Engineer, Professor, and Entrepreneur, p. 29

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From the Editor's Desk

Welcome to Our Booth

The 135th Audio Engineering Society (AES) Convention will be held in New York, NY, on October 17–20. The event brings together thousands of audio professionals who will explore the latest in products and technologies.

We'll be there, too, introducing our readers to a newly redesigned *audioXpress* magazine that will debut with the November issue.

The redesign will offer a new layout, more pages, and expanded content catering to a variety of reader interests. Those interests include acoustics and applications ranging from vintage equipment repairs to building guitar FX pedals and innovations in professional audio.

Intrigued? Then stop by our booth (number 2560) at the AES convention to find out more. We'd appreciate the time. After all, there will be a lot to distract you at the Jacob K. Javits Convention Center. Audio professionals come to the convention for its four days of workshops, masters classes, skills training, research discussions, and more—including special events that explore audio industry innovations and frontiers.

In 2011, the convention drew more than 350 businesses and 18,000 attendees to New York. According to the AES, business participants ranged from boutique enterprises to Fortune 500 companies. Their array of specialties included commercial recording, broadcasting, film and video production, audiovisual computing software, Internet audio, and more.

This year's convention will also offer some new attractions, including a "Sound for Picture" workshop featuring Emmy and Oscar winners. Visit www.aes.org to learn more.

Regards,
Mary Wilson
editor@audioxpress.com

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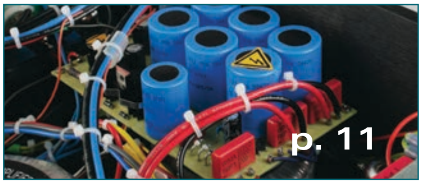

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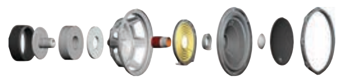
 By Mike Klasco and Steve Tatarunis (United States)

Speaker Protection for Mobile Consumer Products



Microspeaker products push signal-processing technology to new heights

Last month, we discussed passive speaker protection, focusing on what happens with the speaker when it is pushed to its limits. This month, we are examining a new generation of effective and powerful speaker protection signal-processing technology.

Purists will ask if this limiting and compression-based signal processing is real. And while these signal processing tricks do not belong in studio monitors or audiophile speakers, most mobile electronics and even soundbars can benefit from this technology.

Sophisticated loudspeaker signal processing for pro-sound applications is commonly used today. These processors are called loudspeaker management systems. These systems require robust operation and stable performance because they are used in large venues where any failure is disastrous.

Using the incredible power of large-scale integrated circuits (ICs), it is possible to achieve this level of sophistication for smartphones, laptops, tablets, and Bluetooth speakers and docking stations. With the efficient use of a DSP, it is now practical and cost effective to make music sound louder and clearer with more bass. Additionally, this can occur while load monitoring the loudspeaker's "vitals" (i.e., mobile consumer electronics can play louder and sound better than they should).

The latest generation of these mobile devices sounds surprisingly clear and loud, from the speakerphone to the stereo music reproduction (e.g., the HTC One smartphone, see **Photo 1**). Even experienced audio engineers are surprised by what seems to be a breaking (or at least a bending) of the rules of physics!

Photo 1: The HTC One smartphone uses new technology to provide improved sound. (Photo courtesy of HTC)

CONSUMER PRODUCTS AND SPEAKER PROTECTION

You may be surprised that touring and permanent installation facility loudspeaker management systems handle loudspeaker failure mode considerations similar to those faced by smartphone speakers. When you consider the quantity of smartphones made each year—close to 1 billion, along with all the tablets, laptops, Bluetooth speakers, and other mobile consumer electronics products—you can see the incentive for semiconductor companies to design their ICs for mainstream mobile consumer electronics products.

The retail distribution channel affected the audio market for years, and the super retailers (e.g., Costco, Best Buy, Target, etc.) permit customers to return just about anything. The retailers then pass the returns along to the manufacturers for credit. Now, consider how costly it would be for a smartphone manufacturer to get a few hundred million units returned with blown speakers! Too many returns get prohibitively expensive for the vendors. Therefore, consumer electronics products must be practically invincible. Integrated defensive engineering are the keywords here.

Limiting the peak signal levels to prevent over excursion and voice coil arcing is a well-known technique. But, the processing for mobile electronics with its heavy dynamic peak limiting and high-compression ratios (e.g., commercial in AM broadcasts) introduces new voice coil thermal considerations.

The problems for high duty cycle audio are the opposite of those found in stadiums, which can have a huge crest factor with peak levels 10 dB or more higher than the average power expectancy. Voice coil heating is not as rapid when the program material has a wide dynamic range, because the duty cycle is not as high and the speaker's natural heat dissipation can keep up with the heat the amplifier signal generates in the voice coil.

When limiters, compressors, MP3s, or radio sources are used, the peak-to-RMS crest factor is far lower and the average power to the speaker is much higher, resulting in more heat. Also, the speaker system's apparent loudness can be far greater when compressor/limiters are used and still don't exceed the speaker's excursion limits.

The speaker conducts heat through the membrane to cool. Other components stay cool due to the moving air (turbulence) from the sound waves themselves. Lower frequencies generate more air movement causing more cooling and providing higher power. Ferrofluids effectively transfer heat from the voice coil, but heatsinking these tiny magnetic structures are only part of the solution.

The assumptions of (open-loop) speaker modeling protection break down if the speaker port is blocked, the air movement is restricted, or the ambient temperature rises. If the air cannot cool the coil, the internal temperature

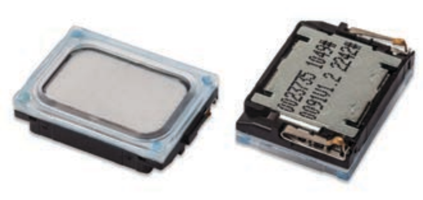


Photo 2: The Knowles IPx8 is a good example of a small microspeaker. (Photo courtesy of Knowles Electronics)

rises much faster than expected and the speaker can be damaged in a few seconds. The relationship between coil temperature, power level, frequency, duration, ambient temperature, and airflow is complex. It is virtually impossible to reliably predict in mobile consumer electronics.

Because microspeakers must be small, it is easy to move the diaphragm further than the maximum allowable excursion (typically around 0.4 mm). As speakers get thinner, the excursion tends to become smaller, which greatly restricts the output sound level. Most smartphones' microspeakers are less than 4-mm thick. They are typically 2 mm or 3 mm total. The Knowles Electronics IPx8 microspeaker is only 2 mm thick (see **Photo 2**). The clearance behind the voice coil to the magnetic structure limits the backstroke. The front of the diaphragm's proximity to the speaker's protective grille limits the forward stroke. Rearward over excursion can quickly destroy a voice coil.

These real-world factors delimit the safe-operation area. In some new speaker protection processors (e.g., NXP Semiconductors), a temperature line limits the power amplifier to avoid self-heating temperatures. And, a frequency line removes frequencies below resonance to prevent over-excursion.

Allowing for changes in the acoustic environment and ambient temperature ensures safe operation, it provides only a modest sound output. The thinking is that smartphones, like AM radio stations, should always operate near peak output. Low-level signals receive more amplification and the dynamic range is compressed upward. If the sound is subjectively louder, the user will not need to further increase the volume.

Because audio signals are dynamic, they only rarely use the amplifier's peak power. Compressing the signal's dynamic range increases the apparent volume without affecting the peak levels (which is why commercials sound louder than the rest of the TV or radio broadcast).

These dynamic compressors work by adding gain to the quiet parts of the music and quickly reducing it at peaks (i.e., the "attack time"). The attack time is typically very fast (50 µs). The corresponding decay time over which the gain is increased is typically much slower (5 s).

Almost all the signal processing and IC amplifier vendors have introduced, or are in the process of introducing, solutions for dynamic signal processing. Differences in the various approaches determine what occurs, from the speaker's preset model (open loop) to the load-monitoring feedback.

RECENT INNOVATIONS

Mike Klasco wrote this portion of the article while attending the Audio Engineering Society's 51st Conference on speakers and headphones in Helsinki, Finland.

Lars Risbo of Texas Instruments (TI) gave an interesting talk entitled "Louder, Cheaper & Better through System-Level Optimization of Speaker, Amplifier, Power Supply & Intelligent Signal Processing." In 1996, Risbo founded Toccata Technology, an early switching amplifier innovator. After TI acquired Toccata, Risbo led the development and successful marketing of TI's PurePath digital audio amplifier devices. His research interests include digital signal processing, system-level and signal-chain optimization, oversampled data converters, and mixed-signal architectures. Risbo's presentation pointed out that active speaker systems used in mobile consumer electronics are commonly designed by choosing one component at a time (e.g., the speaker, the amplifier, the power supply, and the DSP).

The design practices and standards have been written over a duration of many years for this component-based approach. However, in practice this often leads to over- or under-designing the system's components, which

Photo 3: Altec Lansing's original InMotion docking station featured MaxxBass technology. (Photo courtesy of Altec Lansing)

pairs its performance and cost. For example, speakers are built heavy and rugged to meet standard power tests, which leads to poor sensitivity and linearity.

Risbo's paper examined the untapped cost and performance potential in co-optimizing the entire chain by taking a system-level view. Now, speakers can be optimized for sensitivity and linearity while the intelligent processing protects them from damage, which enhances the audio quality.

TI's SpeakerGuard protection circuitry is designed to prevent amplifier output clipping distortion and excessive power to the speaker. The automatic gain control (AGC) is only a limiter function. It works by automatically adjusting amplifier gain based on the audio signal level. This prevents speaker damage from audio that is too loud. The AGC function can also be used to improve apparent (subjective) audio loudness without increasing the peak power delivered to the speaker. If the audio signal is higher than the limiter level, the gain will decrease until the audio signal is just below the limiter setting.

WAVES'S MAXX PROCESSING

Waves, an established leader in professional signal processing, expanded its business to include consumer electronics some time ago. Waves's MaxxBass is generally acknowledged as the secret ingredient in the original Altec Lansing InMotion docking station (see **Photo 3**). MaxxBass is also credited with the breakthrough bass performance from its 1" full-range speakers. Other Maxx processing from Waves includes powerful compression and limiting found in the JBL audio package for Dell's XPS laptops.

NXP Semiconductors has also introduced a state-of-the art speaker protection processing system. It provides essentially everything pro-audio loudspeaker management processors achieve and more at a lower cost.

NXP Semiconductors does not accept unreasonable sound levels nor the practical limitations of small microspeakers. The company's position is that simply limiting the output power makes for a poor user experience and doesn't protect against blocked speaker ports or high ambient temperatures. Temperature monitoring can help but it does little to improve sound quality. High-pass filters reduce the speaker excursion at the resonant frequency but they cut out too much bass.

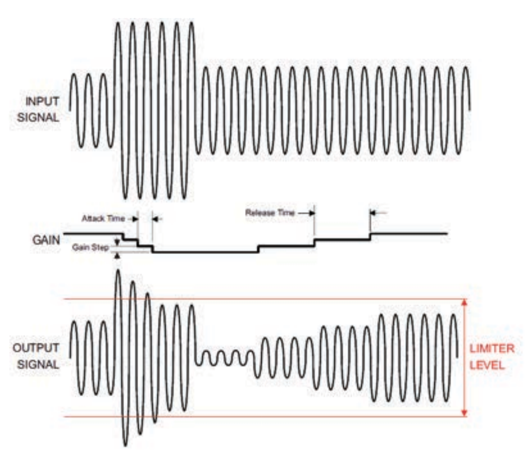


Figure 1: The compression limiter's input and output signals are depicted. (Photo courtesy of NXP Semiconductors)

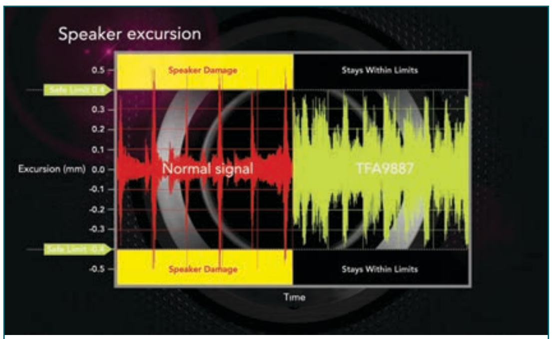


Figure 2: A musical signal is recorded before and after NXP Semiconductor's processing system is used. (Photo courtesy of NXP Semiconductors)

Shawn Scarlett of NXP Semiconductors said the way to get more sound from a speaker is to increase electrical power. Small 0.5-W rated microspeakers can generally handle many times that power amount for very short periods. But, all the extra power has to be released.

Maximizing efficiency converts as much power as possible into sound. However, much of the power is still wasted as heat in the voice coil. This "self heating" is directly related to the current in the voice coil. If the temperature climbs too high, the glue holding the voice coil together can fail (see **Figure 1** and **Figure 2**).

DSP TECHNOLOGY

In so many ways, these processing techniques are compromises, essentially shortcuts. Most people don't want to pay $100 or more for a ticket to a live concert to hear heavily compressed audio. On the other hand, a smartphone's speakerphone function is not going to deliver clear and intelligible communications without psychoacoustic magic nor will stereo Bluetooth speakers reproduce anything resembling bass without some level of processing.

Users' rising expectations of micro-audio device performance is going to continue, and the practical limits of transducer engineering and performance can be dramatically enhanced through signal processing. The continued evolution in the quality and quantity of signal processing found in mobile consumer devices shows no signs of slowing down in the years to come. *aX*

RESOURCE

NXP Semiconductors, www.nxp.com

SOURCES

IPx8 Microspeaker
Knowles Electronics, LLC | www.knowles.com

PurePath digital amplifier and SpeakerGuard speaker protection circuitry
Texas Instruments, Inc. | www.ti.com

MaxxBass plugin
Waves Audio, Ltd. | www.waves.com

Mugen—A Hybrid Amplifier

Design uses a mixture of transistors and valves to achieve audiophile-worthy performance

Valves (i.e., vacuum tubes) are experiencing a revival in the audio world, which is apparent from the number of commercial amplifiers and DIY designs that have appeared in recent years.

Unfortunately, valve amplifiers are relatively costly compared with transistor amplifiers, in part due to the need for a high-voltage supply and output transformers. Output transformers, in particular, are a major investment.

The Mugen Hybrid Amp design replaces the output valves and transformer with a solid-state circuit using modern transistors, which can directly drive a loudspeaker (see **Photo 1**). Valves are used in the input stage.

Photo 1: The Mugen Hybrid Amp got its name from the Japanese word "mugen," meaning infinite or endless. The design was inspired by "Endless," which is a track from the Keith Jarrett Trio's CD, *Changeless*.

DRIVER CIRCUIT

The Mugen amplifier consists of a voltage and a current stage (see **Figure 1**). The voltage stage, which is the driver portion, is built around valves V1 and V2, and it must provide adequate input signal amplification. Here, 20 to 30 dB is a practical figure. The current stage, which is built around transistors Q4 and Q5, enables the amplifier to drive 4- or 8-Ω loudspeakers. The current stage acts as buffer and does not have any gain.

The voltage stage has to supply a solid 25 V_{EFF} to the current stage to drive the amplifier to its maximum output level. A key factor here is that the signal must have sufficiently low distortion since overall negative feedback is not used in this design. The circuit must also be able to drive a 10-kΩ impedance load since the driver circuit sees R11 (20 kΩ) in parallel with the P3 and R16 (20 kΩ) combination. The impedance could be increased by bootstrapping or using MOSFET drivers, but these techniques do not fit with this amplifier's concept.

After designing other projects implemented entirely with valves, I have gained experience with driver stages that must supply output signals with large amplitudes and low distortion. The "long-tailed pair" circuit is exceptionally well suited to this task. I also chose this configuration because it can act as a phase splitter, which enables a certain trick to be used.

The long-tailed pair can be regarded as a differential amplifier that increases the difference between the signals on the two control grids. The input signal is connected to the "left" input. The "right" input is tied to ground, so the output signal is an amplified version of the input signal. This arrangement enables feedback to be connected to the right input, where it will be subtracted from the original signal. The negative feedback reduces the amount of distortion.

I used a JJ Electronic ECC83 high-impedance amplifier triode. The common cathodes of the ECC83's two halves (the US equivalent is the 12AX7) can be regarded as a third input, which provides 6 dB of local negative feedback.

A characteristic of the long-tailed pair is that it has two outputs with opposite phases (180° phase offset). The left anode is "in phase," while the right anode is "out of phase." The long-tailed pair normally has a common cathode resistor, giving it its name. Instead, this design uses a current source. The current source's high internal impedance improves the circuit's characteristics, including the distortion. A trimpot can be used to easily adjust the ECC83's operating current.

Due to its high-amplification factor (100 dB) and excellent availability, the ECC83 is the right choice for this application. The need for high gain can be explained as follows. The long-tailed pair has 6 dB of local negative feedback. A normal cathode-resistor or grounded-cathode amplifier built around an ECC83 can provide a gain of more than 35 dB, and the long-tailed pair circuit used here can provide more than 29 dB.

I originally intended to avoid using overall negative feedback with this amplifier. However, I have included an option for adding 6 dB of overall negative feedback. A jumper/header is provided on the PCB for this purpose. This enables everyone who builds the amplifier to decide for themselves. Even with overall negative feedback,

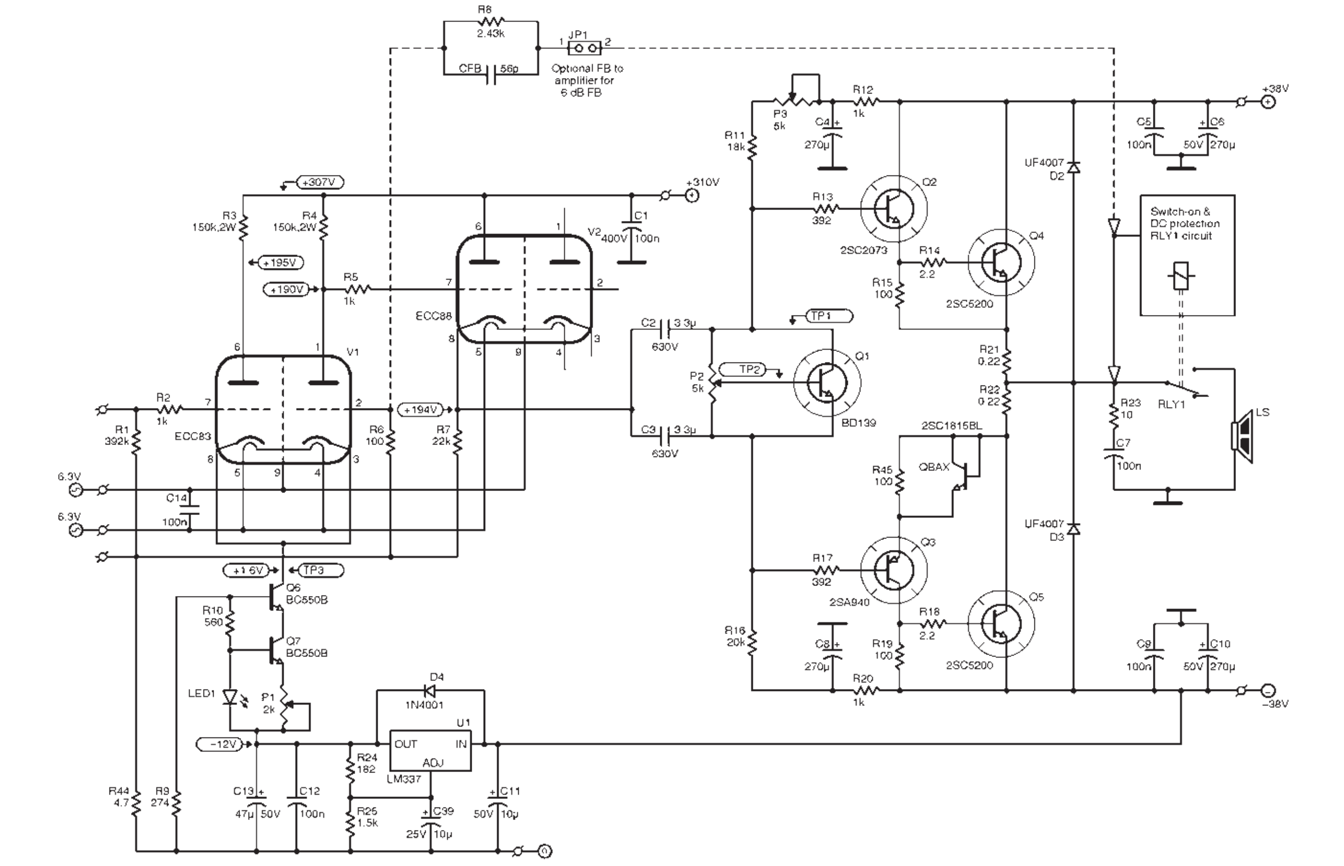


Figure 1: The Mugen Hybrid Amp's voltage stage amplifies the input signal. The current stage helps the amplifier drive its loudspeakers.

the gain is still high enough (23 dB) to provide adequate input sensitivity.

With a normal cathode-resistor circuit, it would not be possible to obtain overall negative feedback by feeding the output signal back to the cathode in the usual manner. The output signal is in phase with the input signal, which causes positive feedback. The long-tailed pair's V1b output signal is out of phase, which makes overall negative feedback possible.

Overall negative feedback is the subject of much debate. I have learned from experience that an amplifier with strong negative feedback has a less open and "pleasant" sound than a design without negative feedback. A 6-dB value represents a good compromise.

A disadvantage of the ECC83 is that it has a relatively high output impedance. Consequently, a cathode follower is included after the ECC83 to provide sufficient drive for the transistor stage. The cathode follower has a low output impedance (less than 500 Ω), compared with around 50 kΩ for the long-tailed pair. After much experimenting, I obtained the best results with an ECC88 in this position. I set the bias to satisfy the ECC88's maximum anode voltage rating (130 VDC). However, the JJ Electronic version of the ECC88, the ECC83, has a 220-V maximum anode voltage rating, the same as the Philips ECC88. The high value of the ECC88's


Photo 2: The amplifier uses a mixture of transistors and valves.

SPECIFICATIONS

(Measured without overall negative feedback)

Input sensitivity:	825 mV (8 Ω) 770 mV (4 Ω)
Input impedance:	300 kΩ
Gain:	29 dB (23 dB with negative feedback)
Output power (1% THD):	70 W into 8 Ω 110 W into 4 Ω
THD + noise	at 1 W/8 Ω: <0.1 % at 10 W/8 Ω: <0.15%
Damping factor:	20 (with 8-Ω load)

cathode resistor enables the ECC83 to be coupled directly to the ECC88. The ECC88 is self-biasing thanks to the large amount of negative DC feedback provided by cathode resistor R7.

An additional advantage of this cathode follower design is that the cathode voltage is 0 V when the circuit is cold and gradually rises to its operating bias level of approximately 194 V as the ECC88 warms. The coupling capacitors are gradually charged during this process so the transistor stage does not experience any spikes.

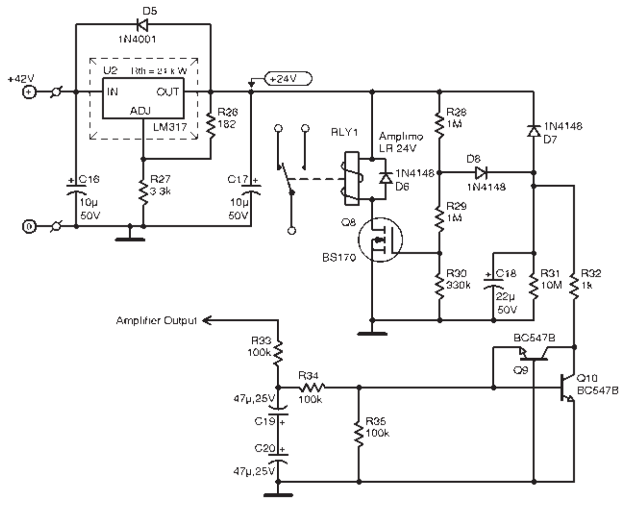


Figure 2: This circuit provides a switch-on delay and DC protection for the output.

Exceptionally good results can be obtained by using a current source in place of cathode resistor R7. When I used an IXYS IXCP10M45 regulator as the current source—a distortion of less than 0.1% (without negative feedback!)—I measured a 45-W output power. However, this IC is difficult to obtain, so I did not pursue this option.

For practical reasons, the Mugen amplifier uses JJ Electronic valves. They are readily available, reasonably priced, good-quality valves that come from current production.

Many people use New Sensor's Sovtek 6N1P as a replacement for the ECC88, but the distortion was not acceptable when I tried a 6N1P. A simple and interesting alternative is to use a Sovtek 5751 in place of the ECC83. These are directly interchangeable types. The amplification factor is slightly lower, but this is not a problem. From an acoustic perspective, I prefer a 5751 (from Philips/ECG or NOS) in combination with a ECC88. If you have adjusted the amplifier for operation with an ECC83, the voltage at TP3 will automatically increase by about 2.2 V if you replace it with a 5751.

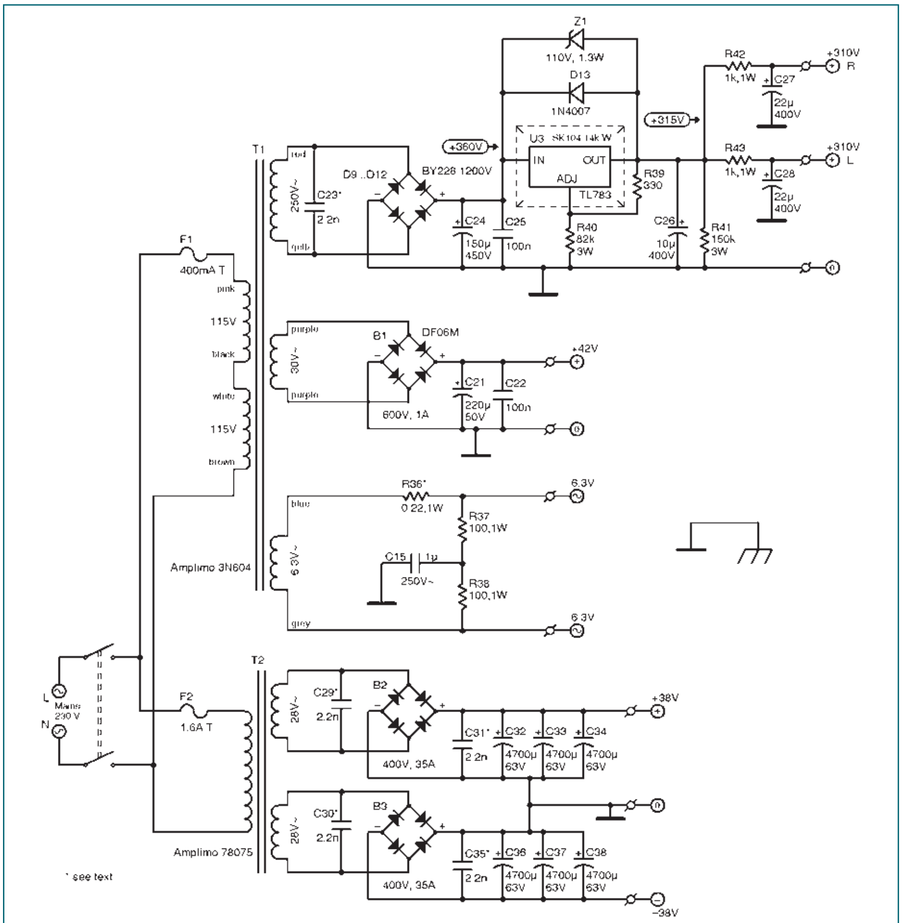


Figure 3: The power supply provides four different voltages.

COUPLING CAPACITORS

The valve and transistor stages are linked by two high-quality coupling capacitors (see **Photo 2**). They cannot be omitted in this design, since the DC voltage on the ECC88's cathode is approximately 194 V. Unfortunately, these capacitors affect the amplifier's ultimate sound.

The capacitors' sound characteristics are the subject of heated debate among audiophiles. Listening tests have clearly shown that these capacitors have an important effect. I finally settled on one from the Clarity-Cap SA series, which has a good price/quality ratio. Thanks to its high working voltage (600 V), the SA series is well suited for use in designs with high voltages (e.g., valve circuits). The PCB layout can also accommodate capacitors from other manufacturers, including WIMA and Solen Electronique. I chose the 3.3-µF value to position the low-frequency rolloff well below 10 Hz. Note that the coupling capacitance combines with the transistor stage's input impedance to form an RC filter with a corner frequency of:

$$\frac{1}{(2\pi \times 3.3\ \mu\text{F} \times 10\ \text{k}\Omega)}$$

the coupling capacitors must have a working voltage of at least 400 VDC.

CURRENT STAGE

The current stage (power stage) is based on bipolar transistors. Although MOSFETS (e.g., the TT electronics

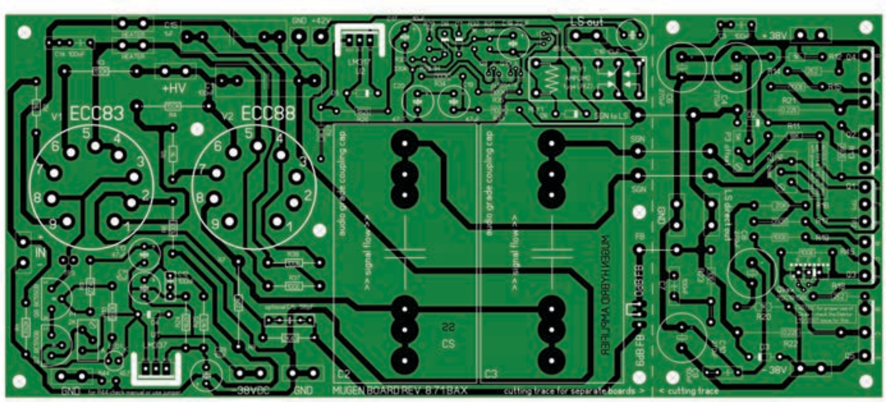


Photo 3: The amplifier board can be divided in two, depending on its arrangement in the enclosure. The component overlay is reproduced at 80% of its actual size.

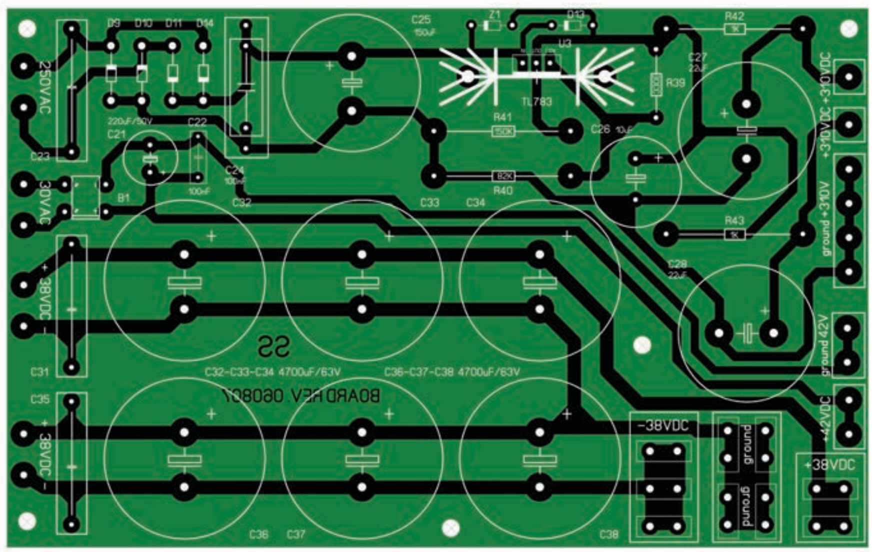


Photo 4: The power supply board is dimensioned for a complete stereo amplifier. The component overlay is reproduced at 80% of its actual size.

Semelab BUZ900P or the Renesas 2SK1058 families) would also be an option. I intentionally did not choose them for this design.

The selected driver transistors are often used in audio amplifiers. They have outstanding characteristics for audio use and they are inexpensive. The output transistors (Toshiba 2SC5200) have excellent characteristics. They are specifically designed for audio applications, readily available (but beware of imitations!), and very robust thanks to their large safe operating area (SOA) range.

The Toshiba 2SC5200 is available in two versions, with an "O" or "Y" suffix. This code designates the transistor parameter that denotes its gain ration (i.e., hFE range). Both types work well, but all the transistors should be the same type. I used the O version in the prototypes and the amplifier's final design.

The current stage is a standard quasi-complementary output configuration (i.e., a configuration with two identical NPN output transistors). This contrasts with the common practice of using a complementary design with an NPN type and a PNP type.

Quasi-complementary output stages were often used in the 1970s and early 1980s because complementary PNP transistors were unavailable or too expensive. This configuration has acquired a bad reputation among many people, but this is not justified. Very good results can be obtained with a quasi-complementary design. The main advantage is that the output transistors are identical. NPN and PNP transistors can never be more than approximately equivalent. This is why manufacturers (e.g., Naim Audio) still use only the NPN/NPN configuration. The ultra-modern Denon PMA1500AE amp also uses a quasi-complimentary NPN output stage, in this case using two UHC n-type FETs. The selected ±38-VDC supply voltage is optimal for this output stage and enables a 4- or 8-Ω load to be driven without any problems.

CIRCUIT DETAILS

Resistor R1 is a grid-leak bias resistor for V1a. Its value is not critical, but the resistor is essential because otherwise the valve would not be able to generate the negative bias that sets its DC operating point. R2 forms a low-pass filter in combination with the ECC83's input capacitance. This prevents any tendency to oscillation. The

COMPONENTS LIST

Amplifier and Power Supply
(for a stereo version, double all the components)

Capacitors
- C1 = 100 nF, 400 VDC
- C2, C3 = 3 µF3, 400 VDC (Clarity-Cap SA 630-V audiograde capacitor)
- C4, C6, C8, C10 = 270 µF, 50 V (Panasonic FC, Farnell #9692436)
- C5, C9, C12, C14, C22 = 100 nF, 50 V
- C7 = 100 nF (Vishay MKP-1834, Farnell #1166887)
- C11, C16, C17 = 10 µF, 50 V
- C13 = 47 µF, 50 V
- C15 = 1 µF, 250 V (e.g., WIMA foil capacitor, see text)
- C18 = 22 µF, 63 V
- C19,C20 = 47 µF, 25 V
- C21 = 220 µF, 50 V
- C23 = 2 nF2 (WIMA FKP-1/700 VAC, see text)
- C29, C30, C31, C35 = 2 nF2 (WIMA FKP-1/700 VAC)
- C24 = 150 µF, 450 V
- C25 = 100 nF, 450 VDC
- C26 = 10 µF, 400 V
- C27, C28 = 22 µF, 400 V
- C32, C33, C34, C36, C37, C38 = 4,700 µF, 63 V (BC056, 30 mm × 40 mm, Conrad Electronics #446286-89)
- C39 = 10 µF, 25 V
- Cfb = 56 pF (optional)

Miscellaneous
- B1 = bridge rectifier 600 V peak-inverse voltage (PIV) at 1 A (DF06M)
- B2,B3 = bridge rectifier 400 V PIV at 35 A
- T1 = mains transformer, secondary 30 V + 250 V + 6.3 V (Amplimo type 3N604)
- T2 = mains transformer, secondary 2 × 28 VAC, 300 VA (Amplimo type 78057)
- RLY1 = output relay 24 V (e.g., Amplimo type LR)
- Heatsink profile for U3 (Fischer SK104 25,4 STC-220 14 kW (e.g., Conrad Electronics #186140-62)
- Heatsink profile for U1 and U2, Fischer FK137 SA 220, 21 kW (e.g., Conrad Electronics #188565-62)
- Heatsink profile for Q4 and Q5, 0.7 kW or better
- Nine-way valve socket (Noval), PCB mount, for V1 and V2
- Amplifier board, #070069-1 (mono), Eurocircuits, www.eurocircuits.com (formerly PCBShop)
- Supply board, #070069-2, Eurocircuits, www.eurocircuits.com

Resistors
(1% metal film, 600 mW unless other rating indicated)
- R1 = 392 kΩ
- R2, R5, R12, R20, R32 = 1 kΩ
- R3, R4 = 150 kΩ, 2W (BC PR02 series)
- R6, R15, R19, R45 = 100 Ω
- R7 = 22 kΩ, 3 W (BCPR03 series)
- R8 = 2 kΩ43
- R9 = 274 Ω
- R10 = 560 Ω
- R11 = 18 Ω
- R13, R17 = 392 Ω
- R14, R18 = 2 Ω2
- R16 = 20 kΩ
- R21, R22 = 0 Ω22, 4 W (Intertechnik MOX)
- R23 = 10 Ω, 2 W
- R24, R26 = 182 Ω
- R25 = 1 kΩ5
- R27 = 3 kΩ3
- R28, R29 = 1 MΩ
- R30 = 330 kΩ
- R31 = 10 MΩ
- R33, R34, R35 = 100 kΩ
- R36 = to be determined (0.22 Ω using 3N604)
- R37, R38 = 100 Ω, 1W (see text)
- R39 = 330 Ω
- R40 = 82 kΩ, 3 W
- R41 = 150 kΩ, 3 W
- R42, R43 = 1 kΩ, 1 W
- R44 = 4 Ω7
- P1 = 2 kΩ, 15-turn preset, T93YB (Vishay) or 3296Y (Bourns)
- P2, P3 = 5 kΩ, 15-turn preset, T93YB (Vishay) or 3296Y (Bourns)

Semiconductors
- D1 = not used
- D2, D3 = UF4007 (if necessary 1N4007)
- D4, D5 = 1N4001
- D6, D7, D8 = 1N4148
- D9, D10, D11, D12 = BY228
- D13 = 1N4007
- LED1 = LED, 5 mm, red
- Z1 = Zener diode, 110 V, 1.3 W
- Q1 = BD139
- Q2 = 2SC2073
- Q3 = 2SA940
- Q4, Q5 = 2SC5200
- Q6, Q7 = BC550B
- Q8 = BS170
- Q9, Q10 = BC547B
- Q_{BAX} = 2SC1815BL
- U1 = LM337
- U2 = LM317
- U3 = TL783 (Conrad Electronics #175012-62)

Valves
- V1 = ECC83 (JJ Electronic)
- V2 = ECC88 (JJ Electronic)

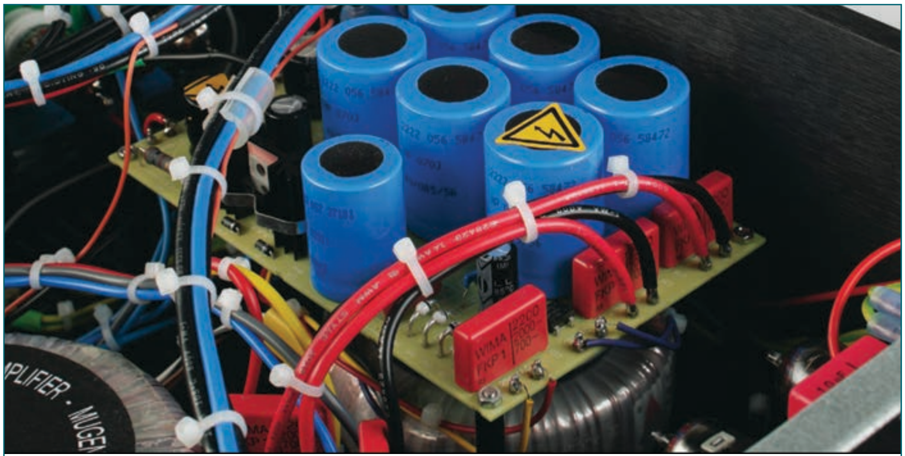


Photo 5: The amplifier's power supply is shown here.

same thing applies to R5 in combination with the ECC88. Anode resistors R3 and R4 are dimensioned to yield a voltage of slightly more than 190 V on V1's anodes. V1 has the right bias with a 0.8-mA anode current. The power dissipation is well within the permitted value.

The long-tailed pair with V1 uses a current source built around Q6 and Q7. The LED provides a reference voltage and the current can be easily set with P1. The total current is equal to approximately 1/P1. A separate power supply, using an LM377, provides a −12-V voltage for the current source.

The overall negative feedback is applied to V1b's control grid. As I previously mentioned, I chose a 6-dB value. This value was determined by the ratio of R8 and R6. A small capacitor (56 pF) can be connected across the feedback resistor to increase stability.

I chose the ECC88's bias to generate an anode current of approximately 9 mA with an effective anode voltage of around 115 VDC. The power dissipation is 1 W, which is beneficial for the valve's service life. The total distortion would be slightly less at a higher current, but the valve's life would be reduced by the higher dissipation.

Q1 sets the output transistors' quiescent current. It must be fitted close to the output transistors to achieve good temperature stability. Minimum quiescent current is obtained when the P2's wiper is turned fully toward the Q1's collector. P2 must be a good quality 10-turn potentiometer.

The R11/P3 pair and R16 ensure the amplifier output's DC stability. These components' values also determine circuit's input impedance, which is approximately 10 kΩ (20 kΩ || 20 kΩ). These values could be increased if MOSFETs were used, but this is not possible here due to the amount of base current required by Q2 and Q3. R12/C4 and R20/C8 are additional decoupling networks that are indispensable. C4 and C8 can also be 220 µF or 330 µF, if desired.

P3 enables the output stage's DC offset to be set to zero. I intentionally did not use an active DC offset control in the form of an op-amp integrator because I think this affects the amplifier's sound quality.

Q2 and Q4 form a Darlington pair that provides adequate current gain, as do Q3 and Q5. Q3 and Q5 form what is called a "Sziklai pair," which is used here to mimic a PNP transistor. Quasi complementary circuits normally use a "Baxandall diode" to improve symmetry and linearise the response. This approach was used in the Ekwa amplifier published in *Elektor* in 1972.

My design uses a transistor configured as a diode (Q_{BAX}) instead of a normal diode. The measured distortion at 1 W was 0.22% with a diode in the circuit. The value with a 2SC1815 configured as a diode was 0.08%. Note that the 2SC1815's BL version should be used here. Although the -O, -Y, and -GR versions can also be used, their results are practically the same as those with a normal diode. The difference gradually decreases at levels greater than 5 W.

The PCB is designed to enable a 2SC2073 or a Vishay 1N4007 transistor to be used instead. This is also the order of preference. Obviously, only one of these three component types can be fitted on the board.

Thanks to the inherent local negative feedback, the output stage is very stable with regard to temperature drift and quiescent current. The emitter resistors should preferably be Intertechnik MOX types. They are noninductive and have relatively small dimensions. The amplifier output has a Zobel network built around R23 and C7, which ensures stability above 100 kHz.

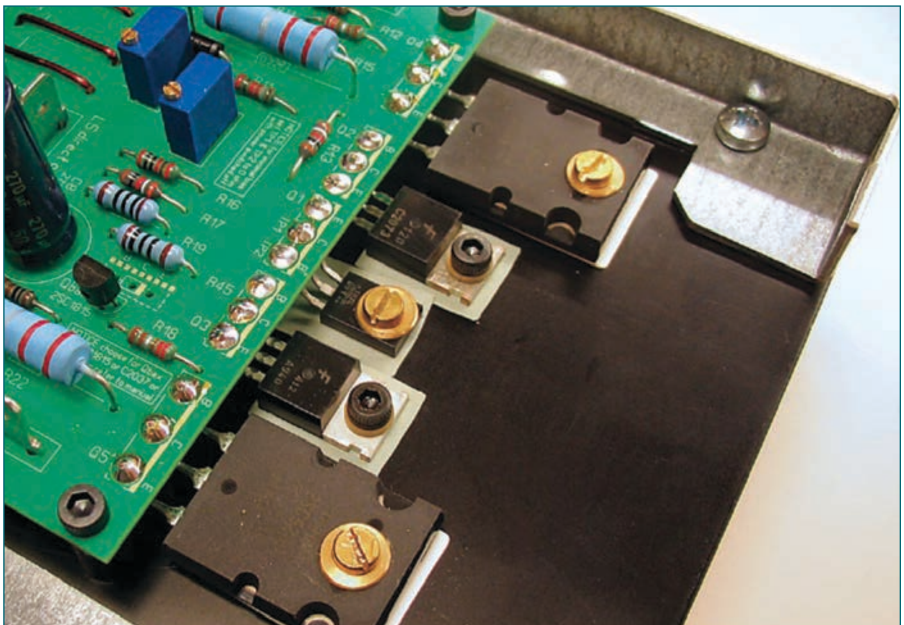


Photo 6: The power stage transistors are fitted to the heatsink first.

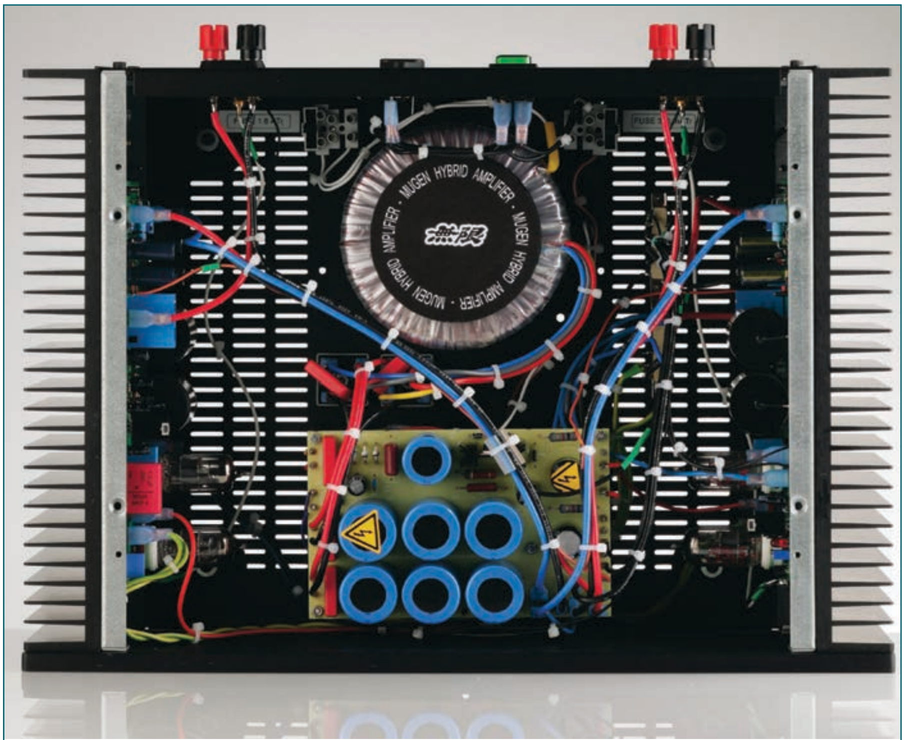


Photo 7: Once assembled, the amplifier's hybrid design can be fully appreciated.

Base resistors (R13, R17, R14, and R18) are used for all transistors in the output stage to prevent oscillation. The resistors for the driver transistors (R13 and R17) are essential.

The heatsink extrusion for each output stage must be rated at 0.7 kW or less to ensure reliable operation.The switch-on delay and DC protection circuit is built around relay RLY1 and MOSFET Q8 (see **Figure 2**). This circuit was previously used in Bob Stuuman's "Valve Final Amp" design published in *Elektor Electronics*, April/May 2003.

The switch-on delay is approximately 30 s. If a hazardous DC voltage is present at the output, the relay will disconnect the amplifier output from the loudspeaker. I used an Amplimo relay with special contacts that make it especially suitable for use as an output relay in audio amplifiers.

A coil can optionally be fitted in series with the output to make the amplifier more general-purpose with respect to the speaker's possible capacitive behavior. This coil is omitted in this amplifer's design. A DIY coil with a 4-μH inductance consisting of 16 turns of 0.75-mm enamel copper wire wound on a 6.3-mm drill bit can be used. A 15-Ω/2-W resistor must be fitted inside the coil and soldered across the coil.

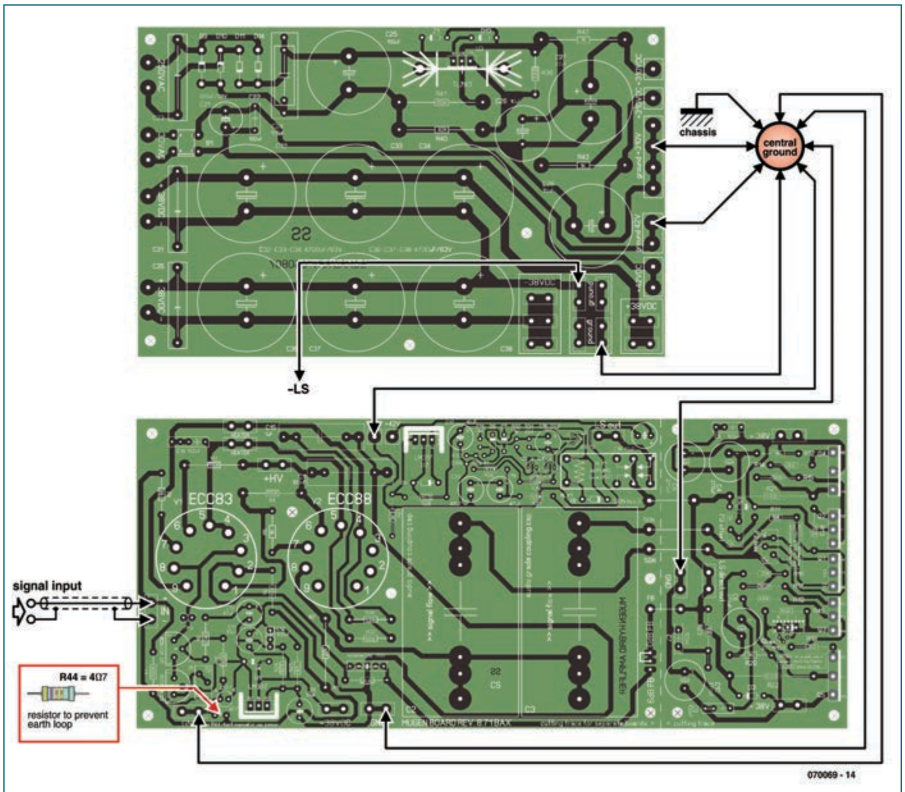


Photo 8: The ground connections must be arranged this way in the enclosure. They are all tied to a single ground point connected to the enclosure.

POWER SUPPLY

The high-voltage supply uses a Texas Instruments (TI) TL783 voltage regulator IC (see **Figure 3**). The TL783's input voltage must be approximately 360 VDC for proper operation. An Amplimo toroidal high-voltage transformer provides this voltage in a somewhat unorthodox manner. The 250-V winding is so generously dimensioned that it is hardly loaded by the ECC83s and ECC88s, so the secondary voltage is a considerably higher than the rated 250 V. Bear this in mind if you use a different transformer. The TL783 is fitted with a small heatsink and must be mounted insulated.

Voltage divider R39/R40 sets the output voltage to around 315 V. Resistor R41 is included to discharge the electrolytic capacitors when the amplifier is switched off. R40 and R41 must be 3 W. R42/C27 and R43/C28 are additional RC filters for the left and right channel, respectively. The high voltage for V1 and V2 is approximately 310 VDC. If you cannot find a WIMA FKP1 for C23 as specified in the components list, omit it.

Transformer T1's 30-V winding is used for the switch-on delay and protection circuit. The AC filament voltage is tied to ground via a capacitor. In this case, it cannot be connected directly to ground. This is because the ECC88's cathode is not close to ground potential here, but instead at 195 V. The capacitor arrangement enables the maximum cathode–filament voltage rating to be met. This floating filament supply works well in practice. A 0.47-μF value can be used instead of 1 μF with equally good results.

The R36 value must be experimentally determined. This resistor determines the filament voltage's value, which must be close to 6.3 V.

The power supply is suitable for stereo use, but it can also be used for a mono-final amplifier. If it is used for a stereo version with a single transformer and a single supply PCB, then R37, R38 and C15 only have to be fitted on one of the two amplifier boards. However, fitting them on both boards will not do any harm.

The ±38-V supply is simple but effective. A toroidal transformer with a secondary 2 × 28 VAC voltage provides the best results in terms of output power. You can use a different transformer—one with a more conventional 2 × 25 VAC value can be used—but the maximum output power will be somewhat less. The 2.2-nF WIMA FKP-1 capacitors provide additional decoupling.

CONSTRUCTION

Photo 3 and **Photo 4** show the PCBs for the amplifier and the power supply. The actual size layouts can be downloaded from the Elektor website, or you can order ready-made boards from Eurocircuits (formerly PCBShop). The board shown in **Photo 3** is for a mono amplifier, so you need two amplifier boards and a power supply board for a stereo version.

The components list provides several parts with specific descriptions or type numbers. Based on my experience, you will obtain the best results from the amplifier if you use these components. However, you are free to experiment with comparable parts.

Assembling the power supply board is straightforward. Use good-quality blade connectors for the various supply and ground terminals. This makes it easy to wire the amplifier. After the power supply board is complete, you can assemble the amplifier boards.

The amplifier board is designed so that it can be split in two to mount the power stage on the heatsink and the driver stage somewhere else (e.g., on the enclosure's base). However, the wiring between the two parts must be kept as short as possible (see **Photo 5**).

Photo 6 shows how the power stage's transistors are fitted (all insulated!). For best results, first fit the transistors to the heatsink, bend their leads at right angles, and then use screws to secure the board to the heatsink. Do not solder the transistors in place until everything is properly positioned.

I used an enclosure with two large heatsinks on the sides for the prototype (see **Photo 7**). It is big enough to hold a complete amplifier board.

The two supply transformers and the bridge rectifiers for the ±38-V supply are fitted in the middle of the enclosure's base. The supply board is located above transformer T1.

The amplifier and supply boards have several ground connections. They must all be connected separately to a single star point (see **Photo 8**).

To avoid ground loops, the grounds of the ±38-V supply, the +42-V supply, and the 310-V supply are not joined on the supply board. The 4.7-Ω resistor (R44) between the input neutral terminal and circuit ground is optional and can be replaced by a wire link. However, in the prototype this resistor proved to be necessary to keep the overall arrangement free of hum.

Use plastic standoffs to mount the circuit boards. Metal types can cause shorts between PCB tracks and the heatsink or chassis.

A mains entry unit, a double-pole mains switch, a pilot light, and a pair of fuse holders for the transformers can be fitted on the primary side. Use caution when working with electricity.

ALIGNMENT

Inspect all components and connections before switching on the amplifier. Check that the transistors are insulated from the heatsink and from each other, check the polarity of the electrolytic capacitors, and confirm that the right valves are fitted in the sockets. The ECC83 and ECC88 are absolutely not electronically interchangeable.

The amplifier has three adjustment points:

- P1 sets the ECC83's operating current
- P2 controls the output transformers' quiescent current
- P3 adjusts the output's DC level

Before switching on the amplifier, ensure that P2's wiper is at the end connected to Q1's collector.

This results in minimum quiescent current. Test points TP1 and TP2 are provided for this purpose on the PCB. Adjust potentiometer P1 to a value of approximately 800 Ω before soldering it to the board.

After switching on the amplifier, adjust P1 so the DC voltage at TP3 is 1.6 V. The exact value is not critical, but the DC voltage measured across R7 must be close to 195 V (±5%). If necessary, readjust P1 to obtain this value. The anode voltage of V1b should be about 190 V. These three voltages are interrelated.

After this, adjust P2 and P3 with no input signal and no load. P3 controls the output offset. The DC voltage measured at the output must lie between 50 and –50 mV. It varies slightly, which is normal. Then adjust P2 to set the quiescent current. The DC voltage across emitter resistor R21 or R22 must lie between 22 and 33 mV (for a quiescent current of 100 to 150 mA). After the amplifier has warmed up for approximately 15 min, check all the values again and adjust the settings as necessary.

You can repeat this procedure several times during the first hour. In between these adjustment cycles, you can use an inexpensive loudspeaker (e.g., a PC speaker) and a bit of music to test run the amplifier.

KEY POINTS

Be careful! High voltages are present at various places on the circuit boards. Remember that residual voltages can be present for a while after the amplifier is switched off.

Be kind to your loudspeakers. Never connect or disconnect inputs or interlinks unless the amplifier is switched off.

RESULTS

Despite the fact that overall negative feedback is not used in this amplifier, it has relatively low distortion. The distortion is less than 0.1% at low power levels. This respectable value is the result of careful component selection and dimensioning.

The damping factor is also suitable for practical use. This is often a problem with final amplifiers that do not use negative feedback.

An amplifier's sound characteristics are often difficult to express in words. The amplifier can create a splendid sound stage, the lows are controlled, and the dynamic behavior is convincing. The listening pleasure is also very good.

The Mugen has an honest character without any signs of an exaggerated "valve sound" (i.e., coloration). By combining a valve driver stage with a transistor power stage, the Mugen amplifier offers the best of both worlds at an attractive price. *aX*

Editor's Note: This article first appeared in Elektor*, October 2007.*

RESOURCES

Elektor, PCB layout, www.elektor.com/magazines/2007/october/mugen-a-hybrid-audio-amplifier.254273.lynkx.

Eurocircuits (formerly PCB-Shop), ready-made PCBs, www.eurocircuits.com.

B. Stuurman "Valve Final Amp," *Elektor Electronics*, April/May 2003.

SOURCES

3N604 and 78057 transformers
Amplimo | www.amplimo.nl

Clarity-Cap SA capacitors
Clarity-Cap | www.claritycap.co.uk

PMA1500AE amplifier
Denon | www.denon.com

MOX resistors
Intertechnik | www.intertechnik.com

IXCP10M45 High-voltage regulator
IXYS Corp. | www.ixys.com

ECC83, ECC88, and 12AX7 high-voltage regulator vacuum tubes
JJ Electronic | jj-electronic.com

Sovtek 6N1P and Sovtek 5751 preamp vacuum tubes
New Sensor Corp. | www.newsensor.com

2SK1058 MOSFET
Renesas Electronic Corp. | www.renesas.com

TL783 Voltage regulator
Texas Instruments, Inc. | www.ti.com

2SC5200 and 2SC2073 transistors
Toshiba | www.toshiba.com

BUZ900P MOSFET
TT electronics Semelab | www.semelab.co.uk

1N4007 Transistor
Vishay Intertechnology, Inc. | www.vishay.com

FKP1 Capacitor
WIMA | www.wimausa.com

By Gary Galo (United States)

OPPO Digital's New 3-D Blu-ray Disc Player Raises the Bar

BDP-105 Universal Audiophile 3-D Blu-ray Disc Player

OPPO Digital, Inc.
2629 Terminal Blvd., Suite B
Mountain View, CA 94043
(650) 961-1118
www.oppodigital.com

Price: $1,199

OPPO Digital recently upgraded its line of universal digital players. The affordable BDP-93 (*audioXpress*, June 2011) was replaced by the BDP-103, which costs $499. The BDP-105, which is the subject of this month's review, replaces OPPO's top-of-the-line BDP-95 (reviewed in *audioXpress*, January, 2012).

With the BDP-95, OPPO moved beyond its previous "great for the money" players, offering a universal player with true high-end audio performance. But, the company has not rested on its laurels. Although priced $200 higher than the BDP-95, the added features and improvements in audio performance make the BDP-105 worth the cost increase.

The BDP-105 is not a modification or simple upgrade of the BDP-95. Although it shares much in common with its predecessor, the BDP-105 is an entirely new design (see **Photo 1**).

The dedicated two-channel stereo audio section, in particular, has been completely revamped and is now housed on its own PC board. The BDP-105 is truly universal and supports all standard 4.75" optical media, video, and audio. These include DVD-video, Blu-ray, CD, HDCD, DVD-Audio, SACD, and all standard recordable CD, DVD, and Blu-ray formats. All popular file formats are also supported, including WAV, FLAC, AICC, and MP3, which can be played from discs, or external storage devices plugged into one of the player's USB inputs. A recent firmware upgrade also enables playback of stereo and multichannel direct-stream digital (DSD) files from external devices or discs.

My player had no problems with DSD64 files at the standard 2.8224-Mb/s data rate, but I found that the DSD128 files at a 5.6448-Mb/s data rate played at half speed, even with my television turned off (to ensure native-mode DSD playback rather than conversion to 88.2-kHz/24-bit PCM). I downloaded these files on the 2L website's free sample page. Hopefully a future upgrade will support DSD128 files.

Perhaps the most important functional improvement in the new player—at least for audiophiles—is its ability to operate as a DAC with external digital sources. The BDP-105's five digital inputs are routed to the player's excellent DACs and analog circuitry. These include USB (2.0-compatible), S/PDIF coaxial, Toslink optical, and two HDMI inputs (one on the front panel and one on the rear). The coaxial and optical inputs will accept PCM datastreams up to 96 kHz/24 bit. The USB and HDMI inputs will accept PCM up to 192 kHz/24 bit. The HDMI inputs will also accept up to 7.1-channels in all the standard A/V audio formats (e.g., Dolby Digital, Dolby Digital Plus, DTS, and AAC).

The flexible array of digital inputs enables the BDP-105 to function as a digital music server, conventional DAC, and stand-alone player. Windows 8, 7, Vista, and XP users will need to download a driver to use the BDP-105 as an asynchronous USB DAC with their computers. The driver and installation instructions can be downloaded from the OPPO website. Just click on "Customer Service" and "Knowledge Base." No driver is needed for Mac OS X, but configuration instructions are provided

a)

Photo 1: The BDP-105 is the latest flagship universal player from OPPO Digital. Compared to its predecessor, the BDP-95, the new player offers many new video and audio features, along with improved audio performance (a). The BDP-105's rear panel features dedicated two-channel audio outputs plus digital audio inputs, enabling other digital audio sources to take advantage of the player's excellent D/A conversion and analog circuitry. A generous assortment of HDMI and USB 2.0 inputs enables the player to process various A/V sources (b).

MANUFACTURER SPECIFICATIONS

Analog Audio Characteristics**

- Frequency Response: (RCA) 20 Hz–20 kHz: ± 0.2 dB, 20 Hz–96 kHz: –1.5 dB; (XLR) 20 Hz–20 kHz: ±0.3 dB, 20 Hz–96 kHz: –1.5 dB
- SNR: >130 dB (A-weighted, auto-mute), >115 dB (A-weighted, no auto-mute)
- THD+N: <0.0003% or –110 dB (1 kHz at 48k/24b, 0 dBFS, 20 kHz LPF); <0.0017% or –96 dB (1 kHz at 44.1 k/16 b, 0 dBFS, 20 kHz LPF)
- Output Level: 2.1 V_{RMS} (RCA) or 4.2 V_{RMS} (XLR) at 0 dBFS
- Dynamic Range: >110 dB
- Channel Separation: >110 dB

BD Profile

- BD-ROM Version 2.5 Profile 5 (also compatible with Profile 1 Version 1.0 and 1.1)

Disc Types*

- BD-Video, Blu-ray 3-D, DVD-Video, DVD-Audio, AVCHD, SACD, CD, HDCD, Kodak Picture CD, CD-R/RW, DVD±R/RW, DVD±R DL, BD-R/RE

General Specifications

- Power Supply: ~115 V/230 V, 50/60 Hz AC
- Power Consumption: 55 W (Standby: 0.5 W in Energy Efficient mode)
- Dimensions: 16.8″ × 12.2″ × 4.8″ (430 × 311 × 123 mm)
- Mass: 17.3 lb (7.9 kg)
- Operating Temperature: 41°F–95°F (5°C–35°C)
- Operating Humidity: 15% –75%; no condensation
- Parts and Labor Warranty: 2 Years

Headphone Audio Characteristics**

- Frequency Response: 20 Hz–20 kHz: ±0.3 dB into 300 Ω
- Signal-to-Noise Ratio: >98 dB into 300 Ω (A-weighted, no auto-mute)
- THD+N: <0.001% or –100 dB into 300 Ω (1 kHz at 48 k/24 b, 0 dBFS, 20 kHz LPF)
- Output Power (per channel): 17 mW into 600 Ω, 34 mW into 300 Ω, 63 mW into 150 Ω, 77 mW into 120 Ω, 120 mW into 60 Ω, 187 mW into 32 Ω (1 kHz at 0 dBFS)
- Dynamic Range: >110 dB
- Channel Separation: >90 dB

Internal Storage

- 1 GB (Actual space available for persistent storage varies due to system usage)

Inputs

- HDMI Audio: Dolby Digital, Dolby Digital Plus, DTS, AAC, up to 5.1 channel/192-kHz or 7.1 channel/96-kHz PCM
- HDMI Video: 480i/480p/576i/576p/720p/1080i/1080p/1080p24/1080p25/1080p30, 3-D frame-packing 720p/1080p24
- MHL Audio: Dolby Digital, Dolby Digital Plus, DTS, up to 5.1 channel/192-kHz PCM
- MHL Video: 480i/480p/576i/576p/720p/1080i/1080p24/1080p25/1080p30, 3-D frame-packing 720p/1080p24
- USB Audio: up to two channel/192-kHz PCM
- Coaxial/Optical Audio: Dolby Digital, DTS, AAC, up to two channel/96-kHz PCM

Outputs

- Analog Audio: 7.1 channel, 5.1 channel, stereo
- Dedicated Stereo Analog Audio: XLR balanced, RCA single-ended
- Coaxial/Optical Audio: Dolby Digital, DTS, up to two-channel/192-kHz PCM
- HDMI Audio: Dolby Digital, Dolby Digital Plus, Dolby TrueHD, DTS, DTS-HD high-resolution, DTS-HD Master Audio, up to 7.1 channel/192-kHz PCM, up to 5.1 channel DSD
- HDMI Video: 480i/480 p/576i/576p/720p/1080i/1080p/1080p24/4K × 2K, 3-D frame-packing 720p/1080p24

*Compatibility with user-encoded contents or user-created discs is on a best-effort basis with no guarantee due to the variation of media, software, and techniques used.

**Nominal specification

on the same webpage. Currently, Linux computers are not supported.

OPPO added a headphone jack to the BDP-105. The headphone output takes advantage of the 32-bit digital volume control built into the ES-9018 DAC chip, but the headphone volume control operates independently from the main analog outputs. By default, the headphone volume is set to 50% at turn-on, but this can be changed in the set-up menu. When you plug in headphones, the analog line outputs are automatically muted and the headphones smoothly fade up to normal volume from silent. When the headphones are unplugged, the main analog outputs fade back to their normal levels. This thoughtful feature is easy on your ears and your equipment.

NEW VIDEO FEATURES

Several new video features have also been added to the BDP-105, including the ability to upscale all video sources to a 4k (3840 × 2160) output resolution, which is four times faster than full HD 1080p. To take advantage of this feature, you'll need a television capable of 4k resolution, of course. If you own a 3-D television, standard DVD, Blu-ray and other 2-D video content can be converted to 3-D by simply pressing the remote's 3-D button. The depth and eye convergence levels can be adjusted to personal taste.

The BDP-105's two previously mentioned HDMI inputs can also be used to connect external video sources, including set-top boxes and network streaming devices. This enables other HD video sources to take full advantage of the OPPO's advanced video and audio processing capabilities. The front-panel HDMI input doubles as a Mobile High-Definition Link (MHL), enabling connection of mobile devices including smartphones and tablets.

OPPO has also added a third USB 2.0 input to the BDP-105—there's one on the front panel and two on the rear. The three USB inputs are in addition to the previously described dedicated USB DAC input. OPPO has dispensed with the eSATA port included on the BDP-95's rear panel. Because USB 2.0 is now the industry standard for external storage device connection, eSATA support has become unnecessary.

OPPO has also eliminated the NTSC composite and component video outputs, but it has included a composite diagnostics video output for troubleshooting, which only works with the set-up menu. If you are unable to get a picture via HDMI, this "lowest common denominator" connection enables you to go through the video set-up procedure to find your television's correct HDMI settings.

Digital media player (DMP) and digital media renderer (DMR) capabilities enable wired or wireless access of audio, picture, and video files stored on DLNA-compatible media servers, including computer and network storage devices. The BDP-105 is also equipped with an experimental feature that can access video, audio, and picture files shared by computers on a local network via the server message block

(SMB) or common Internet file system (CIFS) protocols.

OPPO has added to the BDP-95's array of media support. The BDP-105 supports CinemaNow streaming and Rhapsody Online Music, along with everything previously supported by the BDP-95. The player can also connect to Gracenote Music ID and VideoIDTM, displaying cover art, title, artist, genre, and other media information for CDs, DVDs, Blu-ray discs, and a range of digital media files.

Free remote control applications are available from the App Store for Apple iOS devices and the Google Play Store for Android devices (see the Resources section at the end of this article). The BDP-105 uses the Marvell Kyoto-G2H video processor—the new "H" version incorporates the latest generation Qdeo technology.

REMOTE

The BDP-105's remote control is compatible with the BDP-95. Nearly all the buttons are in locations familiar to previous OPPO users. To accommodate the digital audio inputs, the old Source button has been replaced with an Input button, which activates the input selection mode. The Up and Down arrow keys, along with the Enter button, can be used to select the desired digital input. Dedicated buttons to access Netflix and Vudu are also included on the new remote, but a couple of others have been removed to make room for them. The discrete On and Off buttons have been eliminated, but they were never really necessary since the single red Power button toggles those two functions, anyway. The video output mode—PAL, NTSC, or Multi—must now be accessed in the Setup menu. A couple of other functions, including Angle, have been incorporated into a single Option button.

AUDIO DESIGN

The BDP-105 has two analog PC boards. One is dedicated to the two-channel stereo audio circuitry, including the headphone amplifier, and also houses a pair of rectifier bridges, raw supply filtering, and power supply regulation for the two-channel circuitry (see **Photo 2**). The multichannel audio circuitry is housed on its own PC board (see **Photo 3**).

Like its predecessor, the BDP-105 uses two eight-channel ES9018 DAC chips, which is ESS's flagship Sabre32 DAC chip that incorporates its patented Hyperstream architecture. One ES9018 is used for the multichannel board and another for the two-channel board. But, OPPO has made some significant changes in the way the ES9018 DAC and analog circuitry are configured for the dedicated two-channel outputs.

In the BDP-95, four channels of the ES-9018 are stacked (paralleled) for both the left and right stereo outputs. The same paralleled sections feed the unbalanced RCA and balanced XLR output circuitry. In the BDP-105, the ES9018 DAC dedicated to the stereo outputs is configured as follows: One pair of DAC channels feeds the RCA stereo outputs, a second pair feeds the stereo XLR balanced outputs, and two pairs are stacked for the headphone output.

There's a valid technical reason for stacking two DAC sections for the headphone output. Paralleling DAC sections results in higher output current, which enables OPPO to use a lower-value I/V conversion resistor. The lower-value resistor has less thermal noise. This lowers noise at the headphone output where the improvement will be most beneficial.

I was curious why OPPO abandoned the parallel DAC arrangement used for the stereo outputs in the BDP-95. In response to my inquiry, OPPO's Lanping Deng and Jason Liao offered the following response: "In the BDP-105, we initially tried to use the same stacked DAC design, but found that the performance became worse. The addition of the USB, coaxial, and optical DAC inputs, and the headphone output, made it impossible to design an optimized PCB layout if we continued to stack the DAC channels. The sub-optimal layout caused degraded performance. After many attempts and revisions, we decided to no longer stack the DAC channels. The new configuration allows us to create a clean layout that minimizes interference and crosstalk. We lose the benefit of the lower thermal noise from using a smaller I/V resistor, but by beefing up the power supply and separating the stereo and multichannel boards, we were able to lower the noise floor and maintain excellent performance. The BDP-105's stereo audio specification is listed to be the same as the BDP-95, and our internal test results actually show that the BDP-105 is slightly better."

In the detailed description of its CD-1 CD player, Parasound notes that "a single DAC is inherently free of any inter-DAC delays" (it uses a single Analog Devices AD1853 DAC for this player, in the stereo mode). Parasound also states: "The delay between multiple DACs working in parallel can measure up to 10 ns (10,000 ps), introducing minute amounts of delay into the signal chain."[1]

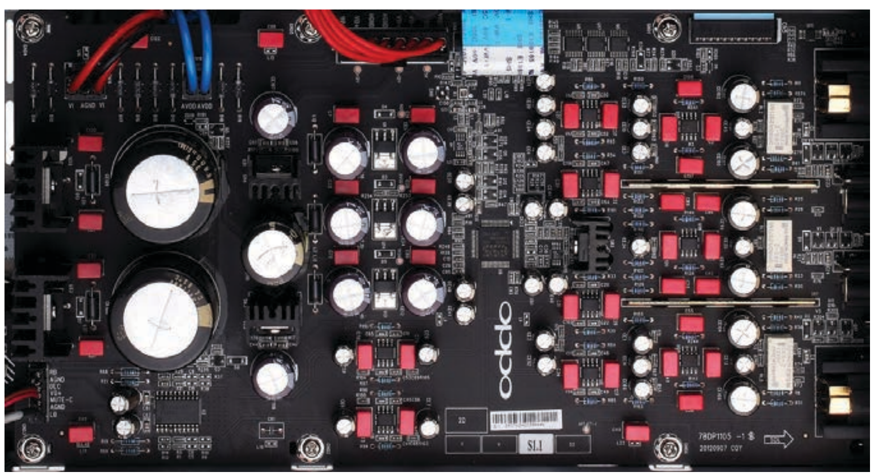


Photo 2: A close-up view of the dedicated two-channel audio board is shown. The board includes an ES9018 DAC chip, a linear supply regulation, a headphone amplifier, and analog circuitry for the balanced and unbalanced stereo outputs.

I asked OPPO if this might explain why the ES9018 actually performs better without stacking sections together. OPPO stated that it has not observed any time delay differences between multiple DAC channels. OPPO's audio design consultant had raised this issue with ESS, but was assured that time delay differences are not a problem with the ES9018. In fact, the ES9012 chip is an internally stacked version of the ES9018. So, it would appear that time delay would not account for the improved performance OPPO has observed with the ES9018's new, non-stacked configuration.

The dedicated stereo outputs' analog circuitry uses the same high-performance ICs as the BDP-95. For each of the unbalanced stereo outputs' channel, a Texas Instruments (TI) LM4562 dual op-amp provides differential I/V conversion. Half of another LM4562 is used as the output amplifier. For each of the balanced stereo outputs, an LM4562 provides differential I/V conversion and a TI LME49724 fully differential op-amp is used as the output amplifier. A TI TPA6120A is used as the headphone amplifier. It has the same excellent current-feedback chip that delivered impressive performance in the Parasound Zdac (reviewed in *audioXpress*, July 2013).

Another pair of LM4562 op-amps provide differential I/V conversion ahead of the headphone amplifier. The multichannel outputs are similar to those in the BDP-95. Each ES9018 DAC channel on the multichannel PC board feeds an LM4562 differential I/V converter. Half of a second LM4562 is used as the output amplifier.

Photo 3: The BDP-105's multichannel audio board is located in the upper left. The dedicated two-channel audio board is on the right. The video circuitry and its switching power supply are fully shielded and located in the bottom portion of the player. The unit features excellent construction quality and a robust steel chassis.

AC COUPLING

The BDP-95's analog circuitry was DC-coupled, but OPPO has capacitor-coupled the BDP-105's multichannel and stereo outputs. I was curious why OPPO chose AC coupling for the new player, since my measurements on the BDP-95 showed the DC levels on all analog outputs—both the multichannel and stereo—to be less than 2 mV, which is a completely safe value.

OPPO noted that a few BDP-95 players came back with damaged audio boards. Its investigation led the company to believe the damage was the result of direct current fed back through the output connectors. OPPO also had a few reports of distorted audio that seemed to be related to the DC coupling.

In one case, the company asked the user to try its interconnect cables with coupling capacitors in the signal path. This eliminated the distortion. OPPO said these few isolated cases were probably the result of DC at the input of the users' A/V receivers or amplifiers. (I agree that this is the most likely explanation.) So, OPPO decided the safest solution for the BDP-105 would be AC-coupling of the output.

For output coupling, OPPO chose the ELNA Silmic II capacitors to ensure the best possible audio performance. The Silmic II capacitors are electrolytic types, which may surprise some readers. (I admit that I was taken aback, at first.) But, these capacitors are a new type of electrolytic design, and I urge you to keep an open mind before jumping to conclusions.

ELNA's website has a datasheet and a detailed description of the Silmic II capacitors. Since there are several translation issues with ELNA's English text, I have taken the liberty of paraphrasing its product description.

The Silmic capacitors were designed specifically for high-performance audio applications and use an entirely new type of electrolytic paper. The new paper's primary ingredient is silk fiber, mixed with Manila hemp fiber to increase strength. Since silk is a relatively supple material, problems associated with mechanical vibration are mitigated. The silk also results in an equivalent series resistance (ESR) that is 20% lower than capacitors made with hemp fiber alone.

ELNA also uses "anode growth foil with more un-etched parts and a 55-µm low-multiplier high-purity cathode foil in order to improve the signal propagation." No permeable metal is used in these capacitors—all leads are oxygen-free copper wire.

ELNA notes that aural evaluations of these capacitors show a significant reduction in high-frequency glare and midrange roughness, with an improvement in low-frequency power and richness. ELNA summarizes by stating that the Silmics yield a "powerful yet mellow sound" that was not previously possible with aluminum electrolytic capacitors.

For a given value and voltage, these capacitors are physically larger than garden-variety electrolytics. OPPO uses 100-µF/16-V capacitors that measure 10 mm × 12.5 mm. But, the large value should ensure excellent low-frequency performance with any load impedance likely to be encountered in A/V receivers, preamps, and amplifiers. OPPO uses these capacitors without any film bypassing, so they may actually perform better on their own.

POWER SUPPLY

OPPO has paid special attention to the BDP-105's power supplies. Like the BDP-95, all audio circuitry—from the DAC chips to the output amplifiers—is powered by a dedicated linear power supply. The power transformer bears OPPO's logo and part number (the BDP-95 transformer has a Rotel label). Raw filtering consists of a pair of ELNA Tonerex 6,800-µF/35-V electrolytic capacitors.

An LM 317/337 pair, in TO-220 cases mounted on heatsinks, are the analog circuitry's main regulators. Four additional IC regulators supply the pairs of 1.2- and 3.3-V rails required for the two-channel ES9018 DAC.

Local supply bypassing is generous, to say the least. I counted 30 local bypass capacitors on the two-channel PC board: 220 µF/35 V, all in parallel with Wima film capacitors.

The headphone amplifier, also located on the two-channel board, is powered by its own pair of TI fixed, low-dropout, three-terminal regulators, an LM2940CT for the 15-V rail and an LM2990CT pair for the –15-V rail.

The multichannel board's analog circuitry is fed from the LM-317/337 regulators on the two-channel board. The LM4562 op-amps used for the multichannel outputs are bypassed with 24 electrolytic capacitors in parallel with WIMA film types.

The ES9018 DAC chip on the multichannel board is powered by its own set of 1.2- and 3.3-V regulators. The BDP-105's video circuitry and switching power supply are fully shielded to ensure that radiated noise does not degrade the analog circuitry's performance.

DIGITAL INPUTS

The ES9018 DAC accepts an S/PDIF input directly, so it does not require a separate input receiver. The S/PDIF coaxial input is coupled with a pulse transformer, which is the preferred interfacing method. The asynchronous USB input is built around an XMOS-programmable microcontroller, which appears to be either the same XS1-series, 128-pin chip used by PS Audio in its NuWave DAC (reviewed in *audio Xpress*, August 2013) or a similar device. Asynchronous operation ensures that the jittery clocks found in most computers won't compromise the USB input's performance.

The BDP-105 is larger and heavier than the BDP-95—the new player is 4.8" tall (vs. 4") and weighs 17.3 lb (vs. 16 lb). The player is beautifully constructed on a steel chassis with an aluminum faceplate and a center-mounted mechanism for excellent mechanical stability. Chinese-manufactured products are still highly variable in quality, and many are downright shoddy. But, OPPO products clearly represent some of the highest-quality manufacturing that country has to offer.

Some users apparently complained about noise from the BDP-95's cooling fan. (The fan in mine is especially quiet and I never noticed any noise, at least not in my normal listening position.) OPPO has dispensed with the fan in the BDP-105. The combination of a larger chassis and ventilation slots in the cover ensure cool and quiet operation without a fan.

The BDP-105 is supplied with a robust 14 AWG, International Electrotechnical Commision (IEC)-style power cord. But, a player this refined can benefit from a high-quality shielded cord (e.g., the cost-effective Pangea AC-14SE or the no-holds-barred D.H. Labs Red Wave).

PERFORMANCE

The BDP-105 is based on the same video architecture as the BDP-95 and BDP-93. The BDP-105 offers the same state-of-the-art performance as its predecessors (along with the added features described earlier). For details, I refer readers to those reviews.

I used the BDP-105 player's dedicated two-channel unbalanced outputs, feeding the main preamplifier in my system to perform sonic evaluations on the BDP-105. In my BDP-95 review, I provided a thorough overview of that player's sonic virtues, including excellent soundstage reproduction, low-frequency weight, and dynamics. I also noted that "the BDP-95 doesn't force you to choose between the last ounce of detail and the most natural-sounding treble. It delivers the best of both worlds: exceptionally articulate, detailed sound combined with a smooth, silky treble region, and overall tonal neutrality that just makes you want to keep listening."

I concluded that "as an audiophile who is surrounded by live, unamplified music on a daily basis, the OPPO continually impressed me by just how 'right' it sounds." I'm not going to repeat all the details here. I suggest reading the BDP-95 review for the full story.

When OPPO announced the BDP-105, the new features alone would have been enough to justify the modest increase in price. I would have been content if the audio performance simply remained on par with the BDP-95. I'm happy to report that not only has OPPO maintained the virtues of the BDP-95, but it's actually improved on that player's performance in a few areas. The most striking enhancement is in the soundstage reproduction, which is wider, deeper, and more precise than before. The new player also improves on the low-frequency weight and impact. The changes in power supply regulation and bypassing have undoubtedly helped in these areas.

Like its predecessor, the BDP-105's tonal balance is neutral. Although the treble region is airy and detailed, it is also smooth and musically satisfying for the long term and doesn't exaggerate the deficiencies of problematic recordings. The over-miked, sizzly strings in William Steinberg's Deutsche Grammophon (DG) recording of Gustav Holst's *The Planets* are less annoying on the BDP-105 than on players and DACs that favor the treble region. Well-recorded strings (e.g., Antal Dorati's Mercury CD of Ottorini Respighi's *The Birds)* are palpably realistic and a pleasure on the ears. Low strings maintain that rich, gutsy quality that I praised in the BDP-95, taking on a bit more weight than before.

The BDP-105 is an extremely revealing player. I am struck by how different digital re-masterings of the same recordings are easily characterized.

A case in point is Georg Solti's legendary Decca recording of Wagner's *Der Ring des Nibelungen*. Two audiophile editions of this recording have been released in recent years—a premium-priced super audio CD (SACD) edition re-mastered in Japan in 2009 by Esoteric and the Deluxe Limited Edition released by Decca in the fall of 2012, which, in addition to 17 CDs and a DVD documentary, also includes a single Blu-ray audio disc containing the entire Ring cycle in 48-kHz/24-bit uncompressed pulse-code modulation (PCM).

Both editions are based on the late James Lock's 1997 re-mastering, where he up-converted Decca's original 1984 digital transfer to 96 kHz/24-bit. (In 1997, Lock told me the original analog tapes had deteriorated to the point where they were no longer useable, a sad situation confirmed in the notes to the Deluxe Edition.)

The SACD edition is clearly superior to the Blu-ray version, having greater low-frequency warmth and heft and a better sense of the acoustical properties of the recording venue. The SACD also has less of the high-frequency glare inherent in the original 1984 transfer. The BDP-105 easily resolves these differences, and to a greater degree than the BDP-95.

The BDP-105's balanced outputs on the maintain the high-performance level I've noted for the unbalanced outputs. As with the BDP-95, the new player offers stellar performance, driving a balanced-input power amplifier directly, adjusting the internal 32-bit volume control using the remote.

The BDP-95's relatively coarse 5-dB volume increments have been changed to 1 dB on the BDP-105. (A firmware upgrade changes this for the BDP-95, as well.) If you use only digital sources, the BDP-105's input flexibility may eliminate the need for a preamplifier.

Throughout my BDP-105 evaluation, I can honestly say the ELNA Silmic II coupling capacitors never drew attention to themselves. In fact, I had done a considerable amount of listening—all entirely positive—before I became aware that the outputs were AC coupled. Perhaps if I had two BDP-105s for A-B comparison—one with the capacitors and one without—the difference would be audible. As it stands, I'm not able to point to any sonic signature created by these capacitors.

In my Parasound Zdac review, I noted "the headphone amplifier is dynamic and punchy, and had no difficulty driving my AKG K701 headphones, which have a 62-Ω impedance rating. As implemented in the Zdac, the TI headphone amplifier chip is clean, detailed, and spacious."

The OPPO BDP-105 takes the TI TPA6120's virtues to the next level. I found the OPPO's performance with headphones to be slightly better than the Parasound in inner detail and soundstage size.

Like the BDP-95, the new player may make an outboard DAC unnecessary, unless you're prepared to spend a lot of money. None of the affordable DACs I've reviewed in recent years beat the BDP-105 as a stand-alone player, though one came within striking distance.

The PS Audio NuWave DAC offers a slight improvement in soundstage size and localization. The low-order THD added by its discrete output circuitry provides a harmonic richness that is musically extremely engaging. In other areas, (e.g., tonal balance, high-frequency smoothness, and detail), I prefer the BDP-105.

I've only had one DAC in my system that clearly beat the BDP-105, and that's the Bricasti M-1. Last winter, I had an M-1 for one weekend and experienced the finest PCM digital playback I've ever heard. But, the M-1 sells for $8,600, so there's no way it was going to become a permanent addition to my listening room.

Realistically, I think you'll have to spend at least $1,500 or more on an outboard DAC to significantly improve on the BDP-105's performance as a stand-alone player with PCM sources. The BDP-105 will output a DSD data stream via its HDMI outputs when playing SACDs. Unfortunately, I have yet to see an outboard DAC with HDMI inputs that can accept DSD data.

FINAL IMPRESSION

With the BDP-105, OPPO has raised the bar once again. The ability to use the BDP-105 as a DAC with a variety of digital sources makes it more than just a universal player. It's really a complete digital music server.

The refined audio performance alone would make the BDP-105 an easy choice for audiophiles. Add state-of-the-art video capabilities and a list of features that leaves little, if anything, to be desired, and you have another high-quality player from OPPO Digital. The BDP-105 is now the reference digital player in my listening room, and I just purchased one for my studio at work. Need I say more? *aX*

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Zdac DAC
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Audio Innovator Amar Bose Remembered

Psychoacoustics inspired his inventions and MIT teaching career

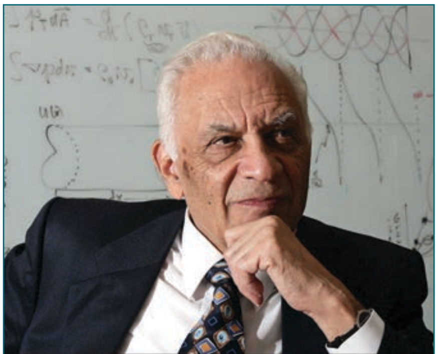


Photo 1: Amar Bose was the visionary, engineer, inventor, and billionaire entrepreneur whose namesake company, Bose Corp., became synonymous with high-quality audio systems and speakers for home users, auditoriums, and automobiles. (Photo courtesy of Bose Corp.)

Dr. Amar G. Bose, founder of Bose Corp., has undoubtedly exerted a large impact on the audio industry. However, many industry people and consumers did not always view this impact as positive. In high-end circles, his product philosophy was often bashed with the same vehemence the MP3 now receives. Bose died July 12, 2013, at the age of 83 (see **Photo 1**).

When he was 27, Bose's interest in sound reproduction was the basis for his study in psychoacoustics at the prestigious Massachusetts Institute of Technology (MIT). Later, he started teaching in the same field at MIT, receiving an award for best professor in 1963 (see **Photo 2**).

In 1964, he started Bose Corp. with Sherwin Greenblatt. In 1968, they bought the rights to the (in)famous 901 speaker system, which was based on Bose's earlier research. The 901 consisted of nine separate full-range drivers, eight of which were rear facing to maximize the reflected sound in the total soundfield. This speaker caused quite a stir, immediately dividing the cognoscenti. There were those who thought this was a revolutionary development, bringing the realistic reproduction of music one step further. Others thought Bose had sold out realistic reproduction for sound effects.

After hearing the 901 at an audio show in 1970 or thereabouts, I came away with the impression that, predictably, this speaker's sound-reproduction qualities depended more heavily on the venue acoustics and on the particular recording. Some situations benefit from increased reflected sound. But for some recordings and rooms, it results in a mash of sound.

Be that as it may, it established Bose's reputation in one fell swoop. Incidentally, the speaker (and the accompanying analog equalizer) can still be purchased today as the Bose 901 MK VI model. That is an extremely long product life, even by 1970s standards.

Psychoacoustics continues to drive the Bose product philosophy. Bose never tried to recreate the soundfield from a particular performance. He was more interested that you as the listener have the same perception you would if you listened to the performance. And as we all know, perception can be manipulated.

My personal favorite device, which plays in my home at some point almost every day, is a vintage sound dock. If you open it up, you'll find a small Class D power amp and a speaker feeding into a small folded horn or pipe. Now, according to established theory, such a small system is unable to reproduce any appreciable low-frequency tones. Apparently the unit uses processing to synthesize the low-frequency tones that it cannot reproduce. Psychoacoustics and perception do the rest. You actually "hear" much more low-frequency sound than you thought would be possible from such a small unit.

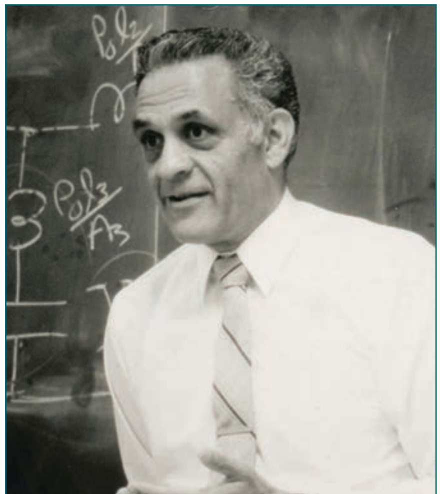


Photo 2: Dr. Amar Bose taught a psychoacoustics class at MIT. (Photo courtesy of Bose Corp.)

Certainly, experienced listeners of quality audio reproduction systems will quickly realize that "something isn't quite right" in the way these psychoacoustics products sound. But in the end, it's all about the music. There is no doubt in my mind that, by external design and/or clever engineering, Bose products have brought musical enjoyment to many people who otherwise would have settled for much less.

Amar Bose was not only a scientist and a businessman who thought outside the box. He was also a gifted teacher.

As one former student said: "Amar Bose taught me how to think." What better compliment is there to give to a man? *aX*

Early Curiosity Leads to a Lifelong Engineering Pursuit

Gregory Charvat—an engineer, professor, entrepreneur, and author—encourages people to explore the opportunities that come their way.

SHANNON BECKER: Your interest in amateur radio equipment began when you were young and continued through your high school years. Can you describe some of your first projects?

GREGORY CHARVAT: I grew up in the metro Detroit, MI, area, and I was very interested in electronics for as long as I can remember, almost naturally. As a young child, I wanted to understand where all the people and things on television came from. One day we took apart an old television set so I could see how it worked.

My mother studied electrical engineering at Lawrence Technological University, which is an engineering school outside Detroit. She occasionally brought me to her classes and labs. Later while she was working at Ford Electronics, she sometimes would take me to work on the weekends, which was super exciting because I was able to make circuit boards, use oscilloscopes, and so forth (this was maybe from age 10–13).

I learned how to use transistors at age 10. I became interested in CB radios. At about the same time, I helped my friends dust off CB radios from their parents' attics and we built a very nice CB radio network. When I was about 13, I transitioned into amateur radio because my mother wanted me to try a more sophisticated hobby—one where you build your own equipment and communicate across great distances.

My first electronics projects that I can remember involved building a large switch panel that ran off an old model train transformer to power a car radio, a fan, a light bulb, and other things. Next, I outfitted my bike with buzzers, lights, and a generator to power them with a rechargeable battery. Shortly thereafter, amateur radio became my main interest. I built a 2-m transceiver kit, a 2-m power amplifier kit, modified old police radio power amplifiers to work on 2 m, and many other things. In high school I became involved in a radio telescope project. My role was to upgrade the receiver to a more advanced scanning spectrum analyzer with analog signal-tracking circuits that I developed. Some of these circuits were later used in a friend's dissertation at Michigan State University.

Gregory Charvat created an intricate work station at his Westbrook, CT, home to design, test, and build projects.

When I was 16, I interned at Aeroflex Lintek, a defense contractor, helping build radars used to measure the radar cross section of stealth aircraft. From there I became interested in radar technology and I was amazed that you could make images with microwaves. I recall repairing the rotator at Mission Research in Dayton, OH, testing low-noise amplifiers (LNAs) on an HP8510 network analyzer, building many boards, military-specification cables, chassis, mounting power supplies, and finding parts in catalogs for designs.

SHANNON: What is your current occupation? Are you still involved with Butterfly Network?

GREGORY: I am the co-founder of Butterfly Network, which is a privately held company that brings together world-class talent in computer science, physics, and electrical engineering to create an entirely new approach to diagnostic imaging and treatment. I lead the hardware team, which is our company's largest group. We are responsible for the development of the hardware for the company's first product.

SHANNON: Can you describe some projects you've done?

Gregory's go-to vacuum-tube transceiver is a 20-m single-sideband (SSB) of his own design.

GREGORY: I built the first three prototypes, set up our lab and machine shop, recruited more than half the company, negotiated a number of contracts, and now I am learning about application-specific integrated circuit (ASIC) engineering. We are looking for an analog ASIC engineer and a chip packaging engineer, for anyone who is interested in working with me. I can be reached at charvatg@gmail.com. I will reply to all e-mails.

SHANNON: Are you still instructing at the Massachusetts Institute of Technology (MIT)? Can you describe the courses you teach?

GREGORY: My Build a Small Sensor Radar course continues to this day. In June, it was offered as an MIT Professional Education course. Numerous universities have adopted this course as either a capstone engineering project or full-semester course. Private institutions and government labs have also adopted the course for internal training. The coffee-can radar project has even made an appearance on Mongolian National Television as part of an MIT EdX program a friend of mine ran last summer. You can now buy the coffee-can radar project as a complete kit from Quonset Microwave (www.quonsetmicrowave.com).

Additionally, two other Build a Radar courses were developed from the original course including Build a Phased-Array Radar and Build a Search and Track Radar, but I am no longer teaching these courses. My relationship with MIT has evolved into me being an advisor for the Camera Culture Group at the MIT Media Lab where we experiment with two interesting technology threads.

SHANNON: You have a main website (www.glcharvat.com), a blog, Mr. Vacuum Tube (http://mrvacuumtube.blogspot.com); several YouTube videos (www.youtube.com/user/charvatg?feature=mhum); and an active Twitter account. What types of projects do you typically share?

GREGORY: I like to read other people's Twitter posts and blogs and I watch YouTube videos. I appreciate the time and effort other enthusiasts put into sharing their work. To this end, I like to share what I am doing and maybe it can help someone someday. For this reason I try to share everything I can, from symposiums that I am involved in to topics in my book, to papers I am publishing to hobbyist projects (e.g., vacuum-tube audio, antique radio restorations, clock and watch restorations, and sailing adventures).

As an undergraduate, I learned from a math professor friend of mine that you must publish everything you create, otherwise the knowledge will be lost.

SHANNON: You are a Senior Member of the Institute of Electrical and Electronics Engineers (IEEE). What does your involvement entail?

GREGORY: I'm a huge fan of the IEEE. The IEEE organization maintains high academic standards for papers in its journals and symposiums and I really appreciate that. IEEE members are the archivists and storytellers of electrical engineering history, past and present. To help out, I was chairman of the IEEE Antennas and Propagation Society Boston Section for two years. Our meetings went from 12 attendees each month to 40 by my tenure's end. This was achieved by the use of social media and the lineup of fascinating speakers. I then became a Member-at-Large for the IEEE Boston Section's Executive Committee in 2012. Additionally, I was on the steering committee for the 2010 International Symposium on Phased-Array Systems & Technology. I am currently on the steering committee for the 2013 symposium this fall in Waltham, MA.

The "Frankenstein" home- theater system used 23 vacuum tubes, powered by 535 VDC of plate voltage over 20 A of filament current, protected by a 10-A main circuit breaker and more than a dozen 3AG fuses.

SHANNON: You have authored many articles, including a two-part *audioXpress* series, "Vacuum Tube Home Theater System" (May–June, 2012), about building a "monster" system with 23 vacuum tubes. Tell us a little about the system. Have you made any upgrades or adjustments?

GREGORY: This system was originally developed in the summer of 2001 while I was a junior undergraduate. It was deployed in my dorm room. The feedback compensation networks were upgraded in 2003. The power supply was upgraded in 2005 and I moved to a direct grid-bias configuration to get more power out of the amplifiers.

In 2009, I again upgraded the feedback compensation networks. This system is in our living room and it is used almost every day. It is currently in a maintenance posture, where I occasionally have to replace one of the original tubes from 2001. Recently I had to replace a high-voltage rectifier diode. I'm due to replace the pentodes on the subwoofer amplifier within the next few weeks.

SHANNON: Your forthcoming book, *Small and Short Range Radar Systems,* is slated for publication in February 2014. What can readers expect to learn?

GREGORY: This will be the first book on small and short-range radar devices. My goal for the book is to show readers how they work, how to build something practical, and do it quickly.

I want to show readers the high-level basics and how to do a back-of-envelope estimate to determine if a system is worth developing. I also wanted to provide working engineering examples of each type of short-range radar, including complete schematics, diagrams, bill of materials (BOM), a demonstration video, data for the reader to process from that video, and usable MATLAB scripts. I also discuss the practical aspects of design so these examples can be scaled to other applications or expanded.

As engineers, we are applied scientists. We use our understanding of physics to build something tangible to help humanity. This book facilitates that process for short-range radar sensors.

SHANNON: Tell us about the amateur radio station you designed.

GREGORY: After hearing stories about a friend of mine at the MIT Haystack

This homemade amateur radio station evolved from an idea to construct a shortwave receiver out of one type of transistor.

Observatory, who as a teenager built his own radio out of one type of transistor his father was able to "borrow" from work, I decided to redo this exercise myself. I developed a shortwave receiver out of one type of transistor. It was a fascinating experience.

From there, I decided to develop my own single-sideband (SSB) transceiver design from scratch. I built a 20-m SSB radio with 40-W peak-envelope power (PEP) that is extremely sensitive. It was an amazing experience to contact Western Europe with a radio that I developed myself!

Shortly thereafter, the American Radio Relay League (ARRL) issued the Homebrew Challenge III, to build a 10-m and 6-m SSB/CW transceiver for less than $200. I took this challenge and submitted a working radio one day before the contest ended. Apparently mine was one of only two submissions. The other was late, but I blew the budget at $450! (It's really about the experience, not the cost.) I was given honorable mention and an article in *QST* magazine for my efforts. More importantly, I had the pleasure of getting to know some of the folks at ARRL.

Most recently, I developed a way of tying a SSB transmitter to an R-390A, which is the tube era's best communications receiver and highly sought after by collectors. An article on this modification will run soon in *QST*'s "Hints and Kinks" column.

In summary, my homemade amateur radio station consists of a 20-m SSB radio, a 10-m and 6-m SSB/CW radio, and an R-390A that is tied into the 20-m radio for when I want to run in "boat anchor" mode.

SHANNON: If you had a full year and an ample budget to work on any design project you wanted, what would you build?

GREGORY: If I had a free weekend in the summer I'd sail to Long Island, NY. If I had a free weekend in the fall, winter, or early spring I'd restore a vintage mechanical wristwatch. If I had a free week I would restore the vintage all-tube Telefunken stereo receiver in my office. Or, I'd combine a 1970s solid-state amateur radio transmitter with my R-390A. If I had one to four months, I would bring to close the projects I am helping with at MIT's Media Lab.

After that, I would look for something long term. I would probably brainstorm with some friends about a new idea for a fun research project or a startup. Generally speaking, I would like to have one long-term project moving steadily along while also closing numerous shorter-term projects along the way. *aX*

By Richard Honeycutt (United States)

Dynamics Processing with Tubes

Introduction to audio processing

In this article, I am going to examine some common audio processing tools that were developed long before solid-state electronics was a reality. But to do so, I need to introduce the ideas and terminology associated with these tools. Next month, I'll look more specifically at the circuitry to implement these tools using hollow-state components.

WHAT IS "DYNAMICS PROCESSING?"

The variation in sound level, from very quiet to very loud, or from pianissimo to fortissimo in music, is called the dynamic range of the sound. In a fairly quiet living room, the background noise may be about 30 dBA. (This means 30-dB sound pressure level, "A-weighted.")

The A weighting is a correction of actual sound pressure level (SPL) according to frequency, to account for the fact that our hearing abilities are less sensitive at low and high frequencies than in the midrange. Sound levels expressed in decibel amperes correlate more closely to our perception of loudness than do uncorrected sound levels, expressed in dB_{SPL}. A good sound-reproducing system—whether solid- or hollow-state—will produce less background noise in the listening room than will the ambient noise sources (e.g., HVAC, traffic, people, birds, crickets, etc.).

The maximum sound level to which we should ever expose our ears, even for brief periods, is 120 dBA. The dynamic range of pretty good modern home-listening conditions is about 90 dB (i.e., 120 to 30 dBA). An excellent concert hall may have an additional 15 dB, because of lower background noise. (Even a full audience can be very quiet, as people often hold their breath during pianissimo passages.) As another example, 78-rpm lacquer records had a dynamic range of approximately 40 dB when

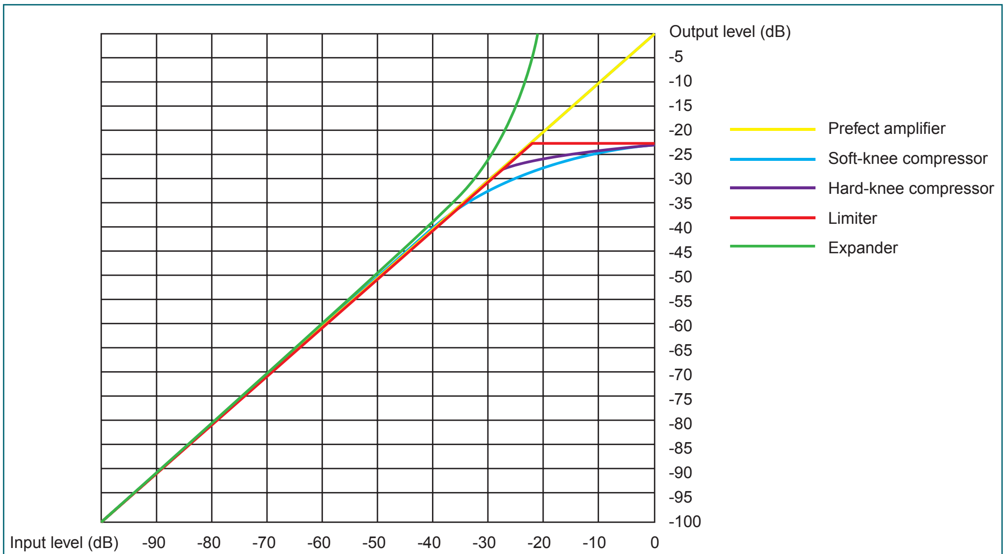


Figure 1: Dynamics processors apply various level-dependent gain adjustments. A downward compressor is shown. The expander has a complementary curve.

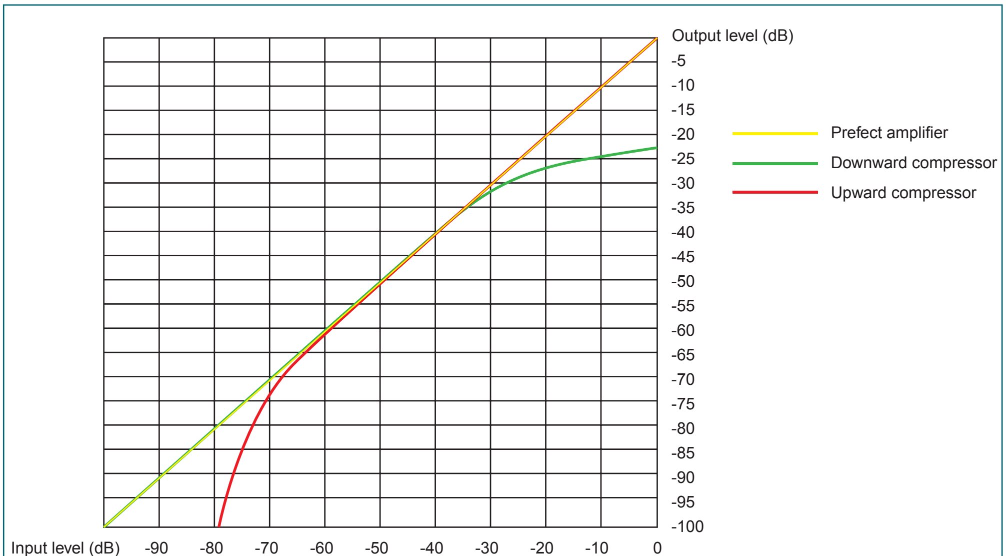


Figure 2: Downward and upward compressors affect the dynamic range in different ways.

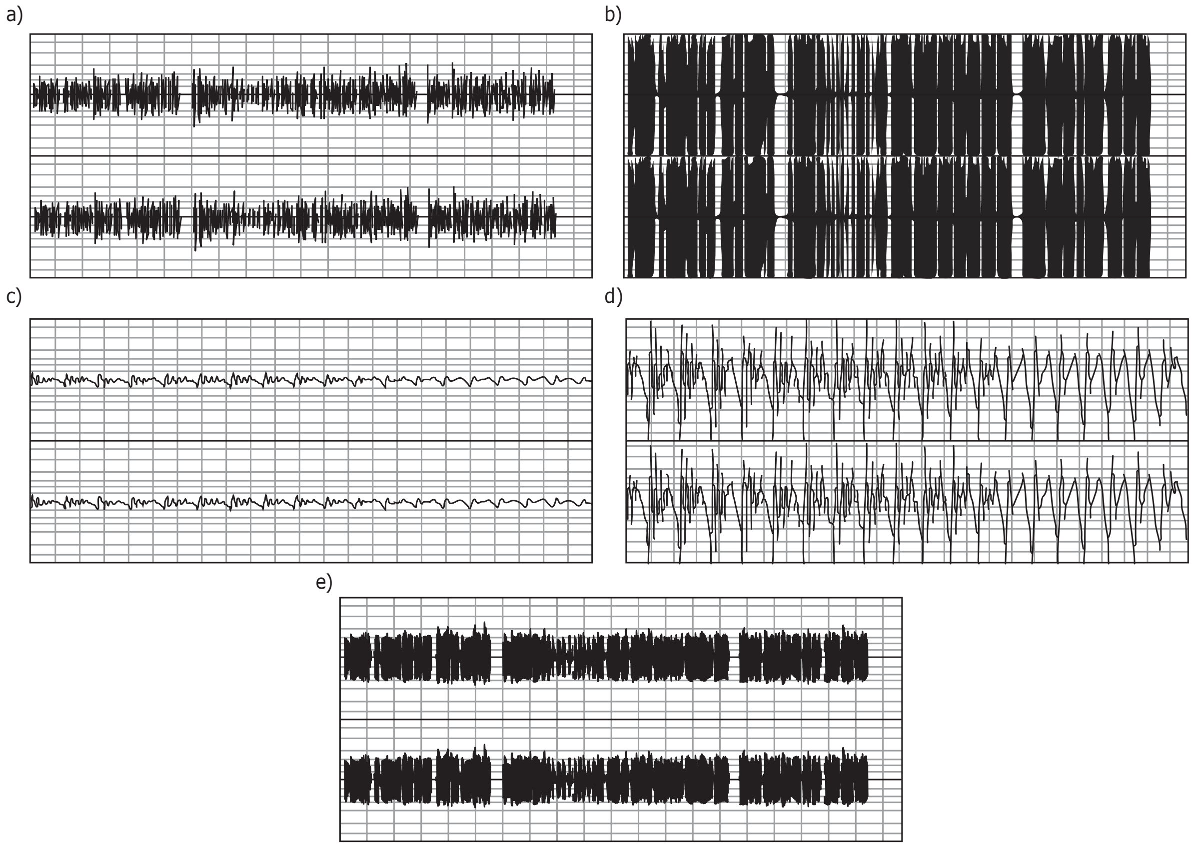


Figure 3: The compression and limiting effects shown include: the original speech waveform (a), amplified and limited (b), original waveform zoomed in (c), amplified/compressed waveform zoomed in (d), and speech waveform with "vocal compression" (e).

new, but this quickly degenerated to approximately 30 dB because of the increase of surface noise with wear.

An LP record can have a dynamic range of approximately 65 dB. Reel-to-reel tape, without noise reduction applied, has a dynamic range of approximately 60 dB, and cassette tapes are not quite as good, at 50 to 56 dB. Theoretically, a CD has a maximum dynamic range of 96 dBA, but in practice this depends on several circuitry variables.

Several tricks can be applied to a signal to better take advantage of the available dynamic range. One is to amplify the weaker signals more than the stronger ones. This process is called "compression." Compression does not increase dynamic range, in fact it compresses dynamic range. But, it does enable the listener to hear sounds that would otherwise be lost in the noise. In some applications, extreme compression, called "leveling" or "automatic volume control" (AVC), is used to reduce dynamic range to almost zero. Examples are devices that reporters use to record speech, telephone recorders, and a distressing percentage of broadcast radio stations.

Another approach is to amplify the entire signal enough that the peaks are regularly clipped slightly. Peak clipping also does not increase dynamic range, but it does bring weaker signals up out of the noise. The obvious disadvantage of peak clipping is that it produces distortion. However, producers of training recordings, such as the military, used peak clipping for many years to overcome the background noise on lacquer records and older film soundtracks. They found that if the peaks were only clipped slightly, and the high frequencies were sharply rolled off above 4 kHz, the distortion was tolerable and the speech intelligibility was actually slightly improved.

A cross between speech clipping and compression is peak limiting. A peak limiter uses almost the same circuitry as a compressor, but the amplification does not change with the changes in signal level until the signal reaches the maximum allowable level. At that point, no further output level increase occurs, but the waveform peaks are reshaped in such a way as to introduce less distortion than a clipper would add.

To overcome the dynamic range reduction caused by compression, dynamic expansion can be applied. For example, a music signal with a 100-dB dynamic range can be fed to a compressor that decreases its dynamic range to 65 dB for recording onto a vinyl record. Then an expander can be inserted between the phono preamplifier and the power amplifier to boost the dynamic range back up. To work well, the expander must have characteristics that precisely mirror those of the compressor. Modified versions of this scheme are the basis for Dolby and DBX noise reduction, introduced in the late 20th century to improve the dynamic range of cassette tapes.

Figure 1 shows the effects of peak

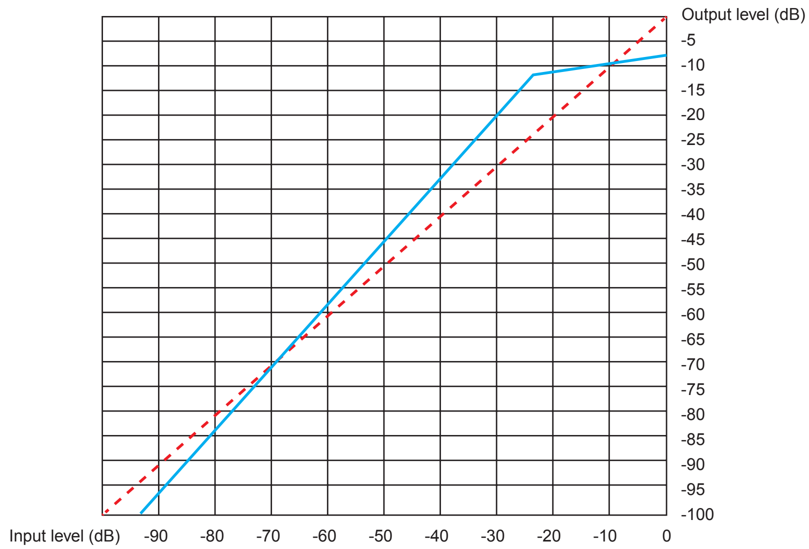


Figure 4: The curve for this "vocal compressor" shows both downward and upward compression.

limiting, compression, and expansion. The vertical axis represents the output voltage and the horizontal axis represents the input voltage. A perfect amplifier would have the same gain (V_{OUT}/V_{IN}) for all levels of input voltage, as represented by the straight diagonal line. A peak limiter would have a similar characteristic, except that at the point corresponding to maximum output, the line's slope would change from diagonal to horizontal.

A compressor's characteristic would be sort of a rounded-off version of the limiter's characteristic. An expander would have a mirror-image version of the compressor's characteristic. A system employing a compressor and a complementary expander is called a "compandor."

OPERATING PARAMETERS OF DYNAMICS PROCESSORS

A speech limiter needs to react quickly to a sudden increase in output voltage to prevent clipping. The time required for an instantaneous change in input voltage to cause a specified change in gain (often 10 dB) is called the "attack time." Limiters must have short attack times, typically 20 ms or less. Compressors and expanders may not need to react as quickly, so they may have longer attack times.

Short attack times produce more audible artifacts than do long attack times. A compromise can be achieved by using the "look-ahead" feature available in some processors. This feature creates a delayed version of the signal. Any necessary gain changes can be applied to that delayed version using a slow attack time, after the need for the change is identified by sensing the undelayed signal. The result is the same as if you had been able to look into the future and see what the signal levels would be in advance.

The time for the gain of the compressor or expander to return to its nominal value is called the "release time." Since the background noise is affected by dynamics processing, the noise level increases as the gain increases. If the release time were equal to the attack time, a "breathing" effect would be heard in the audio. To avoid "breathing," the release time is usually set for longer than attack time: 50 to 500 ms is a common range.

The input voltage at which the gain of the dynamics processor begins to change from the nominal value is called the "threshold." In audio equipment, threshold is often specified in decibels referred to 1 V (dBV).

The "ratio" is the number of decibels by which the input voltage must change to produce an output voltage change of 1 dB. Thus, a compression ratio of 4:1 would mean that if the input voltage increased from −34 to −30 dBV (an increase of 4 dBV), the output voltage would increase by 1 dB.

Compressors using ratios higher than 10:1 are often considered to be limiters. Extreme ratios, 20:1 up to ∞:1, represent what is usually called "brick-wall limiting."

So far, we have only described one type of compression: Gain reduction when the input voltage exceeds the threshold. This is called "downward compression," since the gain is adjusted downward in response to an increase in input level. There is also "upward compression," in which the gain is increased when the input voltage drops below the threshold. **Figure 2** shows the difference in characteristic curves for downward and upward compressors. There are also downward and upward expansions—downward expansion mirrors upward compression and vice versa.

A downward compressor will decrease the signal's overall average level by decreasing the gain for the wave's loudest parts. Some processors compensate for this effect by applying "makeup gain" to keep the average level unchanged. Of course, applying makeup gain requires more compression of the loud parts.

The manner in which the gain changes at the threshold is classified as either a "hard knee" (sudden change in gain) or a "soft knee" (gradual change in gain). Hard-knee processors produce more noticeable audio artifacts than soft-knee processors.

Some dynamics processors sense the input signal's peak level; others, the average or RMS level. Peak-sensing processors more reliably prevent clipping, but average or RMS sensing is less likely to create audible artifacts.

In a stereo or discrete surround-sound signal, any dynamics processing that is applied to one channel usually should be applied equally to the other channel, to avoid changes in instrument balance and/or directional cues. This is achieved by a "linking" control that derives the control signal from all channel inputs, then applies it to all channels.

Figure 3 shows the effect of dynamics processing on a speech signal. Each part of the figure shows two

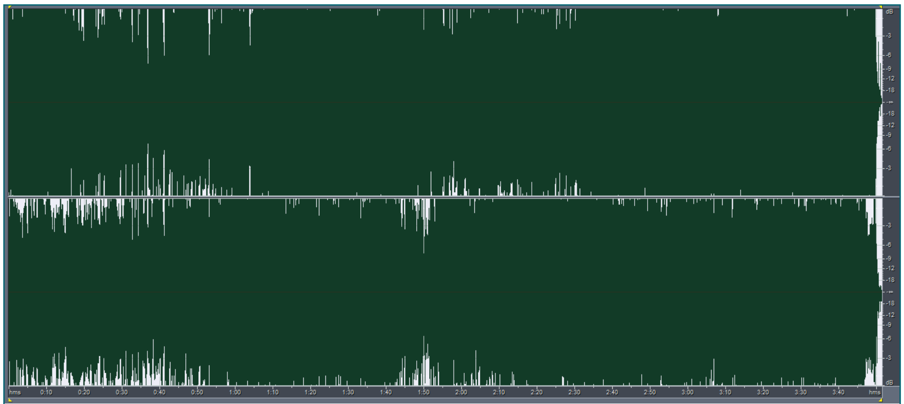


Figure 5: An overcompressed music wave looks like a ribbon.

waveforms: the left channel waveform is on the top and the right channel is on the bottom. **Figure 3a** shows the original unprocessed signal.

Figure 3b illustrates the signal's effect after 25 dB of amplification and peak limiting. Notice that the peaks are flattened above the limiting threshold. **Figure 3c** and **Figure 3d** (zoomed-in versions of **Figure 3b**) show how the peak limiting affects the peak's shape. **Figure 3c** is the unprocessed peak. **Figure 3d** is the peak after limiting.

This hard limiting was applied digitally, making it possible to better preserve the shape of the peaks than an analog compressor could have done. However, notice that the variation in amplitudes of adjacent peaks was removed by the limiting.

Figure 3e shows the effect of applying a common vocal compression curve having attack and release times of 1 and 250 ms, respectively, a threshold of –24 dB, a downward compression ratio of 5.5:1 for inputs above the threshold, and an upward compression ratio of 1.3:1 for smaller inputs. Notice that the average level is increased more than the level of the loud parts of the signal. **Figure 4** shows the "vocal compression curve" shown from **Figure 3e**.

Figure 5 shows a music signal compressed to what some engineers call a "ribbon," using a 20:1 ratio. This degree of compression is used by almost all commercial radio broadcasters except for classical stations, and by many "pop" CD producers. It has been likened to turning the volume up to "10" and then mashing the signal with a sledge hammer. If you would like to hear what this overcompression sounds like, visit www.youtube.com/watch?v=3Gmex_4hreQ.

Next month, I'll discuss some of the history of dynamics processing, detail how it's done using hollow-state components, and then examine some classic circuits. *aX*

MEMBER PROFILE

Member Name: Dennis L. Green

Location: Farmington Hills, MI

Education: BSEE

Occupation: Retired

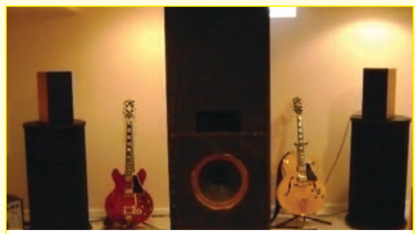
Member Status: Dennis has been a subscriber since the first issues of *Audio Amateur* and *Speaker Builder.* He switched to *audioXpress* after their demise, mostly for the component ads and the solid-state and speaker projects.

Affiliations: Dennis is a member of the Institute of Electrical and Electronics Engineers (IEEE), the Audio Engineering Society (AES), and Southeastern Michigan Woofer and Tweeter Marching Society (SMWTMS).

Audio Interests: His primary audio interests focus on homebrew audio and field recording.

Most Recent Purchase: Dennis recently added an Apex Jr. subwoofer amp to his audio lineup.

Current Audio Projects: He is currently working on an optical sensor-based servo control upgrade for the Kilmanas Rabco tonearm featured in an *Audio Amateur* article in March 1976.



Dennis recently moved and has a temporary setup for his vast sound system.

Dream System: Dennis said his dream system includes a home theater using Quad ESLs in a custom room. However, Dennis said he is "not at all unhappy" with his present system." His current system includes: Swan IV satellites built from an April 1988 *Speaker Builder* article, four Peerless 8" midrange woofers (each side in concentric Sonotubes with sand damping in the space), and a Cerwin-Vega 189E driver (salvaged from the movie *Earthquake* Sensurround system) in an enclosure that formerly contained a Jolly Giant system. The Jolly Giant system used a Hartly 24" tweeter and was also built using a *Speaker Builder* article. (Dennis intended some sarcasm here since that speaker radiated more energy at 20 kHz than at 20 Hz.) His present system measures ±1 dB from 60 to 16 Hz and –10 dB at 12 Hz.

Dennis also has a Swan crossover modified with Linkwitz-Riley stage for a subwoofer (his design) and a Lampton preamp built from an *Audio Amateur* schematic. However, his design added switching and a rack-mount enclosure. He restored Quad 303 and Dyna Stereo 120 power amps and a Leach Low TIM amp (built from plans in *Audio* magazine) that are temporarily driving the subwoofer (a new amp will be installed soon). He said his system is not very presentable since he has moved to a new home and remodeling projects have taken priority.

Dayton Audio's AMT1-4

This air-motion transformer is a front-firing monopole

This month's Test Bench reviews a high-frequency planar driver from Dayton Audio's OEM division. The AMT1-4 air-motion transformer is typical of the device pioneered by Oscar Heil in the 1970s, the Heil Air Motion Transformer. Because Heil died and his patents have expired, the device now goes under its generic name, the air-motion transformer (AMT). Several loudspeaker manufacturers have renamed this type of transducer (e.g., Elac's JET, Precide's Air Velocity Transformer, Adam's Accelerating Ribbon Technology, Emotiva Pro's Air-Motiv, and Martin Logan's Folded Motion Tweeter).

Heil was a German electrical engineer and inventor. He studied physics, chemistry, mathematics, and music at the Georg-August University of Göttingen and was awarded his PhD in 1933 for his work on molecular spectroscopy.

Heil and his wife, Agnesa Arsenjewa-Heil, published a pioneering paper that developed the concept of the velocity-modulated tube. The concept accepts that a beam of electrons could be made to form into "bunches" and generate with reasonable efficiency radio waves of considerably higher frequency and power than were possible with conventional vacuum tubes/thermionic valves. This resulted in the "Heil tube," the first truly practicable microwave generator. These devices were significant milestones in microwave technology development (particularly radar). Velocity-modulated tubes are still in use. Heil is also mentioned as the inventor of an early transistor-like device, based on several patents issued to him in 1934 describing a field-effect transistor (FET) concept, although there is no evidence that this device was ever "prototyped." It's difficult to understand how a physicist of Heil's caliber got involved with audio, but he studied music as well as physics.

Heil received the following six AMT patents:

- US Patent #3,636,278 (January 18, 1972)—Describes the Heil AMT variations, some of which resemble the ELAC JET transducer
- US Patent #3,832,499 (August 2, 1974)—Describes the Heil AMT embodiment used in the ESS AMT series
- US Patent #4,056,697 (November 1, 1977)—Movable diaphragm connector with a flexible hinge diaphragm surround
- US Patent #4,039,044 (August 2, 1977)—A low-frequency AMT. The patent also describes the principle used by Tymphany for its LAT transducer including the embodiment type
- US Patent #4,107,479 (August 15, 1978)—Describes further low-frequency AMT refinements
- US Patent #4,160,883 (July 10, 1979)—An acoustic transducer and the method to make it

Photo 1: Dayton Audio's AMT1-4 features a low-mass German-made Mundorf diaphragm, neodymium magnets, and a cast-aluminum faceplate.

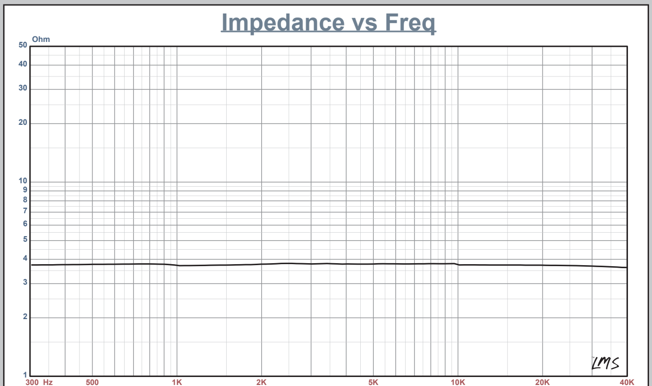


Figure 1: Dayton Audio AMT1-4 free-air impedance plot

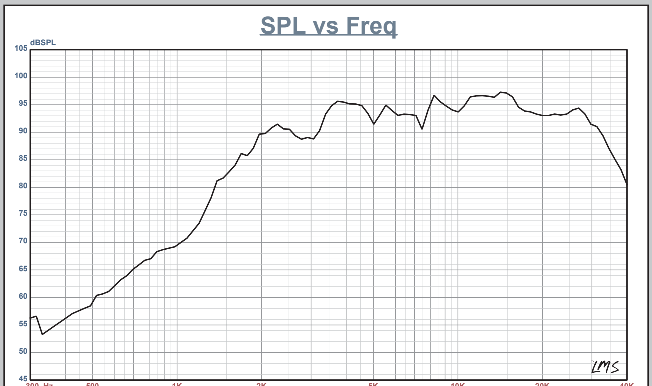


Figure 2: Dayton Audio AMT1-4 on-axis frequency response

THE AMT1-4

Dayton Audio's version of Heil's invention is the AMT1-4 (see **Photo 1**). This AMT is a front-firing monopole (the first ESS Rock Monitor's AMT was a dipole) that incorporates a felt damping pad for the AMT's rear radiation. The diaphragm measures about 30 mm × 44 mm, with a stated Sd of 12.6-cm^2. The AMT1-4's features include a low-mass German-made Mundorf diaphragm, neodymium magnets, and a cast-aluminum face plate.

I used the LMS analyzer to produce the 300-point impedance sweep shown in **Figure 1**. Like a lot of planar devices, the AMT1-4 has no discernible resonance. The company's literature lists the AMT1-4 at 4 Ω. The impedance varies between 3.6 to 3.8 Ω with a 3.79-Ω DCR.

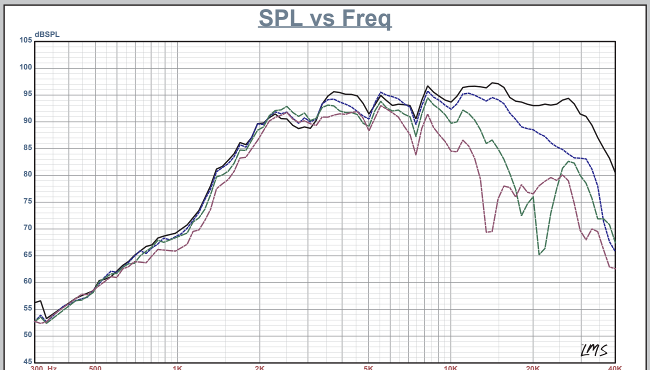


Figure 3: Dayton Audio AMT1-4 horizontal on- and off-axis frequency response (0 = solid black; 15° = dot blue; 30° = dash green; 45° = dash/dot purple)

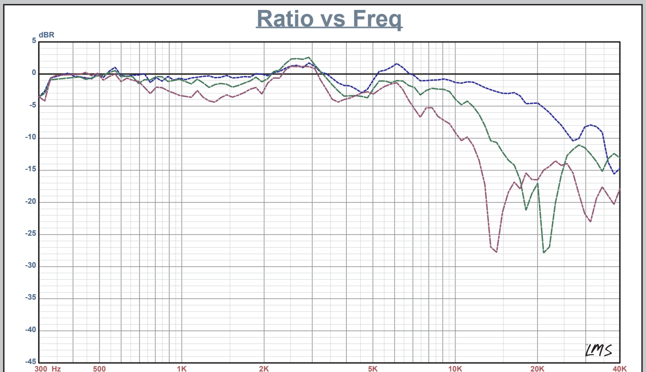


Figure 4: Dayton Audio AMT1-4 normalized horizontal on- and off-axis frequency response (0 = solid black; 15° = dot blue; 30° = dash green; 45° = dash/dot purple)

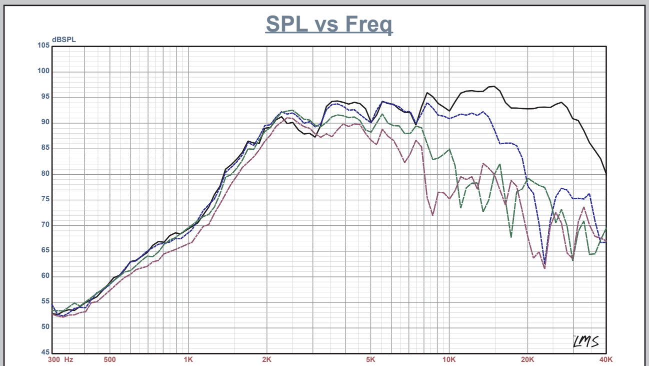


Figure 5: Dayton Audio AMT1-4 vertical on- and off-axis frequency response (0 = solid black; 15° = dot blue; 30° = dash green; 45° = dash/dot purple)

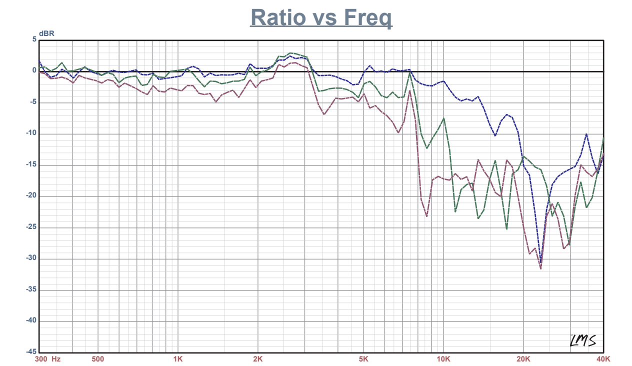


Figure 6: Dayton Audio AMT1-4 normalized vertical on- and off-axis frequency response (0 = solid black; 15° = dot blue; 30° = dash green; 45° = dash/dot purple)

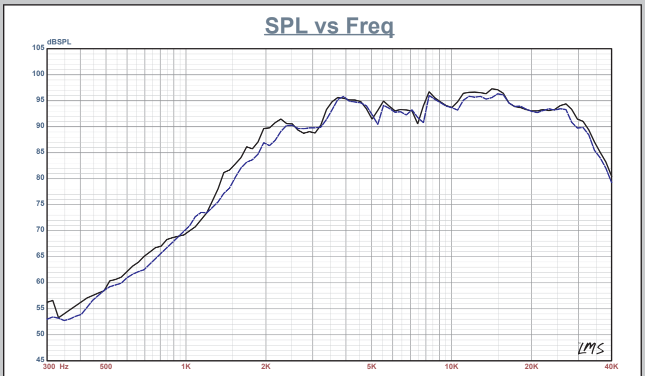


Figure 7: Dayton Audio AMT1-4 two-sample SPL comparison

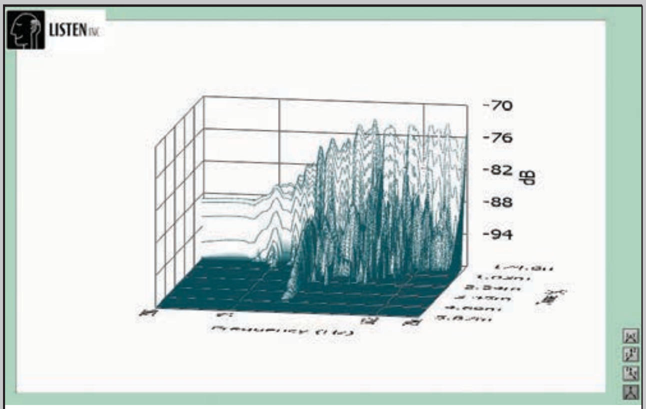


Figure 8: Dayton Audio AMT1-4 SoundCheck CSD waterfall plot

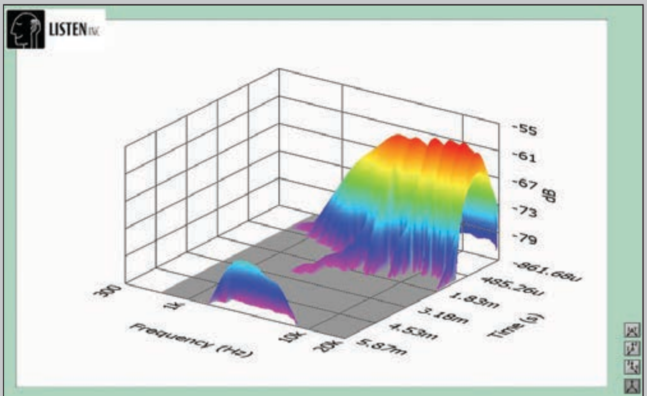


Figure 9: Dayton Audio AMT1-4 SoundCheck STFT plot

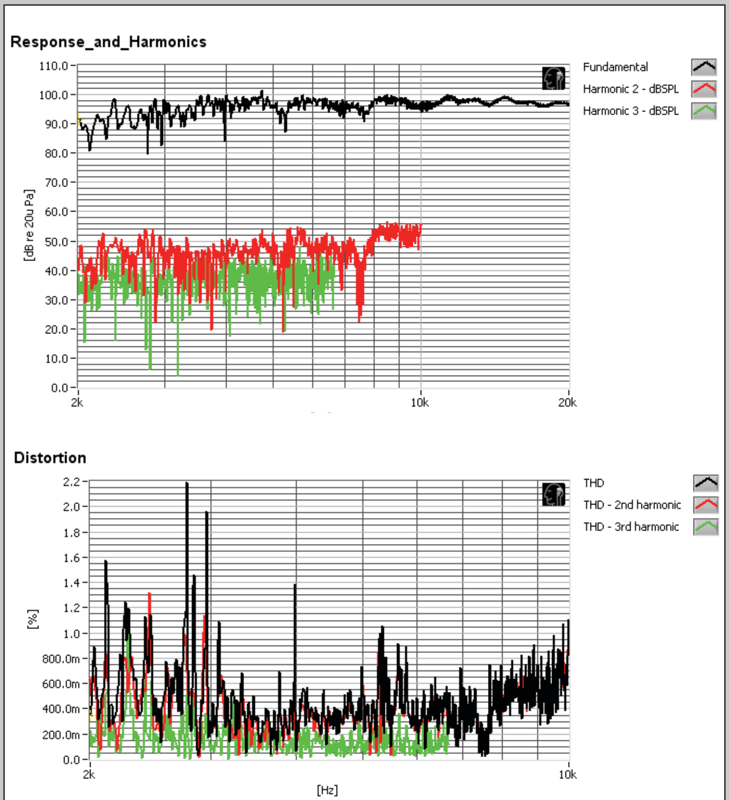


Figure 10: Dayton Audio AMT1-4 SoundCheck distortion plots

Following the impedance testing, I recess mounted the AMT1-4 tweeter in an enclosure that had an 8″ × 10″ baffle area and measured the on- and off-axis frequency response in the horizontal plane with a 100-point gated sine wave sweep at 2.83 V/1 m. **Figure 2** shows the on-axis response to be ±4.2 dB from 3 to 30 kHz. **Figure 3** shows the AMT1-4's on- and off-axis response, with the off-axis curves normalized to the on-axis response (see **Figure 4**). Since this diaphragm has an asymmetrical aspect ratio, I also measured the vertical on- and off-axis response (see **Figure 5**). It is shown normalized in **Figure 6**. **Figure 7** shows the two-sample sound pressure level (SPL) comparison, indicating the two samples were matched within 0.5 to 1.5 dB.

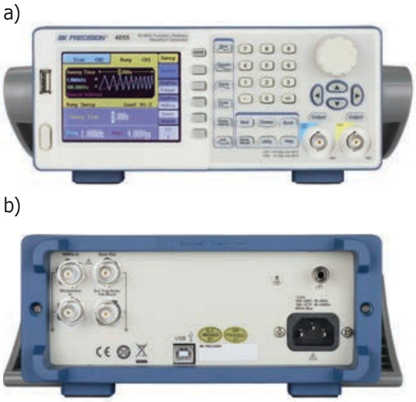
Following the SPL testing procedure, I set up the Listen SoundCheck AmpConnect ISC analyzer and the SC-1 0.25″ microphone (courtesy of Listen) to measure the impulse response with the tweeter recess mounted on the test baffle. Importing this data into the Listen SoundMap software (now included in SoundCheck 12.0) produced the cumulative spectral decay "waterfall" plot shown in **Figure 8**. **Figure 9** shows a short-time Fourier transform (STFT) displayed as a surface plot.

For the final test, I used a pink noise stimulus to set the 1-m SPL to 94 dB (4.56 V), and measured the second- and third- harmonic distortion at 10 cm (see **Figure 10**). This is a well-executed AMT, especially considering Dayton built the unit around a Mundorf diaphragm. *aX*

B&K PRECISION EXPANDS WAVEFORM GENERATOR SERIES

B&K Precision announced the launch of its 4050 Series, a line of four dual-channel function/arbitrary waveform generators. These instruments can generate waveforms up to 50 MHz for use in education and applications requiring stable and precise sine, square, triangle, and pulse waveforms with modulation and arbitrary waveform capabilities.

a)
b)



B&K Precision's 4050 Series offers several modulation schemes and the display functions are clearly labeled (a). The back panels are universal for all the 4050 Series units (b).

All models are dual-channel and provide a main output voltage that can be varied from 0 to 10 V_{PP} into 50 Ω. A secondary output can be varied from 0 to 3 V_{PP} into 50 Ω. Large 3.5" color LCDs, rotary control knobs, and numeric keypads with dedicated waveform keys and output buttons make waveform adjustments quick and effortless.

Equipped with a 14-bit, 125 MSa/s, 16k point arbitrary waveform generator, the 4050 Series provides users 48 built-in arbitrary waveforms and the ability to create and load up to 10 custom 16 KPT waveforms using the included waveform editing software via a standard USB interface. An optional USB-to-GPIB adapter is available for GPIB connectivity.

The 4050 Series offers: amplitude and frequency modulation (AM/FM), double sideband amplitude modulation (DSB-AM), amplitude and frequency shift keying (ASK/FSK), phase modulation (PM), and pulse width modulation (PWM). Other standard features include linear and logarithmic sweep function, built-in counter, Sync output, trigger I/O terminal, and USB host port on the front panel to save and recall instrument settings and waveforms. A standard external 10- MHz reference clock input is provided for synchronization of the instrument to another generator, a feature not typically found in generators at this price point.

B&K Precision's 4050 Series products are backed by a standard three-year warranty. The 4052 (5 MHz) costs $499. The 4053 (10 MHz) costs $599. The 4054 (25 MHz) costs $850. The 4055 (50 MHz) costs $1,050.

B&K Precision is offering a 10% discount until November 30, 2013. Visit www.bkprecision.com for details.

LINEAR INTEGRATED SYSTEMS OFFERS NEW DUAL JFETS

Linear Integrated Systems, a full-service manufacturer of specialty linear semiconductors, now offers the LSK 489, a 1.8-nV at 1-kHz, low-capacitance, N-channel monolithic dual JFET. The LSK 489 is part of a family of ultra-low-noise, dual JFETs specifically designed to provide users with less expensive solutions for obtaining tighter drain-source saturation current (I_{DSS}) matching and better thermal tracking.

Available in surface-mount and ROHS-compliant versions, the LSK 489 is an improved, functional replacement for JFETs that have similar noise characteristics but greater gate-to-drain capacitance. The LSK 489 SOT-23 and SOIC packages are ideal for space-limited circuits in audio and instrumentation applications. LSK 489 available packages are: TO-71; SOT-23-6L, SOIC-8L. The LSK 489 combines a noise level nearly as low as the LSK 389 with lower gate-to-drain capacitance, 4 pF vs. 25 pF.

The LSK 389 and the LSK 489 feature unique monolithic dual-design construction, interleaving both JFETs on the same silicon piece to provide thermal tracking and a low-noise profile. Lead-free, ROHS-compliant versions are available. Linear Integrated Systems's in-house fabrication and domestic factory stock guarantee short lead times, ensuring no disruption in production schedules.

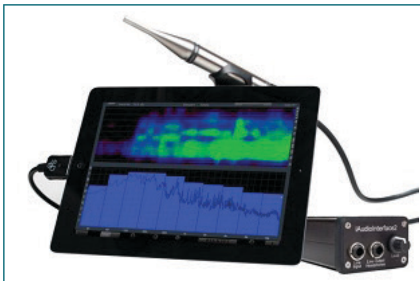
The LSK 489 can be used in several applications (e.g., microphone amplifiers, phono preamplifiers, audio amplifiers and preamps, discrete low-noise op-amps, battery-operated audio preamps, audio mixer consoles, acoustic sensors, sonic imaging, instrumentation amplifiers, wideband differential amplifiers, high-speed comparators, and impedance converters.

Linear Integrated Sytems now offers the LSK 489, a new monolithic dual JFET.

For more information, visit www.linearsystems.com.

TurboCal PROVIDES A LOW-COST SOLUTION

Home Acoustics Alliance (HAA) now offers an audio TurboCal package. This kit is designed for quick acoustic calibrations you suitable for every job. AVPro 2.0 will even walk you through the process. With this setup, you only need to add the AVPro iMux 4 and you will have the complete professional kit. Just add an iPad and a few apps.

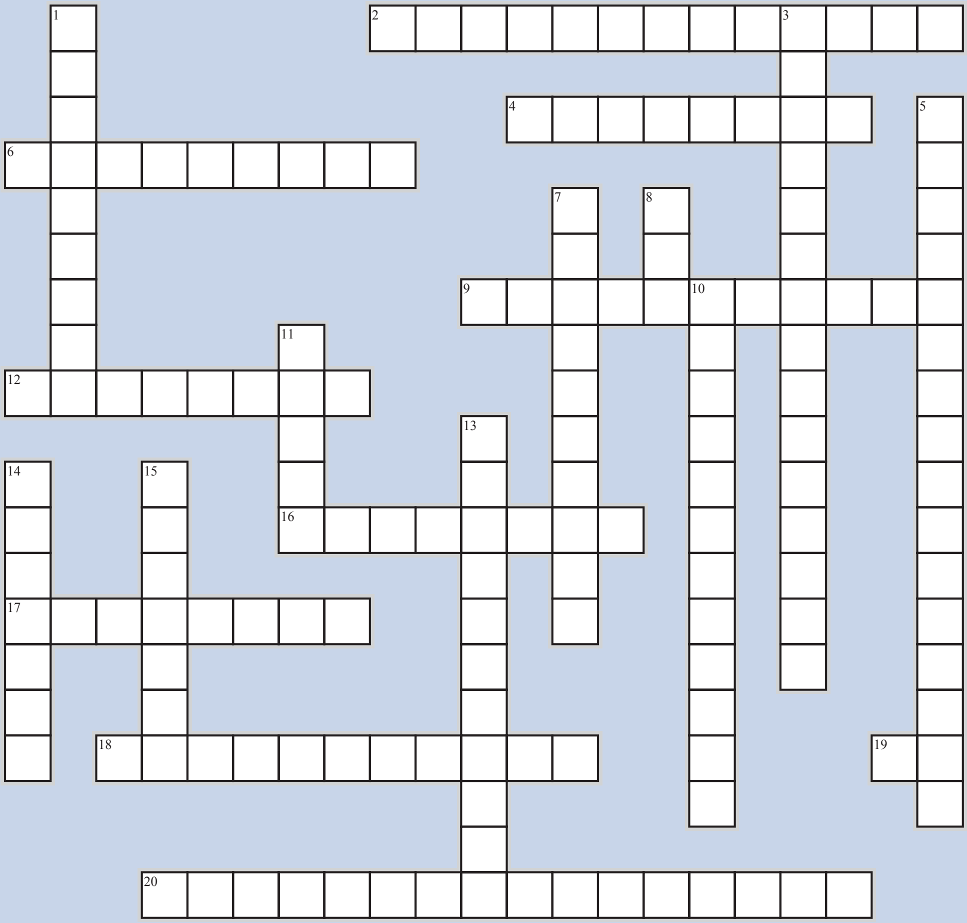


HAA's TurboCal Suite is designed for quick acoustic calibrations.

The kit includes an iAudioInterface2 preamp/generator, an HAA-approved Type 2 measurement microphone, AVPro 2.0 audio reporting software, and a SC05 sound-level calibrator. The TurboCal package costs $1,150.

For more information, visit www.avproalliance.com. *aX*

Crossword



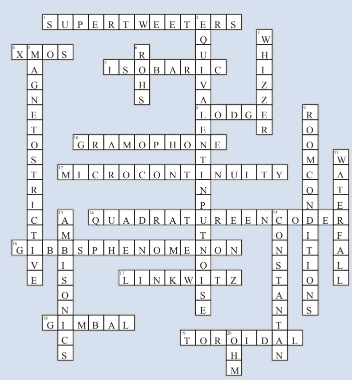
Editor's Note: Answers to this puzzle are posted at audioxpress.com/crossword and will be available in the next issue.

ACROSS

2. Known for its flat time delay and uneven amplitude response [two words]
4. FLAC, WAV, or AIFF are these file types
6. A short group of uniform-frequency and uniform-amplitude sine waves [two words]
9. Electrical signal with no discernible frequency or amplitude pattern [two words]
12. Pristinely clean reproduced sound
16. The "space" between an audio waveform's peak and the maximum available loudness
17. Ensures that digital audio transmitted across networks or physically copied from one location to another is the same when it reaches its destination
18. A circuit that extracts information from an RF signal
19. A circuit or amplifier's output impedance
20. Caused by barely audible signal distortion [two words]

DOWN

1. In SPL measurements, human hearing's threshold [two words]
3. An oscilloscope trace that shows two signals' vector sum [two words]
5. A device with several resistor pairs used as fixed-voltage dividers [two words]
7. Trademarked name of a failed RCA product
8. An upside down Ω
10. A rapid audio anomaly that occurs when playing back optical soundtracks [two words]
11. Inherently omnidirectional driver
13. Used in research, military, and commercial applications to generate microwaves
14. Introduces an inductive element into a circuit
15. An open reel of audiotape on a hub that is not supported by flanges



Here are the answers to the September 2013 puzzle.

Audio Marketplace

CONTRIBUTORS

Wim de Haan ("Mugen—A Hybrid Amplifier," p. 11) is a medical engineer with a degree in electronics and a strong interest in DIY audio. He started his first audio project at age 14 and received his first Keith Jarrett CD as a present at age 16. Some of his projects have been published in *Electronics World*, *Elektor*, and *audioXpress*. He is webmaster of the official websites of classical pianist Mikhail Voskresensky and pianist Evelina Vorontsova. His adoration of music is apparent from the websites, www.tatiana-nikolayeva.info and www.robmadna.info.

Vance Dickason ("Dayton Audio's AMT1-4," p. 38) 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*. Although he has been involved with publishing throughout his career, he still works as an engineering consultant to a number of loudspeaker manufacturers. This currently includes Samsung, The AVC Group (Niles and Sunfire), Artison, and Emotiva.

Jan Didden ("Audio Innovator Amar Bose Remembered," p. 28) has been writing for *audioXpress* almost since the start in the 1970s. He has retired from a career with the Netherlands Air Force and NATO, working in logistics, air defense, and information technology. His retirement finally gave him time to finish all those audio projects that piled up for decades, enabling him to write about them on his website (linearaudio.nl). His main audio interests are in power amplifiers and audio test equipment. Jan is also the publisher and managing editor of the twice-yearly bookzine *Linear Audio*.

Gary Galo ("OPPO Digital's New 3-D Blu-ray Disc Player Raises the Bar," p. 20) retired in October 2010 after 34 years as an Audio Engineer at The Crane School of Music, SUNY, in Potsdam, NY. Not ready to quit entirely, he works half-time in that capacity. Gary is also a graduate of The Crane School of Music, where he earned a BA in Music Education in 1973 and an MA in Music History and Literature in 1974. He is the author of more than 260 articles and reviews on both musical and technical subjects. An active member of the Association for Recorded Sound Collections (ARSC), he was co-chairman of the ARSC Technical Committee from 1996–2004 and the Sound Recording Review Editor of the *ARSC Journal* from 1995–2012. He has been a regular presenter at ARSC conferences and several of his conference papers have been published in the *ARSC Journal.* Gary is a regular contributor to *audioXpress*.

Dr. Richard Honeycutt ("Dynamics Processing with Tubes," p. 33) has been an audio enthusiast, musician, recording engineer, broadcast radio engineer (First Class Commercial licensee since 1969), radio announcer, electronics repairman, tube guitar amplifier designer, speaker system designer, pro-sound/video designer, and college electronics teacher. He is now a consultant in acoustics and electroacoustics. He earned a PhD in Electroacoustics from the Union Institute and lives in Lexington, NC, with his wife, Betty Jane, near his daughter Alyson, her husband, and two of his three grandchildren.

Mike Klasco ("Speaker Protection for Mobile Consumer Products," p. 8) is the president of Menlo Scientific, a consulting firm for the loudspeaker industry, located in Richmond, CA. He is the organizer of the Loudspeaker University seminars for speaker engineers. Mike specializes in materials and fabrication techniques to enhance speaker performance.

Steve Tatarunis ("Speaker Protection for Mobile Consumer Products," p. 8) 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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