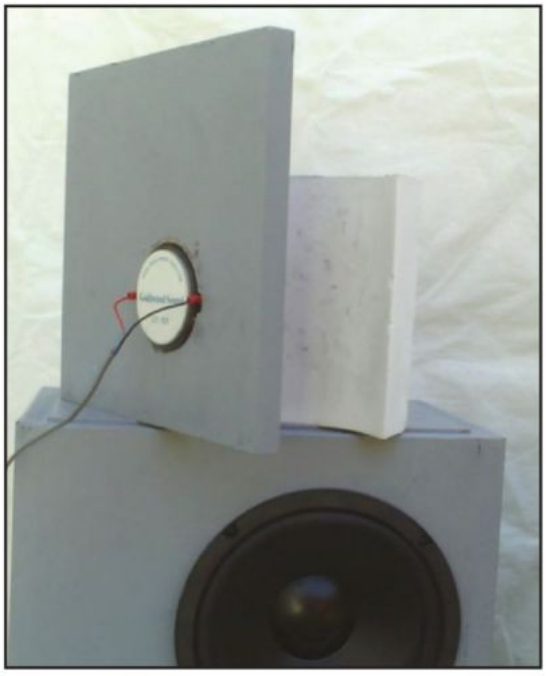

Photo 3: In the left channel system shown, the woofer is offset to the right-hand side. With the center of the woofer 5" in from the right side of the tower, this results in the center of the woofer directly underneath the vertical line on the reflector where sound from the tweeter is reflected laterally at a right angle to the woofer's baffle.

Tweeter/Reflector Assembly Attached to a Mounting Board

Figure 3a shows a top view of the assembly of the flat baffle (shown in Fig. 1b) for the left channel tweeter and nearby POP reflector (shown in Figure 1a) on a 12" square mounting board. **Figure 3b** is a top view of the assembly of the baffle for the right channel tweeter and reflector on a mounting board of the same dimensions. Both left and right channel assemblies are constructed in the same manner, except that the positioning of the baffle and the reflector of one is the mirror image of the other.

The final depth of the mounting board shown in Figure 3a and Figure 3b equals 7-1/2", or the depth behind the cut line. The board of larger depth is initially needed as the position of the center of the opening for the tweeter cut into the flat baffle and the radius center of the concavity of the reflector both lie forward of the final front edge of the mounting board.

The baffles shown in Figure 3a and Figure 3b are attached as follows. With the baffle held in the position shown in the figures, draw an outline of the bottom (mounting) edge of the baffle on the mounting board. Mark the centers of two mounting through-holes on the mounting board along the middle of the thickness of the baffle and behind the cut-line. Drill pilot holes and then countersinks of the diameter needed for inserting dowel centers at the pilot holes. Insert the dowel centers and place the baffle in the correct position pressing down to leave center marks in the bottom edge of the baffle. Attach the baffle to the mounting board with two drywall screws. Also a corner brace should be attached to the inside rear face of the baffle and mounting board.

With a compass, draw the outline of the concavity of the reflector in the positions shown in Figure 3a and Figure 3b. Place the reflector shown in Figure 1a with its concavity aligned with the arc drawn with the compass. Trace a pencil line around the base of the reflector. Mark the centers of two mounting through-holes on the mounting board within the outline of the base of the reflector behind the cut-line. As was done for the baffle, use dowel pins to mark the centers of pilot holes in the reflector. Attach the reflector to the mounting board with drywall screws.

Finally, detach the baffle and the reflector from the mounting board shown in Figure 3a and Figure 3b and reduce the depth of the mounting board to 7-1/2" by cutting off the material of the board forward of the cut-line. Reassemble. **Photo 1** shows the completed

final assembly of the design described in Figure 3a with the mounting board fitting into a recess of the cabinet for the woofer and held in place with machine screws.

Building the Towers

Figure 4 shows the plan, front and side views, of the tower providing a closed-box for the 8″ woofer of my system and supporting the tweeter/ reflector assemblies shown in a top view at Figure 3. The position of the woofer's opening shown is that for the left channel system. The opening for the woofer of the right channel system is the mirror image of that of the left channel system. That is, the center of the opening of the right channel system is 5″ in from the left-hand side of the tower. Otherwise, left and right towers are identical (see **Photo 2**).

The woofer of each of the tower pairs shown in Figure 4 is offset to one side so that the center of the woofer is directly beneath where the POP reflector above it reflects laterally at a perpendicular angle to the front panel of the tower. The upper front view of the left channel loudspeaker system is shown in **Photo 3**.

Ear Level and Elevation of the Driver-Pair

The tweeter/reflector assembly supported on top of the cabinet holding the woofer as shown in Photo 3 results in a greater vertical spacing between the woofer and tweeter than is desirable. Thus, preserving the compensation for the woofer's lagging response necessitates that the listener's ears are at a height that is close to midway between the two drivers. Given the listener's ears at that height, then, by reflection off of the POP reflector, sound from the tweeter to the listener travels an additional 4.5″ more than does sound from the woofer. The ideal ear level for the design of my loudspeaker system shown in Figure 1 and Figure 4 is 28.5″. This could be adjusted by altering the height of the towers.

Figure 5 is a demonstration of how not making an ear-level midway between the pair of drivers of a two-way loudspeaker system vertically 9″ apart can eliminate compensation for the lagging response of the system's woofer. The loudspeaker system shown in Figure 5 is a simplified design without a reflector for the tweeter. Here, the tweeter is simply positioned 4.5″ behind the mounting surface of the front panel or baffle holding the woofer.

As shown at Figure 5a, given ear level equal to 37″ above the floor and the radiation axis of the woofer 32.5″ above the floor, then the additional distance traveled by ray 1 from the tweeter relative to ray 2 from the woofer equals 2.24″ and compensation for the woofer's lagging response is preserved. In Figure 5b, the height of the woofer's radiation axis has been decreased to 13.7″ above the floor while the ear level-height is the same as that shown in Figure 5a. This results in the sound rays from the tweeter and woofer, respectively, ray 1 and ray 2, of equal length and compensation for the woofer's lagging response has been defeated.

Crossover and Baffle Step Compensation (BSC) Networks

The crossover network shown in **Figure 6a** is second-order filtering for both drivers and all-pass or Linkwitz-Riley. The crossover frequency is at

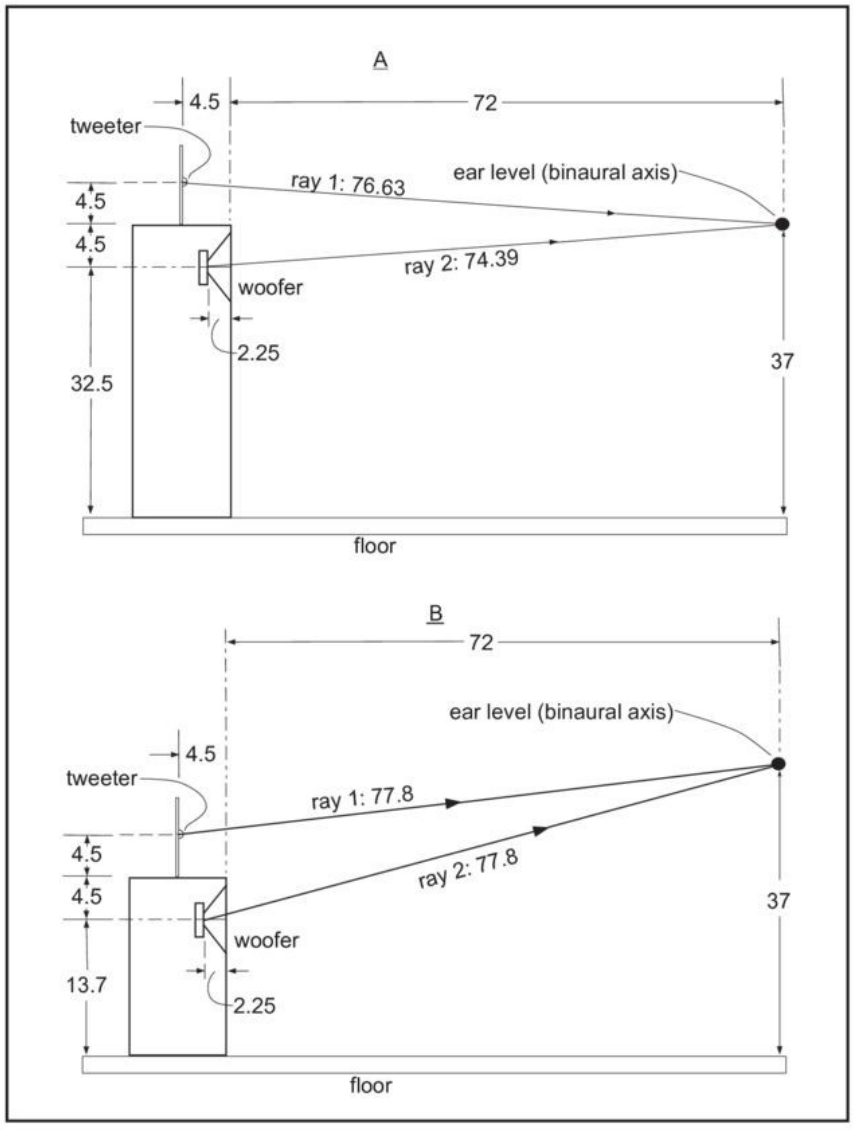

Figure 5: The vertical spacing between the woofer and the tweeter of my loudspeaker system equal to 9″ makes it necessary for ear level to be close to midway vertically between the heights of the drivers. In the example shown here with dimensions in inches, given a fixed distance between the drivers, decreasing the height of the woofer above the floor by 19″ eliminates compensation for the woofer's lagging response accomplished by placing the tweeter 4.5″ behind the woofer.

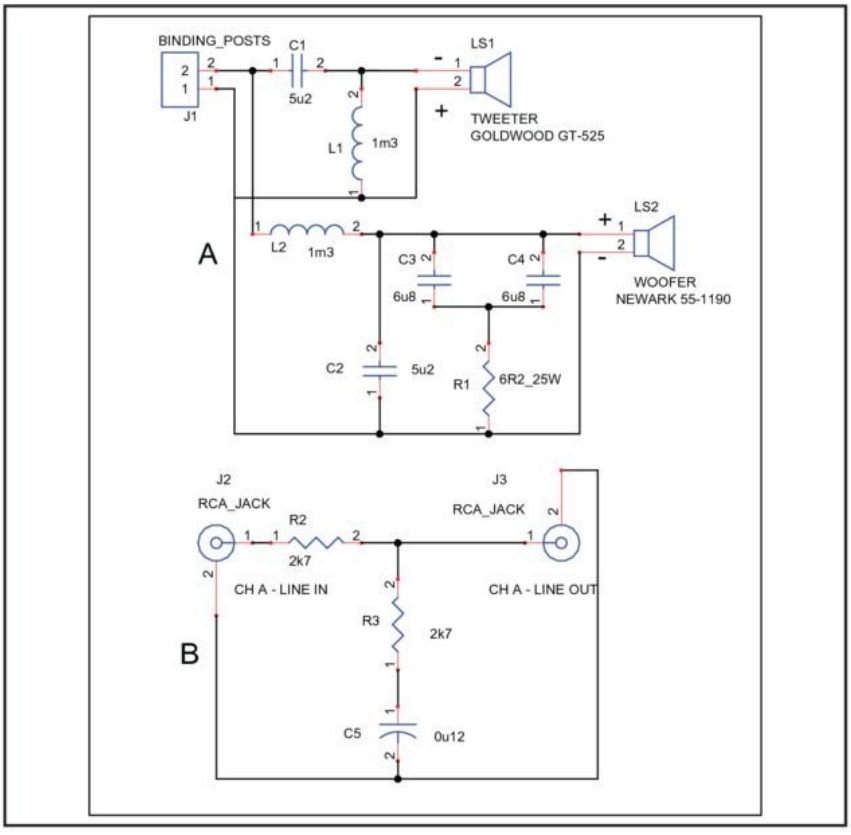

Figure 6a: This is the schematic diagram of the crossover network and impedance equalizer (Zobel network) for the woofer of my loudspeaker system. 6b: Here is the circuit diagram of the baffle step compensation network inserted between each of the two stereo channels of pre-amplification to power amplification corresponding to the width of my pair of towers.

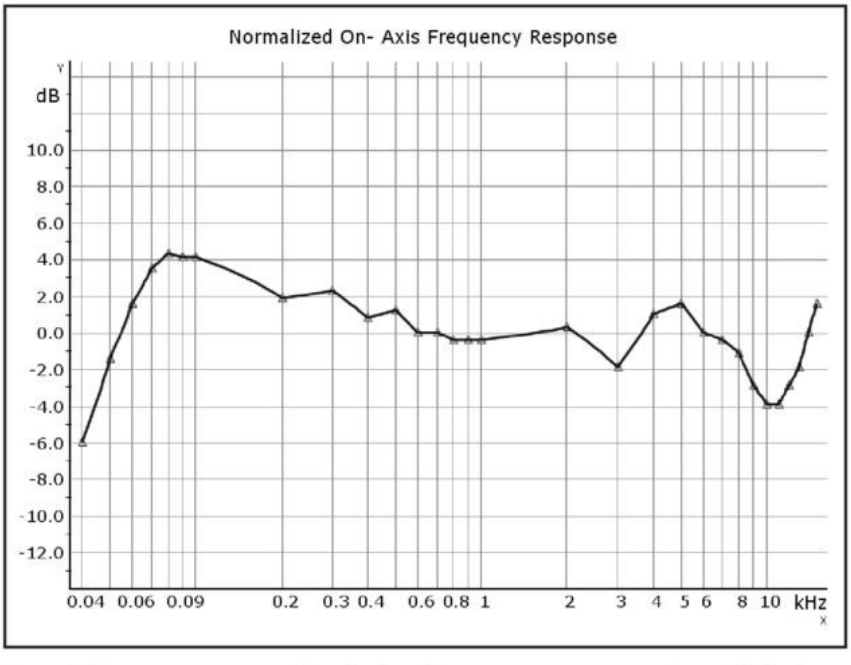

Figure 7: Frequency response of my loudspeaker system was measured near-field and far-field in the frequency ranges of respectively 40 Hz to 600 Hz, and 600 Hz to 15 kHz. In both cases, the output level of the measuring microphone was normalized to the level measured at 600 Hz.

2 kHz. The component values are based on the nominal impedance of the woofer and the tweeter equal to 8 Ω. The Zobel network shown in Figure 6a, that is capacitors C3 and C4 in parallel circuit and resistor R1, was measured to have impedance equal to 8 Ω ±2 Ω in the frequency range of 500 Hz to 10 kHz.

The resistance of resistors R2 and R3 of the BSC network shown in **Figure 6b** equal to 2.7 kΩ is large enough that little signal loss occurs even if output impedance of the preamplifier providing line level input signal to the BSC network at input jack J2 is as high as 1 kΩ. There is also little signal loss if the input impedance of the power amplifier that the network is connected to by output jack J3 is as low as 10 kΩ, relative to R2 equal to 2.7 kΩ.

The tower shown in Figure 4 has a width of 1.13'. Using John Murphy's formula for calculating the baffle step frequency, step frequency F(3) equals 380/1.13 equals 338 Hz. Capacitor C5 shown in Figure 6b made equal to 0.12 µf has reactance equal to 3.82 kΩ at the step frequency of 338 Hz. This produces –3 dB attenuation at the step frequency.

On-Axis Frequency Response Testing

Frequency response testing of my loudspeaker system in the range of 40 Hz to 600 Hz was done using the near-field technique. The measuring microphone was placed as close as possible to the woofer's dustcover without rubbing against it. Output voltage level of the microphone was measured with respect to frequency given constant amplitude and gain settings of a sine wave generator at input to the power amplifier driving my loudspeaker system. Output level of the microphone at 600 Hz was made the reference level. I found the normalized decibel level at each test frequency according to:

$$dB = 20\log_{10}\frac{V_{OUT_of_mike_test_frequency}}{V_{OUT_of_mike_at_600_Hz}}$$

Given the crossover frequency of my loudspeaker system equal to 2 kHz, I could safely assume that the tweeter was not contributing significantly to the system's output level at 600 Hz. The piston diameter of the 55-1190 8" woofer measures 17 cm. Thus, the limit of accurate measurement of the woofer's near-field frequency response is at maximum 644 Hz.

Measuring frequency response in the range 600 Hz to 15 kHz was done far-field applying the amplified output of the cosine tone burst generator previously described across the input terminals of my loudspeaker system. This was done outdoors taking the precautions of elevating my loudspeaker system at a sufficient height above the ground and keeping obstructions laterally (house, shed, trees, etc.) at an adequate distance away from my loudspeaker

system and measuring microphone.

The measuring microphone was placed 39" (about 1 m) distant from the front panel of my loudspeaker system in the vertical plane of the woofer's radiation axis and vertically midway between the tweeter and the woofer. Monitoring the output voltage of the mic on an oscilloscope, for each test frequency, I measured and recorded the voltage of the peak of the fifth half-wave of the tone burst on the oscilloscope's screen. Where V_{PK} equals the peak voltage of the fifth half-wave on the scope, decibels at each test frequency was calculated according to:

$$dB = 20\log_{10}\frac{V_{PK}@test_frequency}{V_{PK}@600_Hz}$$

Figure 7 shows a graph of the normalized frequency response measurements that I made on my loudspeaker system.

Conclusion

Only speculating, I think that a possible improvement of my design might be lowering the crossover frequency to 1 kHz. Listening to stereo recordings of music with the pair of speakers described here, I'm very pleased with the enhanced stereo separation, clarity, and fullness of reproduction that I hear. Reproduction with "conventional" stereo speakers by contrast sounds somewhat muffled to me.

Resources

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Measurements Rate New SMT Low-Voltage JFETs Under Consistent Conditions

An Update Using Modern JFETs

In 2013, Dimitri Danyuk wrote an article titled "Measurements Rate SMT Low-Voltage JFETs Under Consistent Conditions," which presented apples-to-apples test results for noise densities for several low-noise discrete, surface-mount technology (SMT) n-JFETs. New JFETs have appeared on the market in the last five years, so he decided to extend his design note from 2013. The tests used the same operating conditions so that designers can make a fair comparison for their applications.

By
Dimitri Danyuk
(United States)

There are several benefits to using contemporary junction field effect transistors (JFETs), which have noise voltages (V_n) equal to those found in low-noise bipolar junction transistors (BJTs). JFETs also have considerably lower noise current (I_n) and outperform bipolar transistors on low-frequency noise at source impedances of several kilohms and above. JFETs are also more linear than bipolar devices due to their square law transfer characteristics compared to the exponential transfer characteristics in BJTs. JFETs can tolerate higher input signals and not rely so much on the feedback to correct distortion.

Input signals can contain substantial out-of-band components (e.g., ultrasound noise "from over-sampled noise-shaping DACs, interference from switched mode power supplies, RF from cell phones, Wi-Fi, Bluetooth devices, etc.). If the input device is nonlinear, these out-of-band signals can be rectified. As a result, an unwanted envelope of rectified waveform appears at the output in addition to the desired signal. BJTs have forward-biased base-emitter junctions and JFETs have a reversed (or zero) biased p-n junction. This fact, together with exponential transfer in BJTs explain why BJT devices exhibit greater susceptibility to out-of-band signals compared with JFETs.

Another distinct difference between the BJTs and the JFETs is that the input control variable for a BJT is a current, while for the JFET, it is a voltage. JFETs do not need a biasing current for normal operation and generally don't require DC blocking capacitor to prevent affecting the DC biasing current from the previous stage or from the signal source.

A JFET has an input voltage noise closely related to the Johnson noise of the conducting channel:

$$V_n = \sqrt{\frac{8kT}{3g_m}}$$

where g_m is the transconductance:

$$g_m = g_{m0}\sqrt{\frac{I_d}{I_{dss}}}$$

where g_{m0} is the transconductance at $V_{gs} = 0$, thus:

$$V_n = \sqrt{\frac{8kT}{3g_{m0}}}\sqrt[4]{\frac{I_d}{I_{dss}}}$$

V_n varies only slightly as drain current (I_d) is varied (4th root dependence), four times increase in I_d reduces noise voltage square root of two times (3 dB).

The V_n in JFETs will be lowest when the devices are operated at zero V_{gs} ($I_d = I_{dss}$) or slightly

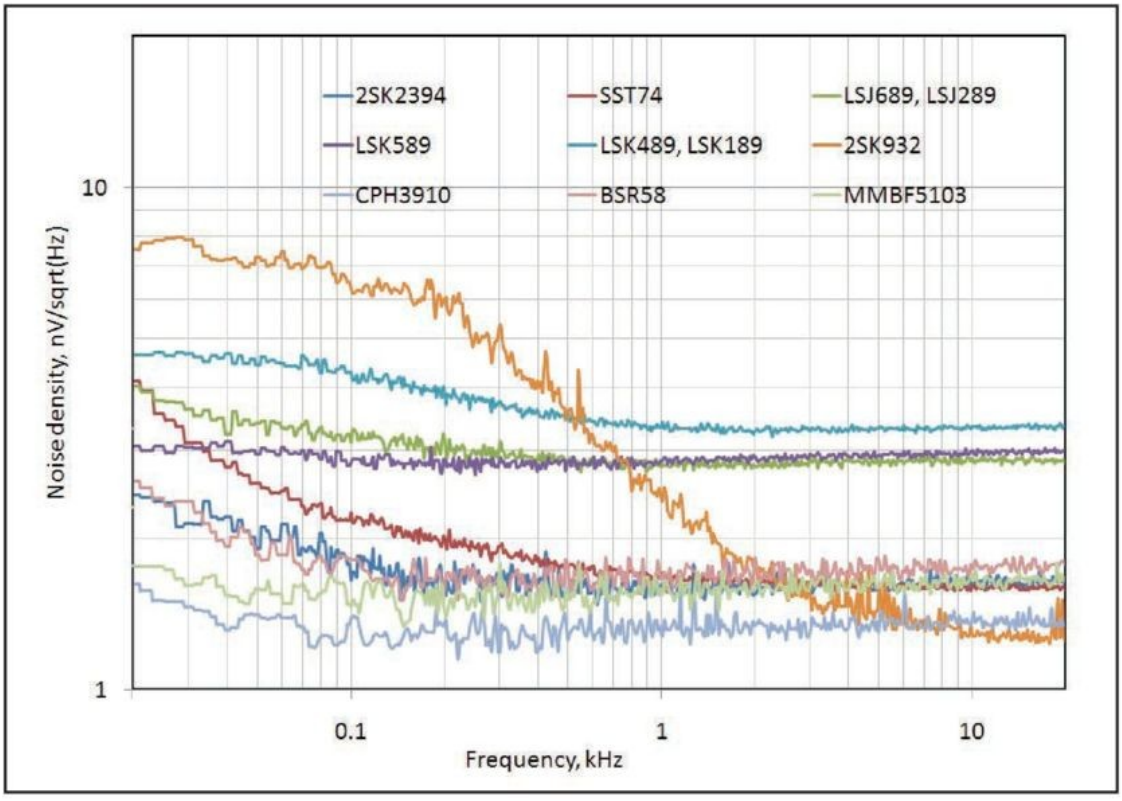

Figure 1: Noise measurement of a number of discrete JFETs show that some p-channel and n-channel JFETs can be suitable for low noise applications.

positive V_{gs}, where transconductance (g_m) is at its highest value. This will be true only if device dissipation is moderate in relation to the heat dissipation capability of the JFET package. It is also important to maintain drain-gate voltage well below the gate current (I_g) breakpoint.

A JFET input current noise is proportional to frequency:

$$I_n \approx \frac{2}{3}\omega C_c V_n$$

where C_{gs} is the gate to channel capacitance. While bipolar transistors may have low-voltage noise, the current noise at low frequencies is much larger than for JFETs due to the electronic shot (Schottky) noise I_s:

$$I_s = \sqrt{2qI_b}$$

where q is the elementary charge and I_b is the DC base bias current. Due to the very low gate bias current, the Schottky noise in JFETs can usually be neglected. It is always possible to lower the noise voltage by connecting several transistors in parallel (increasing g_m twice lead to square root of 2 (3 dB) noise reduction). At frequencies below a certain corner frequency, all devices exhibit 1/f noise or flicker noise, where noise spectral densities higher than V_n given above are observed.

Low-Noise Measurement System

We measured voltage noise under consistent conditions using a low-noise measurement system. I_d was set at 1 mA, V_{ds} was 2.5 V, and ambient temperature T was 25°C. The thermal resistance of the SOT23 package is about 200°C/W, so the 2.5 mW power dissipation gives negligible rise of junction temperature. As in my original article (see Resources), we measured 16 samples of each device and averaged the noise data from 20 Hz to 20 kHz of 1/f and thermal noise. The curves shown in **Figure 1** illustrate input noise density for various devices. Measurements of equivalent noise voltage are described in **Table 1**.

Two new n-channel devices from ON Semiconductor exhibited voltage noise densities from 1.5 to 2 nV/√Hz: the ON Semiconductor CPH3910 and the ON Semiconductor 2SK2394. Another high transconductance n-channel JFET from ON Semiconductor has significant low-frequency noise, but can be used successfully at the frequencies above 5 kHz is the 2SK932. A long-awaited p-channel device from Linear Systems, the LSJ74/SST74, has 1.7 nV/√Hz noise. All these JFETs are high-gain devices with moderate drain current (I_d = 5 to 40 mA, g_m = 10 to 40 mA/V).

Another three JFETs from Linear Integrated Systems—the LSJ689/LSJ289, the LSK589, and the LSK489/LSK189—showed higher noise in the range 2 to 3 nV/√Hz. These devices have moderate transconductance from 4 to 10 mA/V.

It should be noted that two measured devices from On Semiconductor—the MMBF5103 and the BSR58—don't have noise data in their datasheets. This means that the manufacturer doesn't control noise parameters for these devices. The MMBF5103 and the BSR58 are designed for low-level analog switching and were included in this study for

About the Author

Dimitri Danyuk has been the principal hardware engineer at Harman luxury audio since 2013. He graduated with honors from Kyiv Polytehnic in 1985. His passion is to help audiophiles in all aspects of audio design. His articles have appeared in a number of magazines including *Journal of the Audio Engineering Society*, *Institute of Electrical and Electronics Engineers Transactions*, EDN Network, *Electronic Design*, and others. Dimitri also provides consulting services for various businesses.

educational purpose.

Noise performance of several low-noise discrete JFETs appeared on the market in recent years was measured. The tests were done in the same way as in the first part of the article that I wrote five years ago for *Electronic Design* (see Resources) so that designers can make a fair comparison and select a proper part for their applications.

Table 1: Measurements of equivalent noise voltage are described.

Measured Noise for SMT JFETs

Part Number	Polarity	Voltage Noise Density, Vn, nV/√Hz at 1 kHz	Equivalent Noise Resistance, R_{eq-n}, Ω	Equivalent Noise Voltage, dBV ±5 dBV, R_g = 0, 20 Hz to 22 kHz measurement bandwidth	Equivalent Noise Voltage, dBV ±5 dBV, R_g = 0, 20 Hz to 22 kHz measurement bandwidth, A-weighted
CPH3910	n-channel	1.4	120	-134	-136
2SK2394	n-channel	1.6	160	-133	-135.8
MMBF5103	n-channel	1.6	160	-134	-135.5
SST74	p-channel	1.7	180	-133	-136.6
BSR58	n-channel	1.8	200	-131.5	-133
LSJ689 (dual), LSJ289 (single)	p-channel	2.8	490	-129.5	-132
LSK589 (dual)	n-channel	2.9	530	-129	-132
LSK489 (dual), LSK189 (single)	n-channel	3.4	720	-128	-130.5
2SK932	n-channel	2.5 (1.2 at 10 kHz)	400 (100 at 10 kHz)	-129	-131.6

R_g-signal source resistance I_d = 1 mA, V_{ds} = 2.5 V, T = 25°C

Resources

D. Danyuk, "Measurements Rate SMT Low-Voltage JFETs Under Consistent Conditions," *Electronic Design*, April 2013, www.electronicdesign.com/power/measurements-rate-smt-low-voltage-n-jfets-under-consistent-conditions.

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BSR58: N-Channel Low-Frequency Low-Noise Amplifier
ON Semiconductor | www.onsemi.com/pub/Collateral/BSR58-D.pdf

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Linear Integrated Systems | www.linearsystems.com/lsdata/datasheets/LSK489_LOW_NOISE,_LOW_CAPACITANCE_MONOLITHIC_DUAL_N-CHANNEL_JFET.pdf

CPH3910: N-Channel JFET, 25 V, 20 to 40 mA, 40 mS, CPH3
ON Semiconductor | www.onsemi.com/pub/Collateral/ENA1965-D.PDF

2SK2394 N-Channel JFET
ON Semiconductor | www.onsemi.com/pub/Collateral/EN4839-D.PDF

MMBF5103 N-Channel Low-Frequency Low-Noise Amplifier
ON Semiconductor | www.onsemi.com/pub/Collateral/MMBF5103-D.pdf

2SK932 N-Channel JFET
ON Semiconductor | www.onsemi.com/pub/Collateral/EN2841-D.PDF

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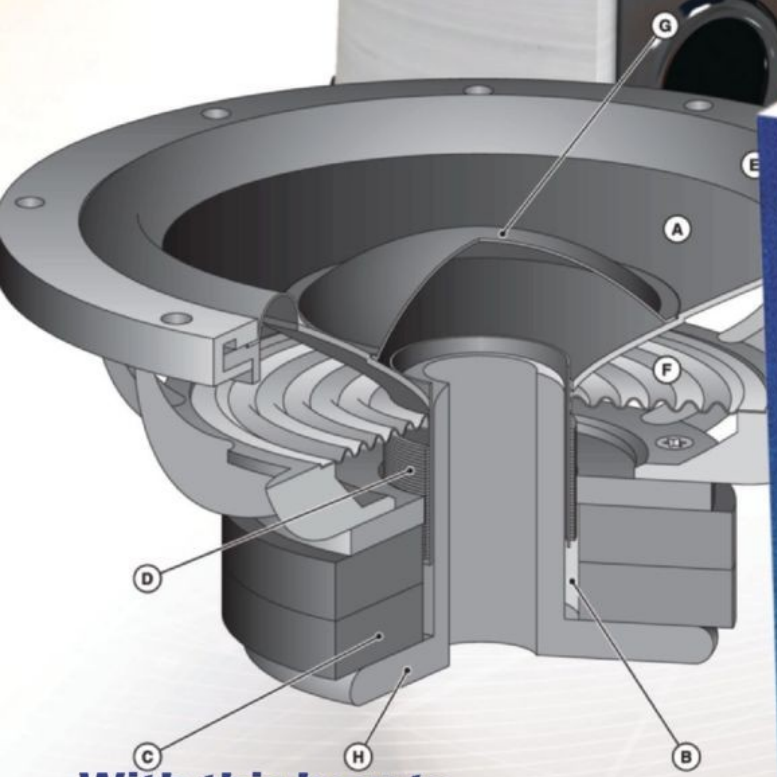

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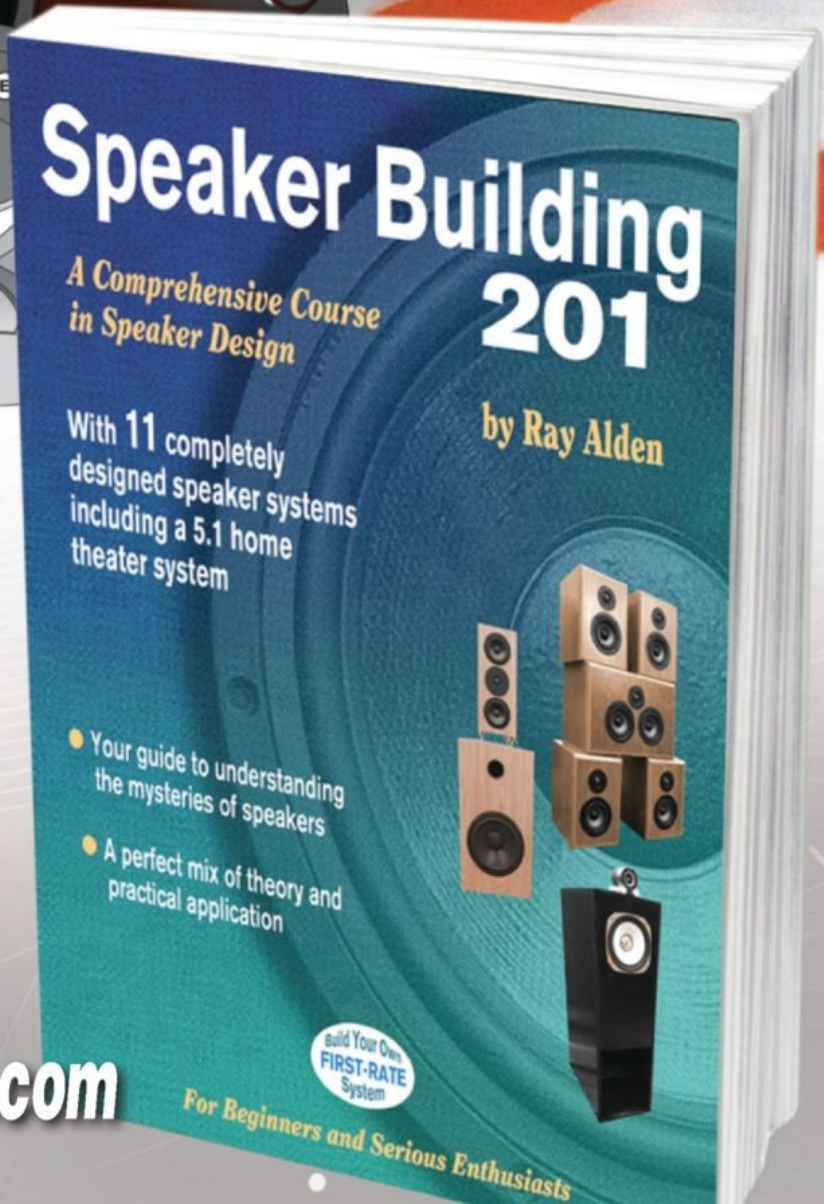

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A Bogen APH-1050 Turned Guitar Amp

In this article, Richard Honeycutt modifies an old Bogen APH 1050 public address amplifier, and gives it a second life as a high-performance guitar amplifier.

By
Richard Honeycutt
(United States)

As you likely know, yard sales, surplus sales, and hamfests can be treasure troves for hollow-state audio buffs. Some years ago, a friend of mine went to a surplus auction held by the local school system, and brought home a load of hollow-state PA equipment.

Among the items were several Bogen APH-1050 power amplifiers (actually manufactured by Stromberg-Carlson, but branded as Bogen). These amps were specified as providing 50 W output power, and utilized a 7199 pentode/triode preamp/phase splitter driving a pair of 7581 beam power tubes connected in push-pull with overall feedback applied. As manufactured, these amps provided a clean 40 W (not 50 W as claimed, but few amplifiers of this vintage ever actually produced the rated output power, and getting within 20% of the specified power was notable). The 7199s are no longer manufactured, but the 6GH8 is a close replacement. New 7581s can still be purchased, but 6L6GCs work just as well in this circuit. Both tubes draw 0.9 A for the filament, and the datasheets recommend the same output transformer primary impedances for the 440 V plate supply used by this amp.

As you can see from **Photo 1** and **Photo 2**, the APH-1050 came with a modification-friendly feature: two extra octal sockets that were originally intended for a remote-control relay and an input

Photo 1: This is the top view of the Bogen APH-1050 before the modification.

transformer. Neither of these is needed for a guitar amp, but more gain is needed, since the APH-1050 was designed for line-level inputs. You can see the original circuit in **Figure 1**.

If you wanted a really grungy heavy-metal sound, you could add a couple of 6SL7s (μ of 70). But for my preference, not so much gain is needed, so I chose 6SN7s (μ of 20). The first one acts as a preamp and a cathode-follower to drive an effects loop. The second acts as another preamp (where the distortion occurs) and another cathode-follower to drive the 7199's pentode preamp. Actually, this arrangement provides enough gain to seriously overdrive the pentode stage, if you like that kind of sound.

Tone Controls

In my 58 years playing guitar, I have found tone controls to be of little use. Electric guitars have their own bass and treble controls, which I usually set at maximum unless I decide to pull back on the treble for jazz. My amp tone controls usually wind up being set somewhere close to flat. In fact, the amps I like least are the ones whose tone controls have no setting that is close to flat. Often, the pickup switch on my Guild Starfire IV (two humbuckers) is set to use both pickups. But for a more twangy sound, I can switch to the bridge pickup, or for a mellower sound, I'll use the fingerboard pickup.

So I decided to save myself the trouble of designing and building tone controls for this amp. I kept the volume control and the high-pass filter ("BRIGHT switch," in guitar-amp parlance), although I moved the volume control from its original location

Photo 2: Here is a partial view of the inside of the chassis bottom before the modification.

to a spot between the tubes. I installed a 1/4" input jack and wired it in instead of the terminal strip that was originally used. I also added two 1/4" normally closed jacks for effects. But the interesting part of the modification is the design of the 6SN7 stages, which I'll now describe.

Preamp Design

Having two octal sockets, I decided to use half of a 6SN7 as the input stage, and the other half as a

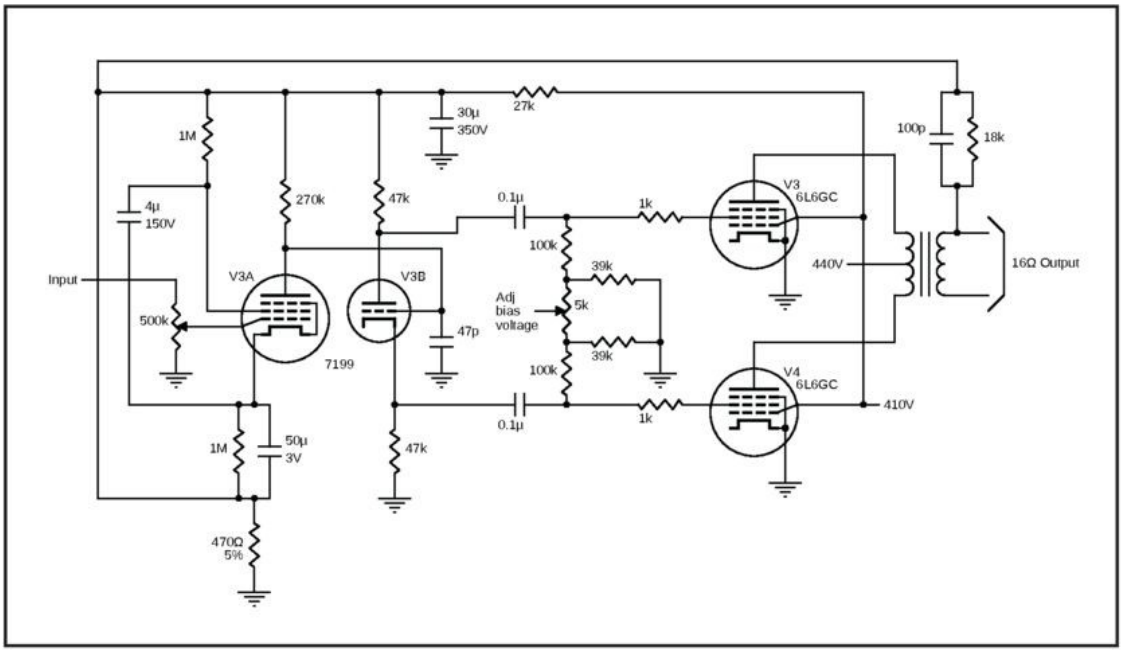

Figure 1: This is a schematic of the Bogen APH-1050.

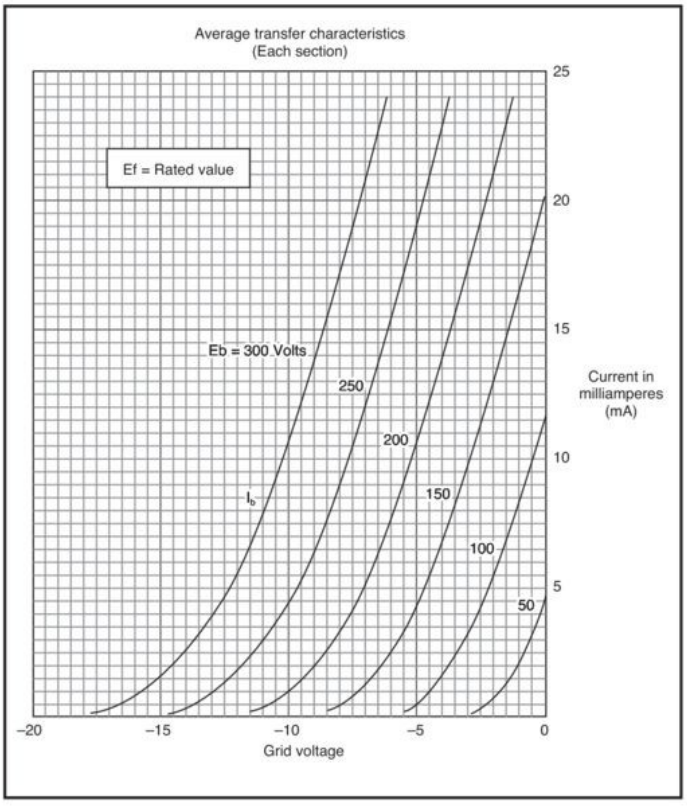

Figure 2: These are the transfer characteristics of the 6SN7.

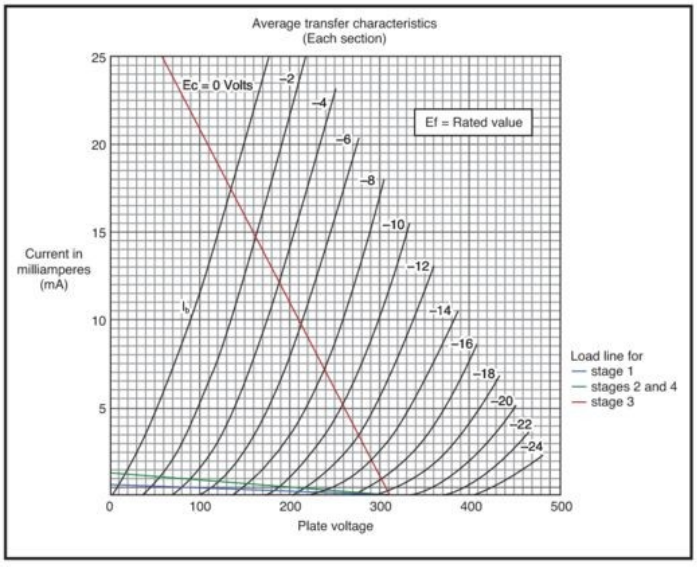

Figure 3: These are the load lines for each of the four stages.

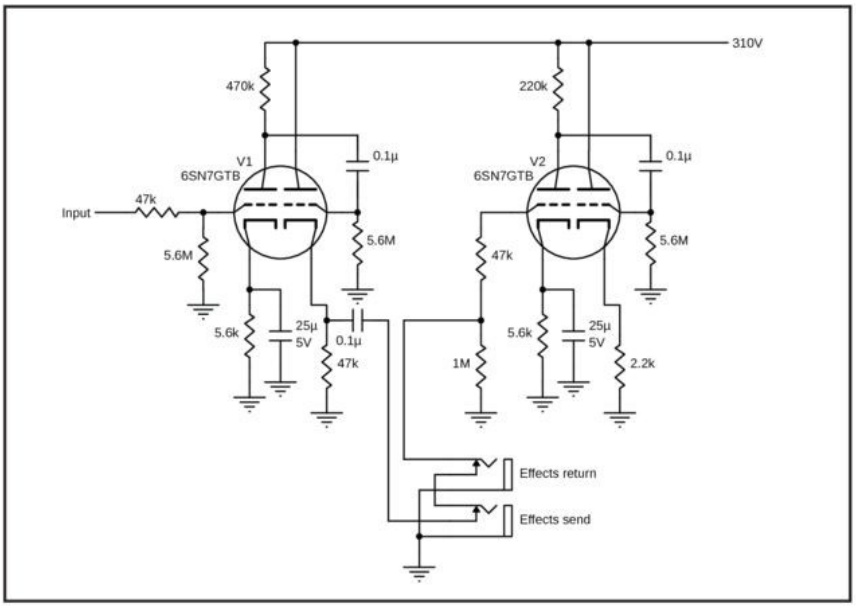

Figure 4: This is the schematic for the new preamp circuit.

cathode follower to drive the effects loop. A second 6SN7 would provide a post-effects gain stage and another cathode follower would drive the 7199 that I retained from the original Bogen design.

The signal level produced by an electric guitar depends upon the type of strings, the height to which the pickup pole pieces are adjusted and, of course, the guitar's control settings and how hard the pick is striking the strings. A common input signal level to produce full output from a guitar amp is 25 mV. I decided to design for noticeable distortion to begin when the input level to the amp is about 75 m_VP.

It is hard to drive a triode stage into much distortion with a signal that low, and besides, I wanted to be able to control the distortion by using both the guitar's volume controls and the amp volume control. So I chose to use the two gain stages to overdrive the second cathode follower, which, in turn, would be capable of overdriving the pentode of the 7199, if desired. The input sensitivity of the original amp was 1 V, so the minimum gain would have to be 1 V/25 mV = 40. More gain would allow more distortion when the guitar and amp volume controls are turned up high. I decided on a total preamp gain of about 200, calling for a stage gain of roughly the square root of 200, which is 14.14. A little higher stage gain will actually be necessary, since the cathode followers never really live up to their reputation of having unity gain.

Figure 2 shows the transfer characteristic of each triode of a 6SN7. As you will notice, at very

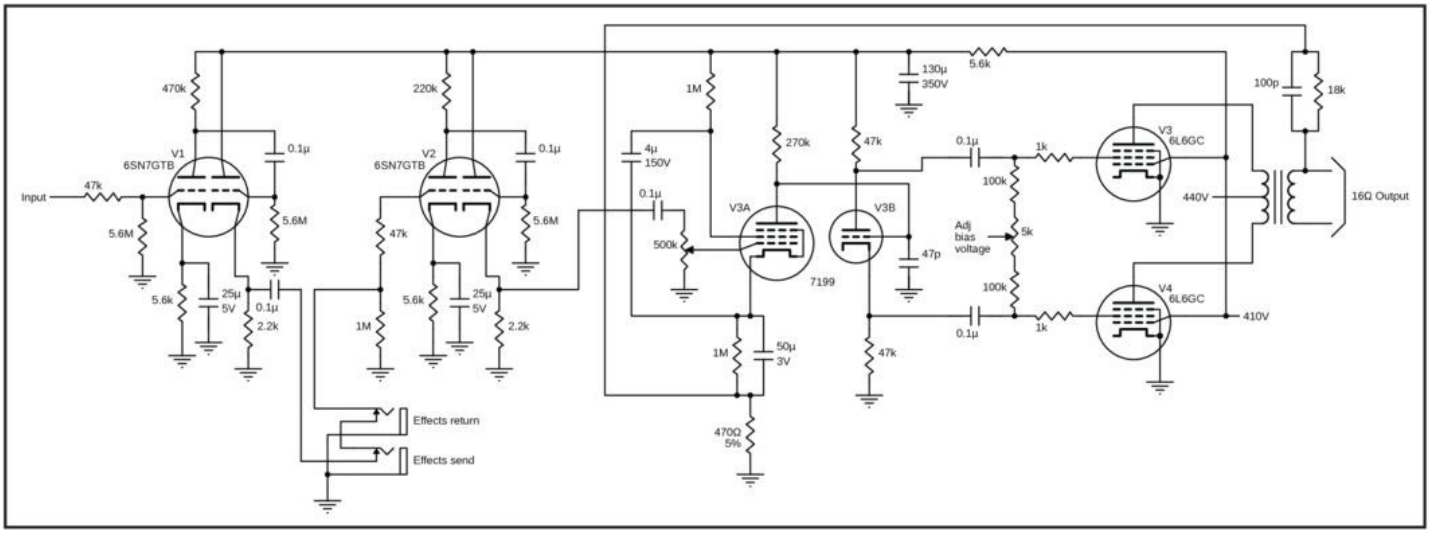

Figure 5: Here is the complete modified circuit.

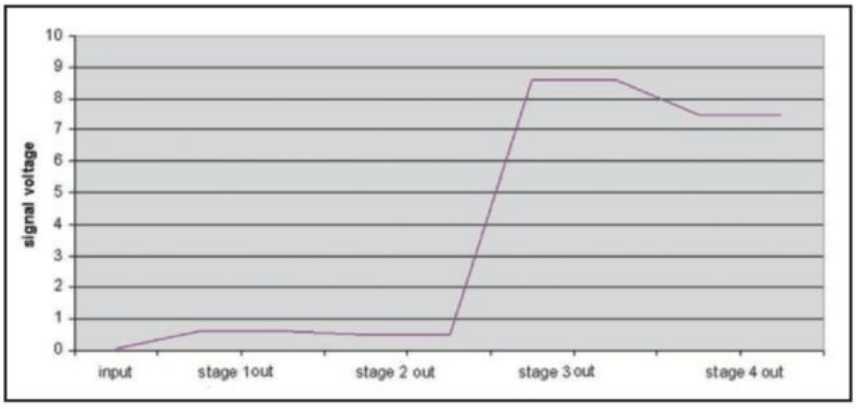

Figure 6: The signal level diagram

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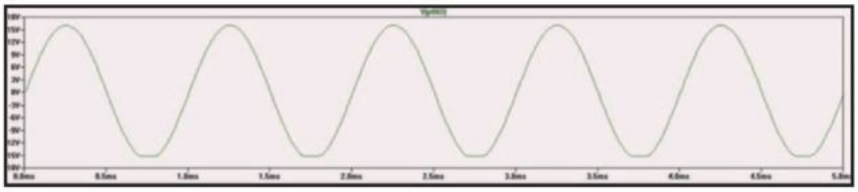

Figure 7: The output wave with a 75 mV peak sine-wave input

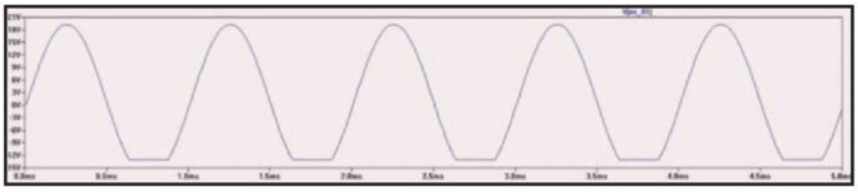

Figure 8: The output wave with a 100 mV peak sine-wave input

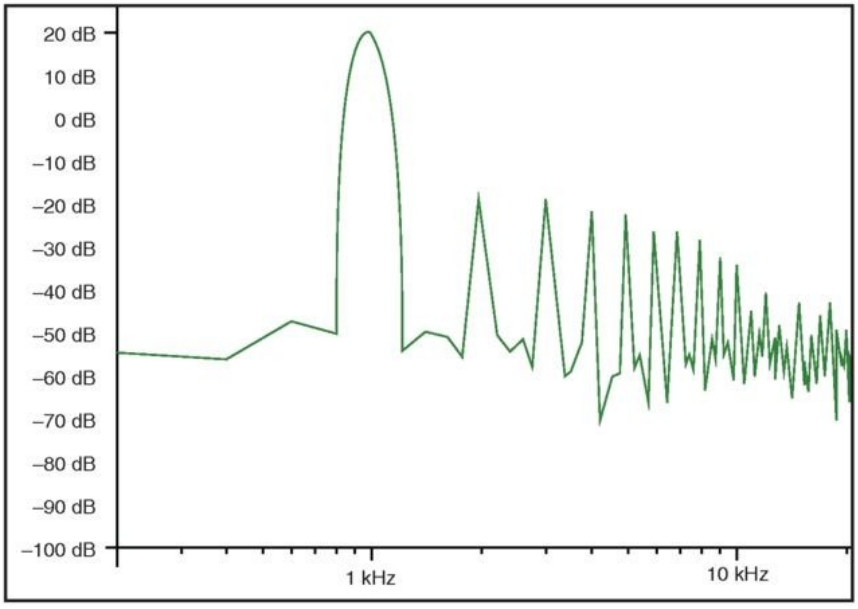

Figure 9: The FFT of an output wave with a 75 mV peak input

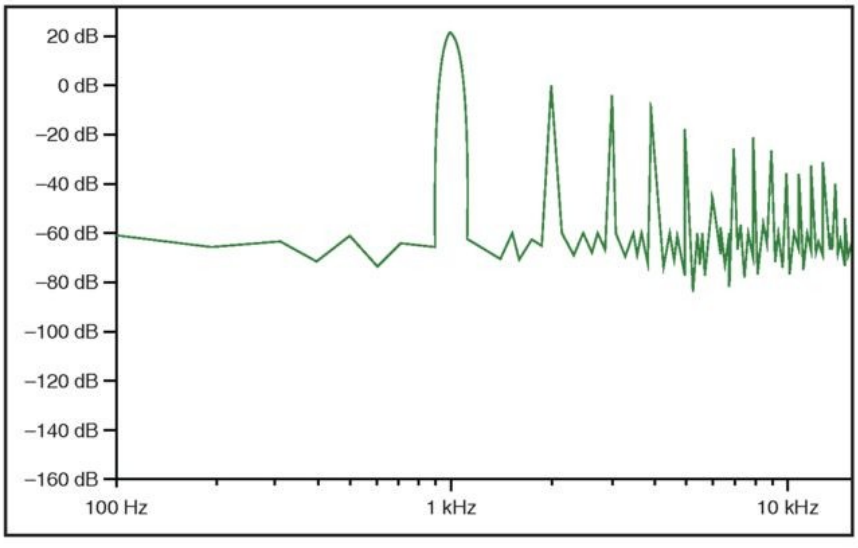

Figure 10: The FFT of an output wave with 100 mV peak input

low currents (below 1 mA), the lines are curved, indicating that the triode will produce more distortion if biased for operation in this area. We can design the four-stage amplifier as described in the last paragraph, with load lines as shown in **Figure 3**.

Stage 1 is the first preamp stage. Stage 2 is the cathode follower that drives the effects loop. Stage 3 is the preamp stage fed by the effects return. Stage 4 is the cathode follower that drives the original input of the amp (the grid of the 7199's triode). **Figure 4** shows the preamp circuit. The voltage gains (from LTSpice modeling) are:

Stage 1: 17.8
Stage 2: 0.8 (with no effects circuit inserted)
Stage 3: 16.8
Stage 4: 0.87 (driving the original 500 kΩ volume control)

This gives us a total gain of 208.13.

The original Bogen circuit uses a 310 V low B+. We can keep this voltage if we change the 27 kΩ dropping resistor shown at the top of Figure 1 to a lower value. The 27 kΩ resistor drops the voltage from 410 V to 310 V. This means its current is (410 V to 310 V)/ 27 kΩ = 3.7 mA. Our four new triode stages draw the following currents:

Stage 1: 515 µA
Stage 2: 6 mA
Stage 3: 890 µA
Stage 4: 6 mA

The total current in the dropping resistor will now be the sum of these currents and the original 7199 currents, which comes to 17.1 mA. So to drop 10 V, the resistor value will have to be changed to 100 V/17.1 mA = 5848 Ω. The nearest standard 10% value is 5.6 kΩ. This will give us a low B+ of 314 V, which is close enough. In order to provide enough decoupling, the 30 µF decoupling capacitor will have to be increased to 130 µF. The complete modified circuit appears in **Figure 5**.

Figure 6 shows the signal-level diagram, based on a 35 mV input signal. **Figure 7** and **Figure 8**, respectively, show the output waveform of the preamp circuit with a 75 mV and a 100 mV sine-wave input. As you can see, asymmetrical clipping (mainly even-order harmonics) barely show up on the negative peaks with a 75 mV input. With a 100 mV input, clipping is more aggressive. Using the guitar's volume controls and the amp's volume control together, quite a wide range of distortion

sound can be created, since the preamp can certainly drive the 7199 into hard clipping.

Figure 9 and **Figure 10** show the Fast Fourier Transforms (FFTs) corresponding to the waveforms of Figure 7 and Figure 8. As you can see, second harmonic predominates—more so when the preamp is driven harder. As is normally the case with hollow-state amps, the harmonic amplitudes decrease at higher harmonic numbers.

Figure 11 is a graph of second-harmonic distortion of the preamp plotted versus input voltage. Of course, by the time the input voltage gets anywhere near the maximum shown on this graph, the 7199 would be in hard clipping. In this case, the 7199's sound would dominate, unless, of course, the amp's volume control were turned down to keep the preamp output level to a volt or less.

Although feeding the output of a hollow-state power amp into a load impedance higher than the design center of the output transformer is generally not recommended, tests have shown that this particular amp works well—and delivers a bit more power—when an 8 Ω speaker is fed from the 16 Ω transformer tap.

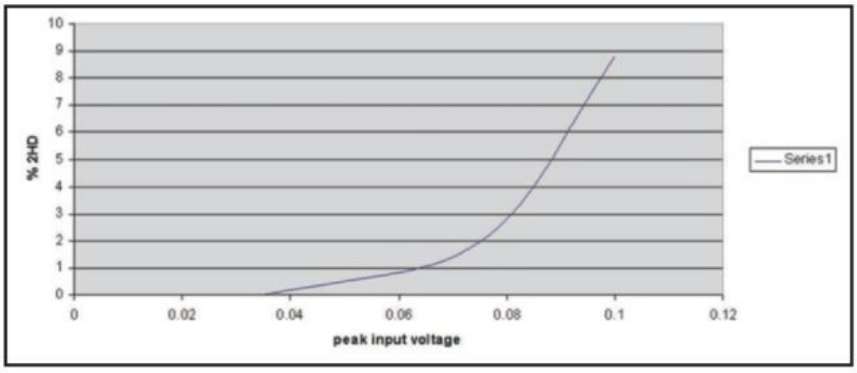

Figure 11: Percent of second-harmonic distortion with various input voltages

One of the joys of hollow-state experimenting is finding a well-designed piece of equipment such as the Bogen APH-1050 that can be easily repurposed into a high-performance amp that fulfills a function totally unlike anything envisioned by the designer. And when that piece of equipment has convenient modification aids (e.g., spare tube sockets), the joy is just that much greater!

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