SPEAKER CONES EXPLAINED, PAGE 7

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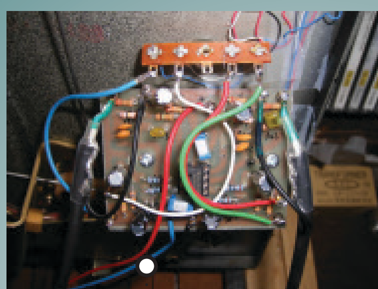

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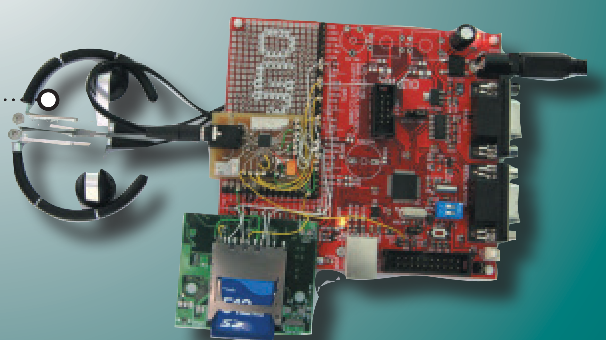

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CONTENTS

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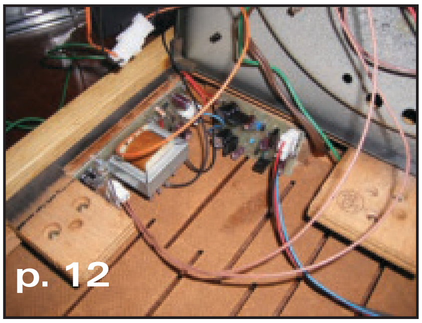
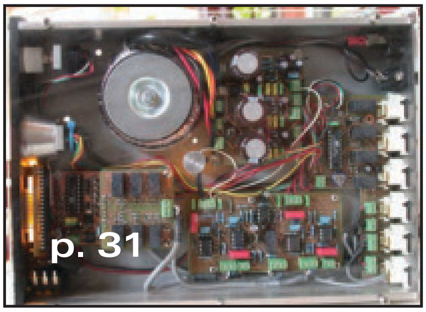
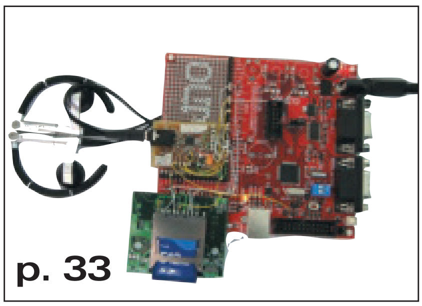
FEATURES







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When Bad Sounds Good

As many of you know, the northeastern corridor of the United States was hit by an unusual late-October snowstorm. During about 24 hours, the storm piled more than a foot of heavy snow on top of trees, many of which still held an unusually large number of leaves for that time of year. The result was disaster. Trees fell and took out power lines. And as a result, more than 800,000 customers in Connecticut alone were without power. Most of this magazine's staffers were among those unlucky residents.

During that time, something interesting occurred. With downed trees preventing road usage, damaged cellular phone towers preventing calls, and nonexistent Internet service, many people were cut off from the world outside their neighborhoods. The one exception was the battery-powered radio. That's right. To get our news and to entertain ourselves, we turned to our battery-powered (or even hand-cranked) plastic systems with horrible sound.

Like most people in our state, my neighbors and I used our battery-powered radios for essential news updates, such as information about road closures, clean-up estimates, and the locations of shelters. But there was more. In the time of expensive high-end audio systems, MP3 players, and 'Net-based streaming content, we used our trusty AM/FM radios to entertain ourselves and, frankly, to keep sane. A local rock station lightened the mood for adults and kids alike as my neighborhood gathered in a backyard to eat grilled hot dogs, sip some beer, and share damage estimates. And it was my $20 AM/FM "sports radio" that enabled me to tune into the always-informative and entertaining *John Batchelor Show* (77 WABC New York) each night at 9 p.m. when it got too dark for reading by candlelight in my house.

audioXpress readers and staffers all admire quality sound, and most of us love this magazine for the DIY audio projects and reliable information about the top products to bring the best sound to our ears. But sometimes, when an uncontrollable event like a massive storm occurs, we're forced to change our focus from the quality of sound to the actual content communicated through the sound. Take my word for it. When you're commiserating with neighbors about storm damage around a small grill, a little Rolling Stones pushed out of a clock radio is a real pick-me-up. And when 9 p.m. comes around and you've got nothing to do for three hours until you fall asleep, a static-infused talk radio show sounds, well, golden.

C. J. Abate
editor@audioxpress.com

Everything You've Wanted to Know About Speaker Cones

A look at materials, fabrication, and quality

The cone is the most critical component of the speaker and has a significant contribution to its sound quality. When mounted in an enclosure, it is the only visible component to the end-user. If the speaker design engineer spends much of his precious bill of material budget on a high-performance flat wire coil, no one will know. But, spend a few extra bucks on a woven Kevlar cone and you can't miss the yellow high-tech look. Consider the speakers you have bought (or speaker projects you have put together)—more likely than not, the look of the speaker cone probably was an influence.

Most speaker cones are fabricated from a recipe of wood (cellulose) fibers in a paper-making process, but polypropylene, woven glass, carbon fiber, Kevlar composites, and metal cones are also popular choices. Getting the soft parts right in a speaker driver is an art. The cone, surround, spider, and dust cap are the secret sauce in speaker design and most critical are their ingredients and fabrication.

Loudspeaker cones are most commonly formed from paper pulp, but plastics such as polypropylene are also popular, and this material can be thermoformed (like melting hot cheese over a form) or injection-molded into the desired shape. Metal cones also appear on the scene, and other high-tech solutions that have achieved success include honeycomb, foam laminates, and composites such as woven and non-woven resins, glass fiber, carbon fiber, and aramids (such as Kevlar).

Typically a specialist cone manufacturer fabricates cones for speaker assemblers, although some of the larger offshore speaker companies have their own in-house cone fabrication facilities.

PAPER PULP CONE FABRICATION

The process starts with sheets of various types of pulp such as bleached and unbleached Kraft pulp. Douglas fir from Canada or the southern United States is common. Exotic blends which include eucalyptus from Brazil or Australia (very stiff), other specialty pulps from New Zealand, or the hemp (Fostex and Dai-Ichi like banana leaves which are part of the hemp family), kapok fibers (poor man's Kevlar), or various synthetic fibers are also popular. The pulp recipes, additives, pulp-slurry beating process, and cone-forming techniques all contribute to the loudspeaker cone's characteristics. Young's modulus (speed of sound), tan delta (internal damping), and mechanical parameters such as tear strength, burst strength, and so on are all factors that separate the toy cones from the audiophile, studio monitors, electric guitar, or prosound diaphragms.

Other considerations are wet strength and moisture regain. Will the cone fail if used in a humid environment or will that studio monitor sound the same on a humid day? What about vulnerability to sunlight (UV), fungus, ozone resistance, and so on? Appearance is still another consideration, and there are many secondary surface treatments that add what cannot be fully achieved within the paper-beating process.

Once the recipe is selected for a particular production run, the appropriate paper pulp is soaked in hot water, for a period of time determined by the "chef." Pieces are torn off the wet pulp sheets (by hand or by machine) and thrown into a water-filled pulp beater (see **Photo 1**).

The beating process disperses the fibers while also fibrillating (fuzzing them up) so they will mechanically tangle together, thereby holding the cone together. Aside from the mechanical bond there is also the oxygen bond from the paper-slurry process. Most of the industry uses the old-style beating machines which offer potentially excellent fibrillation. Sometimes hydropulpers are used, which can be faster in producing the slurry but may not do as much fibrillating work on the fibers. Better use the hydropulper for making newspaper instead.

Measurement techniques such as Ca-

Photo 1: Beater shown making a batch of pulp

Photo 2: Canadian Standard Freeness (CSF) tester for monitoring batch-to-batch consistency for pulp fibrillation

nadian Standard Freeness (CSF) are used for monitoring batch-to-batch consistency for pulp fibrillation (see **Photo 2**). Freeness has to be in a certain range in order for the fibers to bond and make good paper.

Additives are thrown into the beater, such as salts to hold the dyes (mostly black), waterborne epoxies, and all this is mashed some more. In the case of the beating machine, there is a beating wheel, which is progressively brought closer to the beater bedplate as the pulp is worked into a slurry of fibrillated fibers.

Eventually this dense soup is transferred to a holding tank, which has an agitator to keep the slurry dispersed and homogenized in the tank. The pulp's fiber length, density, and secret sauce are factors that distinguish different speaker cones intended for subwoofers, midranges, musical instruments, and so on.

There are three common cone paper forming techniques: pressed (see **Photo 3**), semi-pressed (see **Photo 4**), and non-pressed cones (see **Photo 5**).

In the case of the pressed cone, the pulp slurry may be dumped into a bin and drained through a fine mesh screen. The paper fiber solids are left on this screen after the liquids have been drained out of the bin. The carcass is removed from the mesh and deposited onto the cone-pressing machine. Heated positive and negative metal cone-shaped forms then press the pulp. The remaining water steams out of the cone, which is removed from the forming tool.

If the cone's density is too low, then the cone may be too dead-sounding. If the cone is pressed too hard and thin, then breakup (cone cry) will be more noticeable, especially at high sound levels. The cone density and pulp composition affects the internal loss (deadness) of the cone material, which will contribute to the sound quality of the speaker.

The speaker cone shape or profile is a critical factor in the sound quality and performance of the speaker. If the cone body angle is narrow, the walls straight-sided with concentric reinforcing ribs molded in the cone body, it will be the strongest and most rigid at very low frequencies.

Decoupling rings can be molded into the cone body, which look like ribs, but a cross-section of the cone shows that while ribs are composed of added material on the face of the cone, decoupling rings are concentric cone corrugations. These corrugations allow the effective radiating area of the cone to decrease with rising frequency. The off-axis response tends to look better over an extended frequency range with decoupling rings (and curvilinear profile), allowing the speaker system designer the option of using a higher crossover point between the woofer and tweeter or midrange.

The angle at which the cone body attaches to the voice coil bobbin is important. If the cone meets the voice coil former at too sharp an angle then the high frequencies will reflect back rather than pass into the cone body, resulting in a ragged response. Cone body weight is also important. Too light and the cone may distort, producing cone-cry at high sound levels or high excursion. Too heavy and efficiency will suffer. The larger the voice coil for a given size cone, the greater the structural integrity of the cone body, making it less susceptible to breakup. Also, if the cone is deep, it is less likely than a shallow cone to "ashcan," buckle, and emit spurious noises. Today, cone and speaker designers have software simulation programs (such as FINECone from LOUDSoft) to model results before starting tooling.

Some paper cones also have non-cellulose fibers mixed in, but getting all these ingredients to stick together is critical. Waterborne epoxies, pastes, and synthetic fibrillating fibers are sometimes used in the slurry. These special addi-

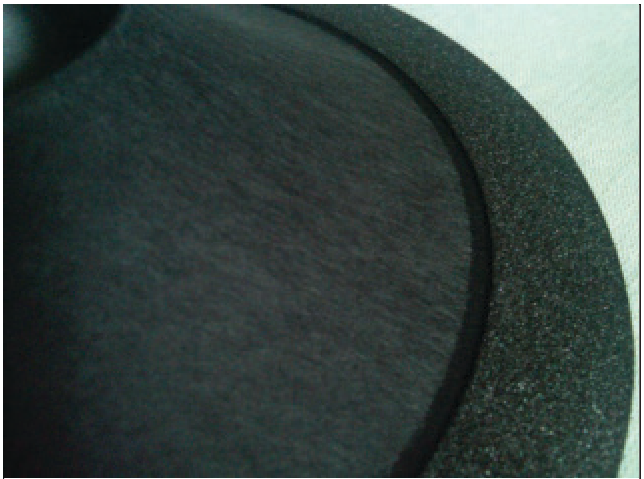

Photo 3: Pressed cone

Photo 4: Semi-pressed cone

Photo 5: Non-pressed cone

tives are processed in the slurry before loading reinforcements such as Kevlar, glass, mica, or ceramics. The treelike nature of fibrillated fibers mechanically hold everything together. Fibers may be fibrillated (many fibrils—fuzzy), short sticks (stable fibers—typically 0.25" long is a good aspect ratio for stiffness), or flocked fibers and flakes.

Adding carbon fiber or aramids (such as Kevlar or Conex) fibers to the pulp allows reducing the cone body weight, while retaining strength. Many speaker designers believe aramid fiber paper cones are audibly cleaner, especially with wide dynamic range music. Kevlar has the rare combination of extreme stiffness and good damping. Other high-performance fibers include Spectra and Vectran. Most hard materials tend to be "noisy." These premium fibers are the caviar of cone materials. Starting at well over $10/lb., it is quite a bit more expensive, while commodity paper pulp costs a small fraction of this. DuPont does not readily sell Kevlar pulp, and most so-called "Kevlar" paper cones from off-shore vendors actually contain glass reinforcements, or sometimes alternative aramids such as Conex or Technora from Teijin. Cones made in the United States from Loudspeaker Components are made with genuine Dupont Kevlar.

Glass fiber paper cones are sometimes used for subwoofers. While these cones may appear to be paper, you can immediately feel the high stiffness if

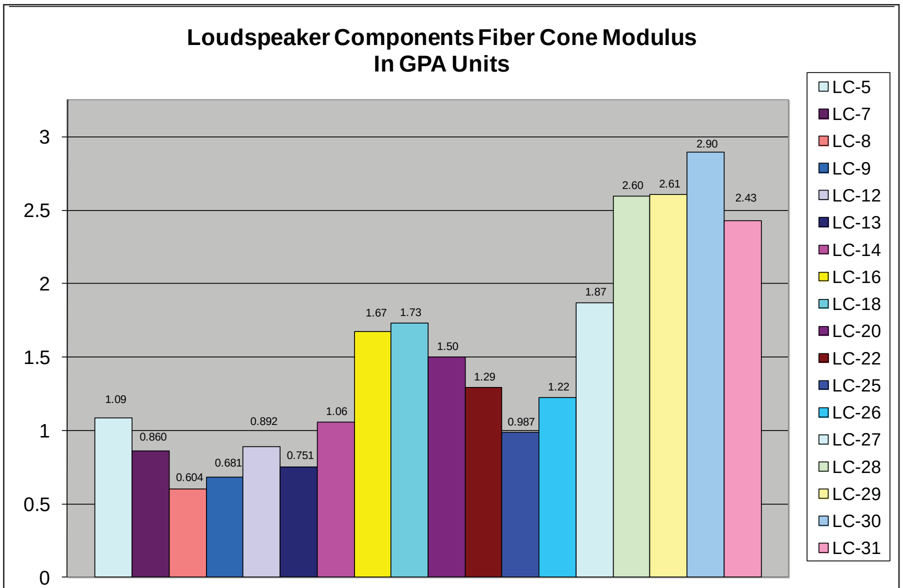

Figure 1: Viewing the graph from left to right, significant Young's modulus advancements have been made over the past 15 years by Loudspeaker Components with the data furthermost to the right derived from X5 technology pulp by LC. (Courtesy of Loudspeaker Components (LC), United States)

you tap or flex the cone. The glass fibers have a "sizing," a special coating such as acrylic so it consolidates (sticks to) the cellulose fibers in the slurry. Keeping the glass fibers dispersed in the slurry is a challenge because their specific gravity is quite a bit denser than the cellulose pulp fibers and the glass fiber wants to drift toward the bottom.

To deaden the cone, sometimes wool or a flocked polypropylene pulp is used as an additive to the paper pulp. In some cases, a goo may be sprayed or brushed onto the cone, often on the rear, to dampen resonance.

Significant Young's modulus (stiffness) advancements have been made over the past 15 years by United States cone supplier Loudspeaker Components (see **Figure 1**).

POLYPROPYLENE CONES

Polypropylene cones are popular for both high-end audio and autosound because they do not absorb moisture (moisture regain) and can have low distortion. Avoiding moisture absorption is more important than you might think because the cone mass, Q, resonance frequency, and box tuning (on vented designs) can all significantly shift with humidity. Conventional fabrication of poly cones is often by thermoforming, where the poly sheet is heated until it is soft and then it is formed (often by vacuum pressure) into shape.

Poly comes in different formulations and its performance highly depends on additives. As with vinyl records, if too much waste from previous production runs are chopped up and recycled into new runs, performance and consistency may suffer. On the other hand, some "regrind" waste actually has some performance benefits. Like paper, there are many additives and fillers used in poly, with mica, carbon black, talc, and glass being most common.

Many cone manufacturers are using injection-molded poly cones, but require a large investment in tooling. The injection-molding process tends to orient the polymer and adds stiffness.

An innovative approach to injection-molded poly cones is Pioneer's IMPP and Panasonic has a similar technique. The process starts out conventionally, using poly pellets (and other ingredients), which are melted and injection-molded into the cone mold. The final step is unusual as the pressure is removed from the mold allowing the material to foam, which increases rigidity.

Other techniques Pioneer has used to further increase the cone stiffness include "diamond-plate" and conical dome contours in the cone walls and one piece cone construction (no dust cap). The face of the cone has a nickel metallization for dramatic aesthetics. Panasonic has a few noteworthy innovations for injection-molded poly, and they have their own forming technique for poly, as well as a variation in the cross-section of the cone for structural reinforcement—and this is all done in-house in their central Japan operation.

WOVEN AND NON-WOVEN HIGH-TECH FIBER CONES

The high-end of the market is where we find the use of woven carbon fiber or non-woven fiber mat saturated with poly or other resins. Fibers include Kevlar, known for its yellow color, high strength, light weight, and excellent sonic characteristics. Carbon fiber, typically black (but red is another option), is another high-performance fiber.

Glass, which is most often used in woven white cones, has very high strength—especially the high-performance grades like E and the even stiffer and stronger S glass. But, it is best used in woofers and subwoofers due to its high density. The resin binder in a woven cone is a big contributor to the performance and weight, and if excessive epoxy is used to bind and seal the cone, the weight benefits of using expensive high stiffness-to-weight ratio fiber may be lost. Poly resin provides damping and low weight but requires proprietary techniques.

HIGH-TECH RIGID FOAM

Rohacell (polymethacrylimide foam) is a product of military aircraft construction. This lightweight but stiff foam is used for the radar domes on AWAC aircraft, but a more down-to-earth application is flat and cone-shaped diaphragms. DIAB's Divinycell is another vendor for foam cores. These are good solutions for high-excursion tight boxes whose pressures otherwise would crush the cone.

Forming is slow, using hydraulic presses. A wide range of skins are typically laminated to form or honeycomb cores.

METAL CONES

Metal is a unique approach to speaker diaphragms that has been around for many decades. But in the last few years, it has become more commonplace in high-performance speakers. Early Western Electric speakers in the 1930s and later Bozak drivers developed in the 1950s used aluminum cones.

Because metal is inherently stiff, the resulting strong and light cones can withstand high excursion with low distortion. High excursion translates to high-acoustic output, especially at low frequencies. And when you are putting big woofers in small boxes, the metal cone withstands the enormous forces that want to burst and crumple the cone. Still another strong point is environmental stability. Typically metal cones are some alloy of aluminum. Forming techniques include spinning, hydroforming, and punching. Today most cones are punched. Aluminum alloys such as Duralumin (also called duraluminum) are common and reasonably priced, but magnesium aluminum metal matrix alloy and beryllium cones are options if you want a cone costing in the $100 range.

The speaker engineer's requirement for stronger, lighter, shallower, and acoustically better-performing cones is only matched by the marketing department's hunger for product differentiation. The cone is the most significant path to resolving these goals, so count on its ongoing evolution.

Next month we will take a close look at the surround and spider. The speaker cone reproduces sound through back-and-forth movement in a mechanical analogy to the electrical signal. The compliance(s)—the surround and spider—guide the cone, keep the voice coil centered in the magnetic gap, provide restoring force, and limit excessive excursion. The various types of compliances will be discussed, including shape, materials, and fabrication. *aX*

 By Hiroshi Obama (Japan)

Rebuild an Old Turntable

A simple circuit and some tinkering enable you to realize clean analog sound

In the 1970s, many electronics manufacturers, including Japanese Hitachi, Toshiba, Matsushita, and so on, produced various audio equipment. It was the heyday for audio lovers. The products were later abandoned and turned to junk units. They are available at incredibly cheap prices on Internet auction markets.

I found a nice Hitachi semiautomatic analog record player for only 1,000 JPY (about $12.50). The reason for the cheap price? The pickup did not have a head. After I received it, I realized the tonearm was broken. Junk is junk! I purchased another junk player of the same type equipped with the arm and head. In Japan, it is popular to pick up usable parts from two units then make a perfect one. We call this "2 ko 1," meaning two into one.

Photo 1: This is a front view of a modified turntable. The blue LED indicates that the player system is on.

UPGRADE

It then made sense to upgrade the player rather than simply restore it. I was determined to convert the player to an "active" system by installing an equalizer with a power supply inside the player. The player, a Hitachi SP-10, is small and the inner space is mostly occupied by the semiautomatic gear (see **Photo 1**). It looked prohibitively difficult to install the amplification units inside the player (see **Photo 2**).

Photo 2: Rear view of the modified turntable

After I measured the vacant space left inside, I realized that the only way to equip electronic units inside was to divide the PCB into three parts: a main power supply, a regulated power supply with ripple filter, and an equalizer, which you should place right under the tonearm to prevent noise and realize good sound reproduction without unnecessarily long wiring (see **Photo 3**). This guarantees pure sound and suggests why most professional turntables are made this way. You would no longer be tempted to choose expensive shielded cable with gold RCA pin plugs!

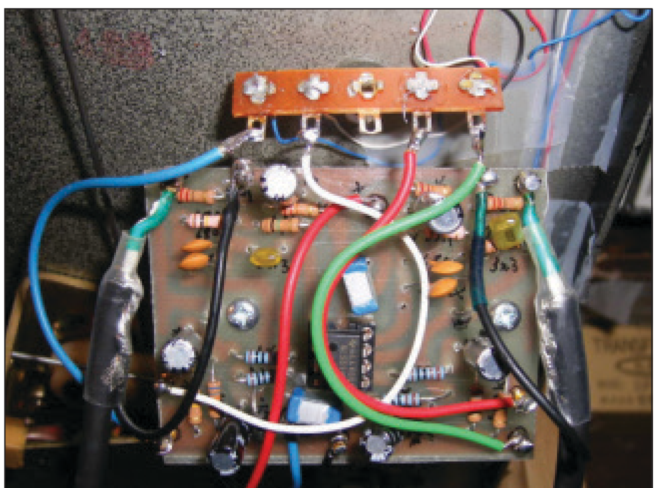

Photo 3: Phono equalizer PCB placed under the pickup arm

SIMPLE CIRCUIT

The rather simple circuit adopted here is not a so-called DC-type amplifier, which is fine for a digital system (see **Figure 1**). But if you applied the method to an analog LP equalizer, it would feed a horrible low frequency generated by a warped black disc to an honest modern speaker through a DC amplifier. You can see the speaker woofer weirdly trembles, and higher frequency reproduction is affected with intermodulation distortion. An old "good" LP record system lives in a world rather different from a modern digital system.

The three PCBs are equipped with connectors for easy maintenance, as with most professional components. I hand-drew the printed circuit board—quite an old-fashioned way, but reasonable for a single product (see **Photo 4**).

I have referred to the *Voltage Regulator Applications Handbook,* (Fairchild Semiconductor, 1974), for the application of the regulated power supply with ripple filter (see **Figure 2**). Because the phono equalizer has a large gain, it is advisable to install the

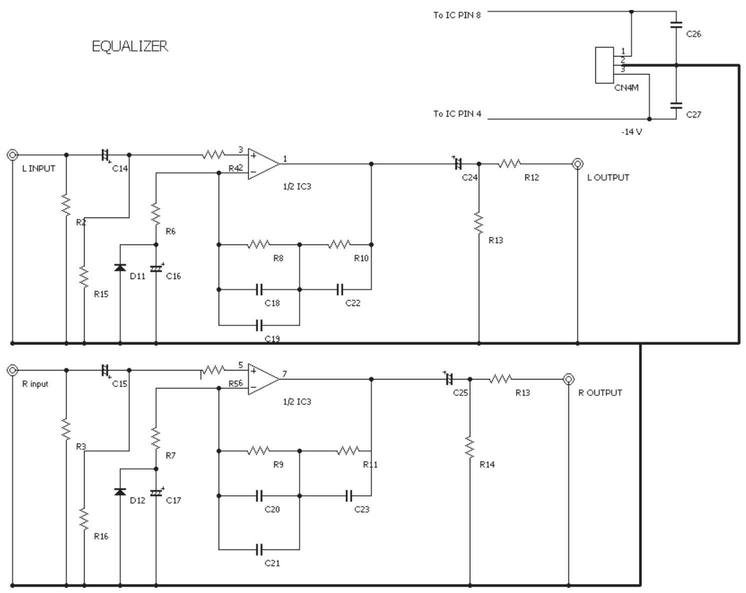

Figure 1: Equalizer circuitry

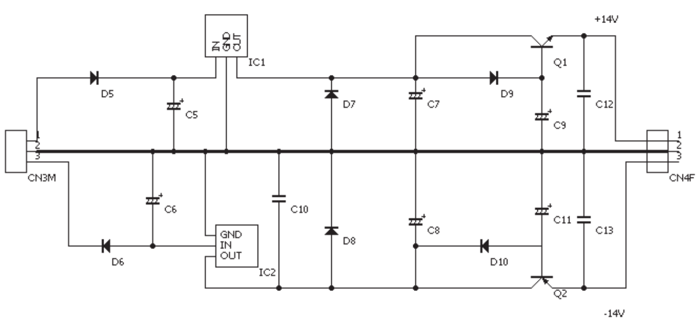

Figure 2: Regulated power supply with ripple filter

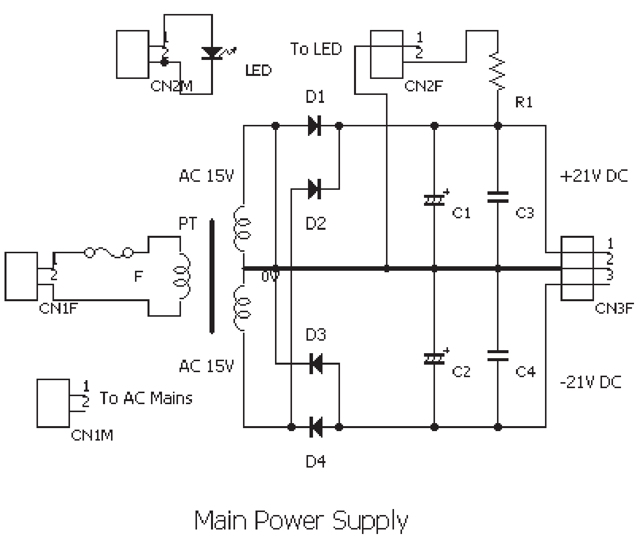

Figure 3: This is the main power supply. The unregulated DC 21-V positive and the 21-V negative is made from AC 15 V × 2 via D1 through D4

ripple filter (see **Photo 5**). Here, small transistors are doped to work for this purpose. In theory, the cap value of C9/C11 is multiplied by the transistor's small signal (Hfe) forward current gain (see **Figure 3**).

I borrowed the topology from a McIntosh C33 pre-amp, with alteration to the value of RIAA network elements. McIntosh is a great amp; however, I believe the network impedance is very low, which may be too heavy for the OP-IC. Therefore, I adopted a popular CR value. D11/D12 is designed to protect C16/C17. In case over –0.6 V appears across the cap, the diode works as a short circuit. Place C26/C27 close to IC pins 8 and 4, respectively, to prevent unnecessary oscillation.

I kept the outlook as it was with one exception—adding a blue LED on the deck to indicate the system is connected with mains. The amplifier is always "ON" because it has no AC switch, while the motor is supplied with AC only when the pickup is in operating position. (Hitachi designed it that way, which I think was wise.)

PHONO CARTRIDGE

The original Hitachi pickup cartridge is no longer available. Because the arm is designed to accommodate its Hitachi genuine cartridge, the arm has no


Photo 4: Hand-drawn PCB

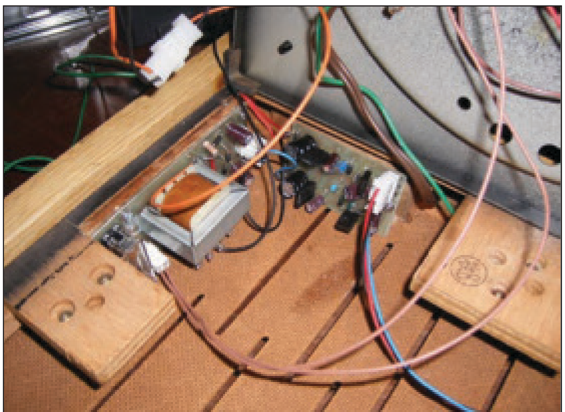
Photo 5: The PCB (left) with a transformer is the main power supply. The other PCB (right) is a regulated power supply with a ripple filter.

elevation height adjustment. The original phono cartridge is rather short in terms of height; therefore, if you look for any alternative, choose one with low height. I tried a SHURE M44-7 and found it suitable.

The player generates equalized signal with the output level almost identical to CD players. The idea to install an equalizer in the player is common among professional analog players; and you may adjust RIAA elements after you measure the actual frequency response in accordance with your cartridge so you can enjoy the total sound quality.

It seems to me that using this method, you can upgrade any old popular analog LP player, not just Hitachi units. You'll be surprised how cool an equalizer installed in the player sounds, eliminating noise without dropping any fine signal. Discover for yourself how you can create a beautiful player system from junk units. *aX*

Parts & Components

Capacitors		
• C1,C2	100 µ 50 V	
• C3,C4	0.1 µ	MKH
• C5,C6	330 µ 45 V	
• C7,C8	4.7 µ 24 V	
• C9,C11	220 µ 24 V	
• C10,C12,C13	0.1 µ	MH
•C14,C15,C24,C25	10 µ 35 V	
• C16,C17	22 µ 16 V	
• C18,C20	680 P	Ceramic
• C19,C21	68 P	Ceramic
• C22,C23	3300 P	Polyester Film
Miscellaneous		
• PT	Transformer	15 V 0.5 A × 2
• D1-D12	1N4002	
• LED	Blue	
• CN1M	2P Male Connector	
• CN1F	2P Female Connector	
• CN2M	2P Male Connector	
• CN2F	2P Female Connector	
• CN3M	3P Male Connector	
• CN3F	3P Female Connector	
• CN4M	3P Male Connector	
• CN4F	3P Female Connector	
• F	Slow Blow Fuse	0.5 A
Resistors		
• R1	3.3 kΩ	0.5 W
• R2,R3	150 kΩ	0.25 W
• R4-R7	1 kΩ	0.25 W
• R8,R9	100 kΩ	0.25 W
• R10,R11	1 MΩ	0.25 W
• R12,R13	220 Ω	0.25 W
• R14,R15	2 kΩ	0.25 W
• R16,R17	68 kΩ	0.25 W
• IC1	Voltage Regulator	15 V Positive 7815
• IC2	Voltage Regulator	15 V Negative 7915
• IC3	OPIC NE5532	
• Q1	2SC1567	
• Q2	2SA794	

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System power 2000watt

4 Front Speaker & Rear
- impedance: 8ohm
- Sensitivity: 86dB
- Frequency: 120-20Hz
- Loudspeaker Unit: 2" x 4+1"

Centre Speaker
- impedance: 8ohm
- Sensitivity: 85dB
- Frequency: 250-20000Hz
- Loudspeaker Unit: 2" x 2+1"

Subwoofer Speaker
- impedance: 8ohm
- Sensitivity: 87dB
- Frequency: 30-220Hz
- Loudspeaker Unit: 10"

Carton Dimension (mm)
- 267x1238x275 (HxWxD)

Subwoofer:
- 505x315x515 (HxWxD)

MSRP:$3,250.00

K-7

System power 1500watt

4 Front Speaker & Rear
- impedance: 8ohm
- Sensitivity: 85dB
- Frequency: 250-20000Hz
- Loudspeaker Unit: 2" x 2"

Centre Speaker
- impedance: 8ohm
- Sensitivity: 85dB
- Frequency: 250-20000Hz
- Loudspeaker Unit: 2" x 2+1"
- Power : 100w

Subwoofer Speaker
- impedance: 8ohm
- Sensitivity: 87dB
- Frequency: 30-220Hz
- Loudspeaker Unit: 8" active
- Power : 100w

Carton Dimension (mm)
- 265x695x505 (HxWxD)

MSRP:$3,999.00

KA-4000

HIGH DEFINITION HOME THEATER

System:
- 5.1 multi channel home theater
- 2000 watt dynamic peak power

Cabinet:
- Low resonance MDF construction-subwoofer
- Low resonance MDF construction-front channel towers
- Low resonance MDF construction-center channel
- Low resonance MDF construction-surround channel towers
- Black matte finish w/ black high gloss accents
- Black mesh cloth grill

Specifications:
- 10 inch active down firing long throw woofer
- Frequency response: 50hz- 20khz
- Nominal impedance 6ohm • Sensitivity 93db • SPL 106

Features:
- Digital 5.1 surround sound amplifier
- LCD multi function display
- VU meter/ measurement
- Full function remote control
- MP3 feature
- Plug n play
- Direct hook up to TV
- AM-FM digital tuner

*Model number and manufacturer's suggested prices are for identification purpose only.
Manufacturer's suggested price do not represent a bona fide selling price in the metro trading area.
Manufacturer reserve the right to make changes on specification and material.

KA-6100

KA-8100

KA-5100

MSRP:$2,495.00

KA-8100

System:
• 5.1 multi channel surround sound speaker system
• 1500 watt dynamic peak power

Cabinet:
• Low resonance mdf constructed subwoofer
• ABS 180 degree multi directional cube-surrounds
• 5 acoustic matched surrounds-center channel
• Pro studio matte finish subwoofer
• Black pro matte finish center-surrounds

Specifications:
• (1) 8 inch long throw passive side firing subwoofer Areodynamic Air Port Design
• (10) 2.5 inch shielded poly-coated midbass driver
• Frequency response: 60hz- 20khz • SPL 103db
• Nominal impedance 8ohm • Sensitivity 93db

Features
• 4 x 180 degree omni directional cube speakers
• Full size shielded center channel
• Gold binding post • Easy hook up cd/ dvd/ blue ray
• System high definition ready • Wall brackets
• Speaker wire included

MSRP:$1,999.00

KA-6100

• 5.1 muti channel surround sound speaker system
• 1000 watt dynamic peak power

Cabinet:
• Low resonance mdf constructed subwoofer
• abs 180 degree multi directional cube-surrounds
• 5 acoustic matched surrounds-center channel
• Black high gloss finish subwoofer
• Black matte finish center-surrounds

Specifications:
• (2) 6.5 inch metallic finished passive front firing subwoofers
• (10) 2.5 inch shielded poly-coated midbass driver
• Frequency response: 60hz- 20khz • SPL 103db
• Nominal impedance 8ohm • Sensitivity 93db

Features
• 5 180 degree omni directional cube speakers
• Gold binding post • Easy hook up cd/ dvd/ blue ray
• System high definition ready • Wall/ ceiling brackets
• Speaker wire included

MSRP:$2,495.00

KA-5100

Bass Alignment sealed cabinet
Recommended Amp.
Requirements: 5-1000
Subwoofer: 250 watts
Surround Sound
Speaker (4): 600 watts
Center Channel Speaker: 150 watts

Subwoofer
Peak Power handling:
250 watts
1 active 10 inch long throw driver
1 passive radiator: 10 inch
Frequency Response:
50 HZ- 150 HZ
Nominal Impedance: 4 ohms
Sensitivity: 89 db +/- 1 watt /1/1meter
SPL (dba): 105

Center Channel Speakers
Peak Power handling: 150w
2 X75mm mid bass driver
1 X 20mm gold dome tweeter
Frequency Response:
150 HZ- 20 HZ
Nominal Impedance: 4 ohm
Sensitivity: 90 db /
SPL (dba): 103

Front / Surround Channel Speakers
Peak Power handling: 150 W
2 X75 mm mid bass drivers
Frequency Response:
150 HZ- 20HZ
Nominal Impedance: 4 ohm
Sensitivity: 90 db
SPL (dba): 103

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Manufacturer reserve the right to make changes on specification and material.

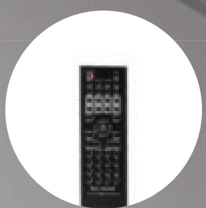
Loud Speaker
Sound Through Energy
www.kineticloudspeaker.com

High Definition Home Theater

KA-4260

KA-4220

KA-4210



Full Function Remote

MSRP:$3,499.00

KA-4260 *Gold Series*

- 5.1 multi channel home theater system
- 1500 watt dynamic peak power

Satellite Speakers

- System: 2 way 2 speakers
- Loudspeaker
- Mid-Bass: 1x4"/Tweeter: 1x1"
- Nominal Impedance:8ohm
- Sensitivity: 85±3db
- Frequency Response: 80Hz-20000Hz
- Dimensions(HxWxD): 140x180x286(mm)
- Net Weight: 2.3 kg

Center Speaker

- System: 2 way 3 speakers
- Loudspeaker
- Mid-Bass: 2x4"/Tweeter: 1x1"
- Nominal Impedance: 8ohm
- Sensitivity: 85±3db
- Frequency Response: 80-20000Hz±20%
- Dimensions(HxWxD): 440x180x153(mm)
- Net Weight: 3.9 kg

Subwoofer

- Loudspeaker
- Mid-Bass: 1x10"
- Nominal Impedance:8ohm
- Sensitivity: 85±3db
- Dimensions(HxWxD): 350x350x450(mm)
- Net Weight: 15 kg

Subwoofer Features

1)With MP3, DVD, AUX Input

Features.

1) VFD, 5 Channel Output
2) Infared Remote Control
3) 5.1 Home Theater
4) Digital AM/FM Tuner

MSRP:$2,495.00

KA-4220

- 5.1 multi channel home theater system
- 1000 watt dynamic peak power

Cabinet:

- Low resonance mdf constructed - subwoofer
- ABS multi directional cubes- surrounds
- Black high gloss finish
- Specifications:
- 10 inch active down firing subwoofer
- Frequency response: 60hz- 20khz
- Nominal impedance: 6ohm
- Sensitivity 89db

Features:

- AM-FM digital tuner
- LCD VU display
- VU meter/ measurement
- Full function remote control
- MP3 ready / plug n play
- Direct hook up to TV

Finish:

- Black gloss - subwoofer
- Black matte - surround

MSRP:$3,495.00

KA-4210

- 5.1 multi channel home theater system
- 1500 watt dynamic peak power

Cabinet:

- Low resonance MDF construction- subwoofer
- Low resonance MDF construction- surrounds

Specifications:

- Dual subwoofers
- 10 inch active down firing subwoofer
- 8 inch passive side firing radiator
- 3 inch membrane mid/bass drivers
- 1 inch poly coated soft dome tweeters
- Phase –aligned 2 way crossover
- Frequency response: 50hz-20kzh
- Nominal impedance: 6 ohm / Sensitivity: 90db
- 12 degree off-set baffling for wider sound

Features:

- AM-FM digital tuner / LCD VU display
- Full function remote control / MP3 ready
- Direct hook up to LCD/plasma TV's

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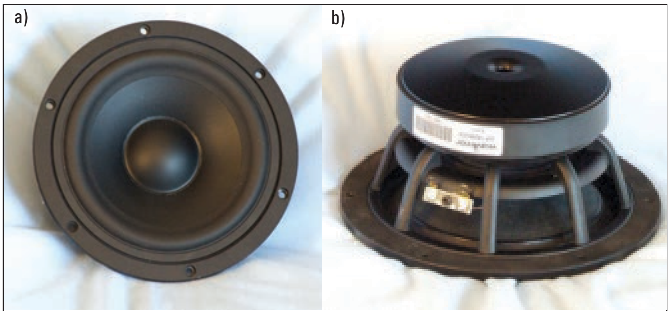
By Vance Dickason (United States)

Wavecor and Scan-Speak Drivers

A test of two high-end home audio drivers

This month's Test Bench driver submissions came from Wavecor and Scan-Speak, representing high-end home audio drivers. From Wavecor, I received a 7" paper cone midwoofer, the WF182BD04-1, and from the Scan-Speak Discovery line of drivers, a 4" midwoofer, the 12W4524G00.



a) b)

Photo 1: Front (a) and back (b) views of the WF182BD04-01

WAVECOR WF182BD04-1

The highlight of this Wavecor midbass driver design (see **Photo 1**) is the Wavecor Balanced Drive system, a proprietary distortion reducing motor technology. Basically, Wavecor's Balanced Drive technology takes the form of a tapered, extended pole as seen in the FEA diagram in **Figure 1**, which shows the dual tapered outlets on the pole vent, used to decrease turbulence in the vent. As with any extended pole, the result is a more linear Bl curve, as you will see in the Klippel analysis of this product.

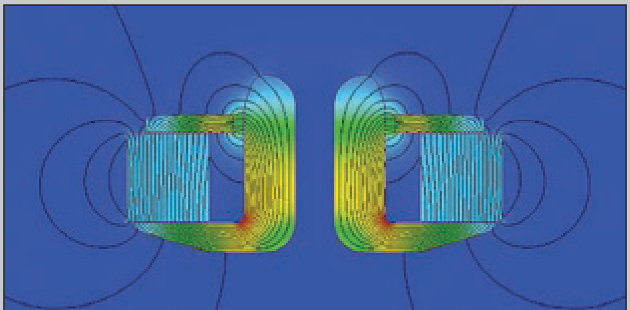


Figure 1: Wavecor Balance Drive FEA diagram

In terms of features, like the WF152BD, the WF182 is built on a six-spoke cast aluminum frame that sports a completely open area below the spider-mounting shelf for enhanced cooling. The WF182 cone assembly includes a very stiff curvilinear black-coated, semi air-dried cone with a 2"-diameter convex, black-coated paper dust cap. These are suspended by a NBR butyl-rubber surround that has a nice shallow angle where it attaches to the cone edge and a 4"-diameter black conex flat spider. Driving the assembly is a 1.5"-diameter (39 mm) voice coil using round copper wire wound on a black non-conducting fiberglass voice coil former that incorporates a series of eight 4-mm diameter former vents just below the neck joint.

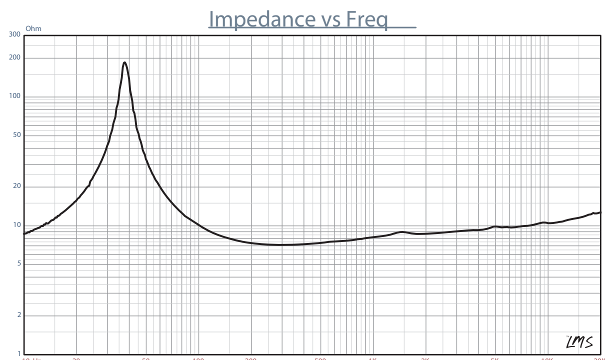


Figure 2: Wavecor WF182BD04-01 midwoofer free-air impedance plot

The motor uses a single 20 mm × 117 mm ferrite magnet sandwiched between a shaped T-yoke and 5-mm high front plate, both with a black emissive coating for enhanced cooling performance. For additional cooling, the T-yoke incorporates a dual flare 10-mm diameter pole vent. The motor also incorporates an aluminum Faraday shield/shorting ring as well as a copper cap on the top of the pole piece for distortion reduction. Voice coil tinsel lead wires are terminated to a set of gold-plated terminals.

I began testing the WF182BD04 midbass woofer using the LinearX LMS analyzer and VIBox to produce both voltage and admittance (current) curves with the driver clamped to a rigid test fixture in free-air at 0.3 V, 1 V, 3 V, 6 V, and 10 V. As has become the protocol for Test Bench testing, I no longer use a single added mass measurement and instead used actual measured mass, but the manufacturers measured Mmd data. Unlike the smaller diameter WF152, the 10-V curves were sufficiently linear to provide a good curve fit, so I included them in the optimization.

	TSL model		LTD model		Factory
	sample 1	sample 2	sample 1	sample 2	
FS	37.7 Hz	37.0 Hz	37.9 Hz	36.9 Hz	40 Hz
REVC	6.11	6.20	6.11	6.20	6.3
Sd	0.0129	0.0129	0.0129	0.0129	0.0131
QMS	10.51	10.45	10.84	10.17	10.9
QES	0.33	0.35	0.35	0.35	0.35
QTS	0.32	0.34	0.34	0.34	0.34
VAS	25.8 ltr	26.9 ltr	25.8 ltr	27.2 ltr	24.4 ltr
SPL 2.83 V	88.1 dB	87.8 dB	87.9 dB	87.7 dB	88 dB
XMAX	5.5 mm	5.5 mm	5.5 mm	5.5 mm	5.5 mm

Table 1: Wavecor WF182BD04-01 midwoofer

Next, I post-processed the 10 550-point stepped sine

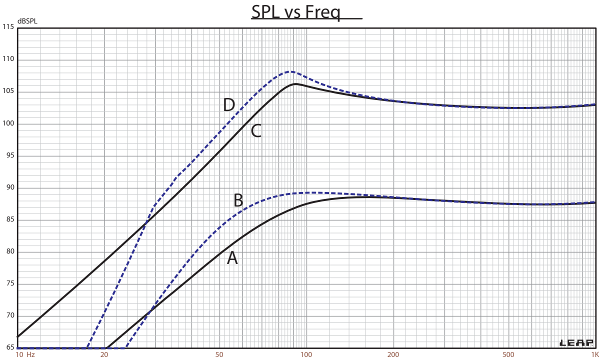


Figure 3: Wavecor WF182BD04-01 computer box simulations (A = sealed at 2.83 V; B = vented at 2.83 V; C = sealed at 24 V; D = vented at 24 V)

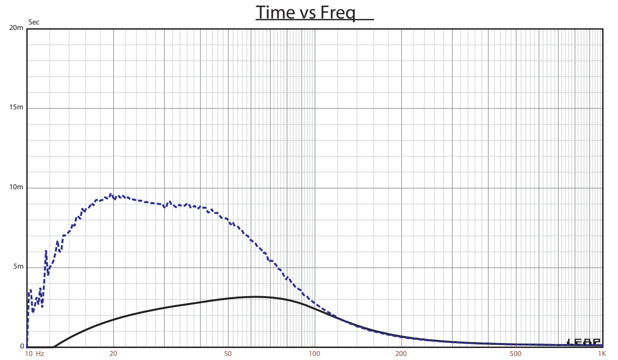


Figure 4: Group delay curves for the 2.83 V

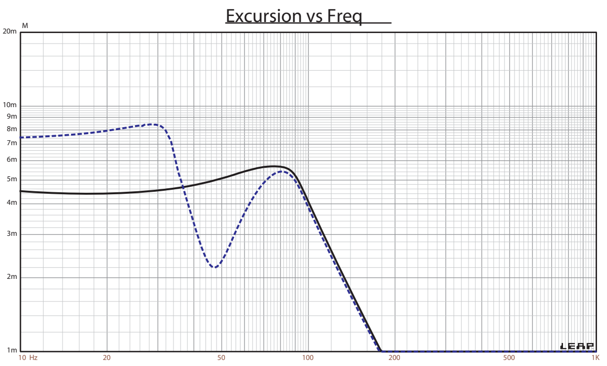


Figure 5: Cone excursion curves for the 24 V curves in Figure 3

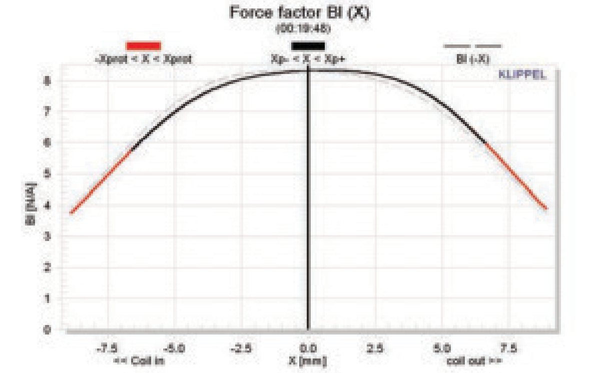


Figure 6: Klippel Analyzer Bl (X) curve for the Wavecor WF182BD04-01

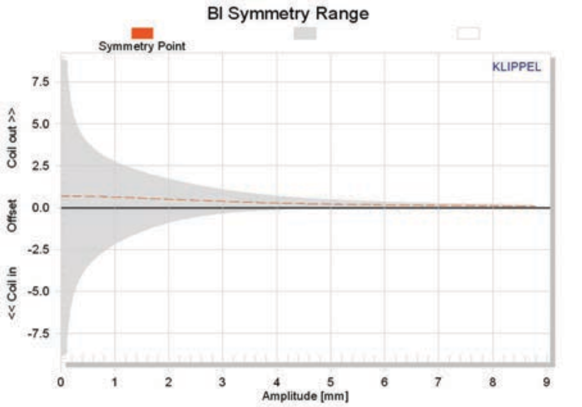


Figure 7: Klippel Analyzer Bl symmetry range curve

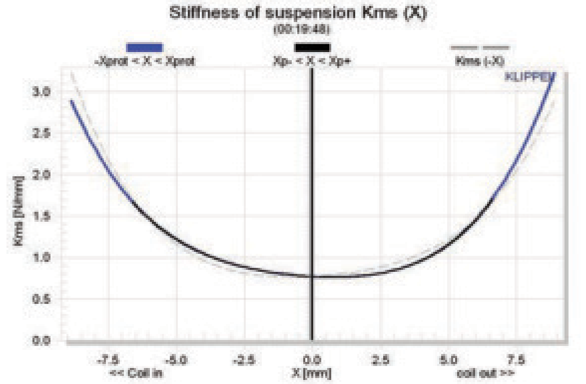


Figure 8: Klippel Analyzer mechanical stiffness of suspension Kms (X) curve for the Wavecor WF182BD04-01

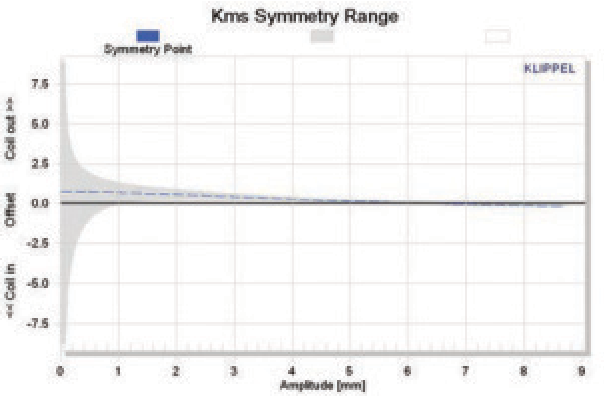


Figure 9: Klippel Analyzer Kms symmetry range curve

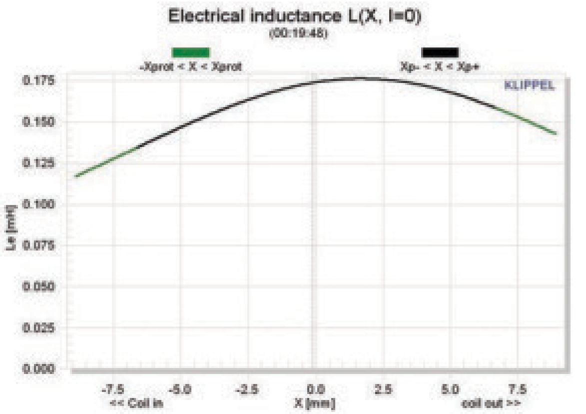


Figure 10: Klippel Analyzer Le(X) curve for the Wavecor WF182BD04-01

wave sweeps for each WF182 sample and divided the voltage curves by the current curves (admittance) to derive impedance curves, phase added by the LMS calculation method, and, along with the accompanying voltage curves, imported to the LEAP 5 Enclosure Shop software. Because most Thiele/Small data provided by OEM manufacturers is being produced using either a standard method or the LEAP 4 TSL model, I additionally produced a LEAP 4 TSL model using the 1-V free-air curves. I selected the complete data set, the multiple voltage impedance curves for the LTD model (see **Figure 2**) for the 1-V free-air impedance curve) and the 1-V impedance curve for the TSL model, in the transducer derivation menu in LEAP 5 and produced the parameters for

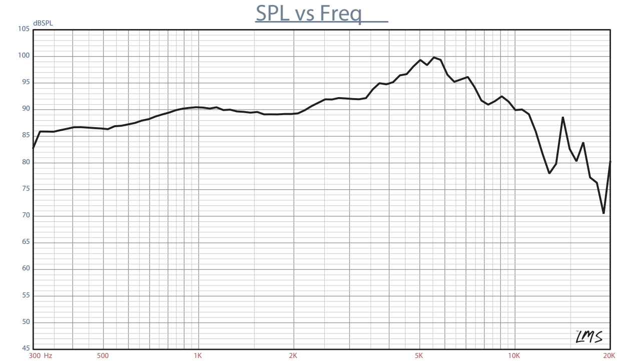


Figure 11: Wavecor WF182BD04-01 on-axis frequency response

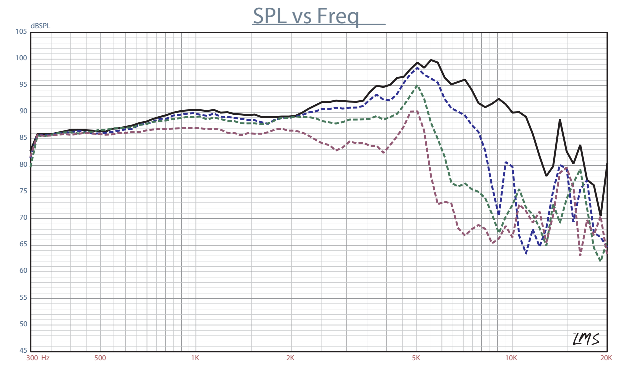


Figure 12: Wavecor WF182BD04-01 on- and off-axis frequency response

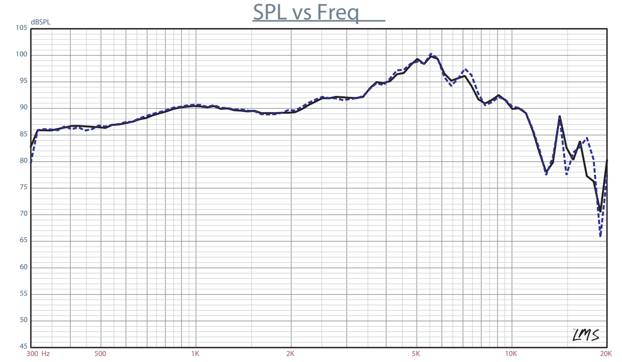


Figure 13: Wavecor WF182BD04-01 two-sample SPL comparison

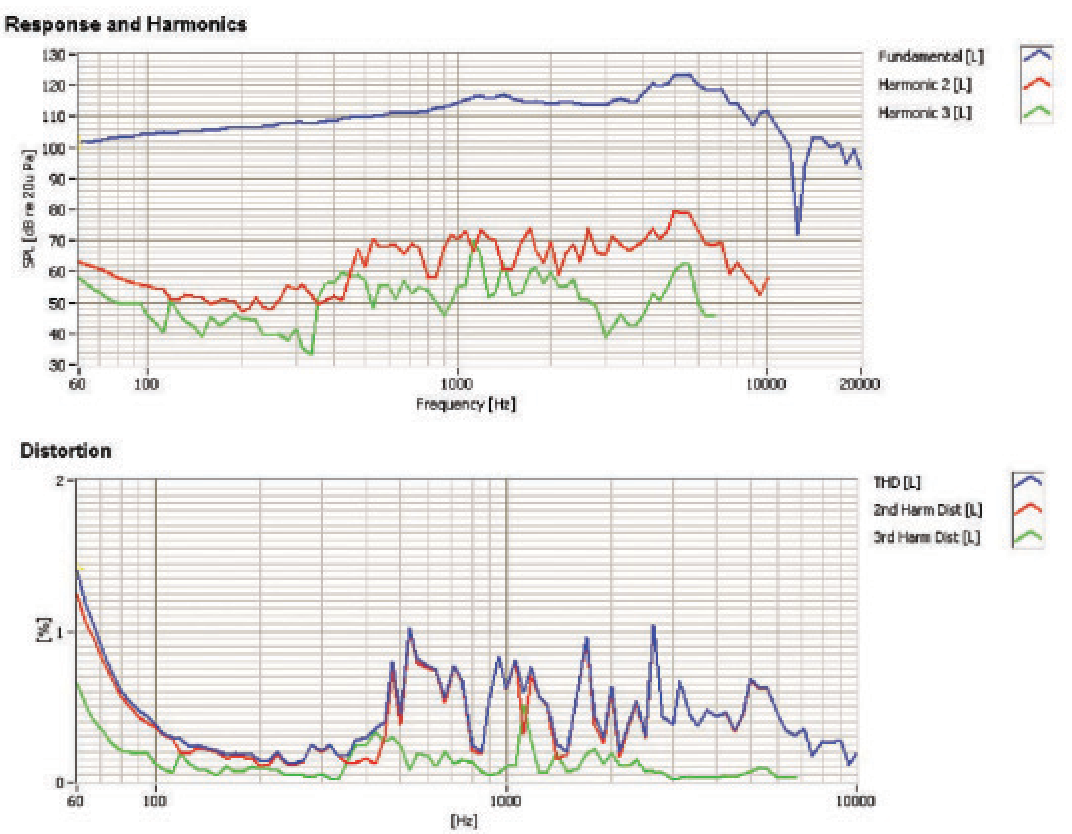


Figure 14: Wavecor WF182BD04-01 SoundCheck distortion plots

PA-HD8000P **MSRP: 5,999**

- 3 Panel LCoS Full HD 1080P resolution
- 16:9 and Cinema Wide 2.35:1 format
- Individual calibration of primary and secondary colors
- Bright - 5000 Lumens
- High contrast ratio - 15,000:1
- HDMI 1.3 connectivity

PA-HD9000-3LCD **MSRP: 5,999**

- 1-Chip HDTV High-Performance Resolution Ready
- Intellgent Core Chip 7"TFT LCD Panel w/Brillant-color
- Processing & Resolution: 1080i/p-720p-480p
- Screen Size: 40"-120" inch
- 16:9 and Cinema Wide 2.35:1 format
- HDMI 1.3 connectivity
- Bright - 5000 Lumens
- High contrast ratio - 15,000:1

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Pro Cinema Series

www.PalladiumHomeCinema.com

the computer box simulations. **Table 1** compares the LEAP 5 LTD and TSL data and factory parameters for both WF182BD04-01 samples.

LEAP parameter Qts calculation results for the WF182 were very close to the factory data, actually even more so than I usually see. Given that, I proceeded to produce two computer box simulations and programmed them into LEAP 5, one sealed and one vented. This resulted in a 0.24 ft^3 Butterworth sealed enclosure with 50% fiberglass fill material, and a 0.37 ft^3 vented QB3 alignment enclosure simulation with 15% fiberglass fill material and tuned to 44 Hz.

Figure 3 displays the results for the WF182BD04 in the sealed and vented boxes at 2.83 V and at a voltage level high enough to increase cone excursion to Xmax +15% (5.75 mm). This produced a F3 frequency of 77 Hz with a box/driver Qtc of 0.68 for the 0.24 ft^3 sealed enclosure and –3 dB = 44 Hz for the 0.37 ft^3 vented QB3 simulation. Increasing the voltage input to the simulations until the maximum linear cone excursion was reached resulted in 106 dB at 24 V for the sealed enclosure simulation and 108 dB with the same 24-V input level for the larger ported enclosure (see **Figures 4** and **5** for the 2.83-V group delay curves and the 24-V excursion curves). Note that excursion begins increasing below 35 Hz, so a high-pass filter below this frequency would greatly increase the power handling, which is the case for most vented designs.

Klippel analysis for the Wavecor 7" woofer (our analyzer is provided courtesy of Klippel GmbH), performed by Pat Turnmire, Red Rock Acoustics (author of the SpeaD and RevSpeaD software) produced the Bl(X), Kms(X) and Bl and Kms symmetry range plots given in **Figures 6** to **9**. The Bl(X) curve for the WF182 (see **Figure 6**) is relatively broad and nicely symmetrical. The Bl symmetry plot curve (see **Figure 7**) shows a trivial 0.69-mm coil forward offset at the rest position that decreases to 0.2 mm at the physical 5.5 mm Xmax of the driver, so this driver looks very well balanced. **Figures 8** and **9** show the Kms(X) and Kms symmetry range curves for the Wavecor midbass woofer. The Kms(X) curve is also very symmetrical, and has a very minor forward (coil-out) offset of 0.76 mm at the rest position that decreases to 0.17 mm at the 5.5 mm physical Xmax location on the graph, which is likewise trivial. Displacement limiting numbers calculated by the Klippel analyzer for the WF182 were XBl at 82% Bl = 5.5 mm, and for XC at 75% Cms minimum was 3.9 mm, which means that the compliance is the most limiting factor for prescribed distortion level of 10%.

Figure 10 displays the inductance curve Le(X) for the WF182BD04-01. Inductance will typically increase in the rear direction from the zero rest position as the voice coil covers more pole area. However, the WF182 inductance shows only minor inductive variation as the coil moves in both directions due to the dual shorting ring configuration. The inductance variation is only a maximum of 0.03 mH from the rest position to the full in and out Xmax positions, which is very good.

Next I mounted the WF182 woofer in an enclosure which had an 18" × 8" baffle and was filled with damping material (foam) and then measured the DUT on- and off-axis from 300 Hz to 40 kHz frequency response at 2.83 V/1 m using the LinearX LMS analyzer set to a 100-point gated sine wave sweep. **Figure 11** gives the WF182's on-axis response indicating a very smoothly rising response to about 5.5 kHz, with no peaking at the high-pass rolloff. **Figure 12** displays the on- and off-axis frequency response at 0°, 15°, 30°, and 45°. The –3dB at 30° off-axis, with respect to the on-axis curve, occurs at 3.0 kHz, so a crossover in the 3 kHz region would be appropriate. And, finally, **Figure 13** gives the two-sample SPL comparisons for the 7" Wavecor midbass, showing a close match to less than 0.5 dB throughout the operating range.

For the remaining battery of tests, I employed the Listen, Inc. SoundCheck analyzer, ¼" SCM microphone, and power supply (courtesy of Listen, Inc.) to measure distortion and generate time frequency plots. For the distortion measurement, I mounted the Wavecor woofer rigidly in free-air and set the SPL to 94 dB at 1 m (4.3 V) using a noise stimulus, and then measured the distortion with the Listen, Inc. microphone placed 10 cm from the dust cap. This produced the distortion curves shown in **Figure 14**. I then used SoundCheck to get a 2.83 V/1 m impulse response for this driver and imported the data into Listen, Inc.'s SoundMap Time/Frequency software. The resulting CSD waterfall plot is given in **Figure 15** and the Wigner-Ville (for its better low-frequency performance) plot in **Figure 16**. For more on this and other well-crafted transducers from Wavecor, visit www.wavecor.com.

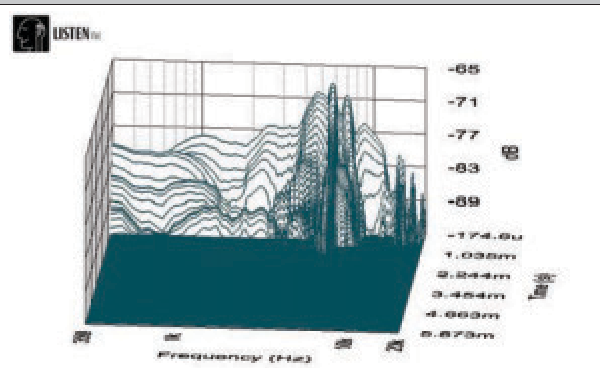


Figure 15: Wavecor WF182BD04-01 SoundCheck CSD waterfall plot

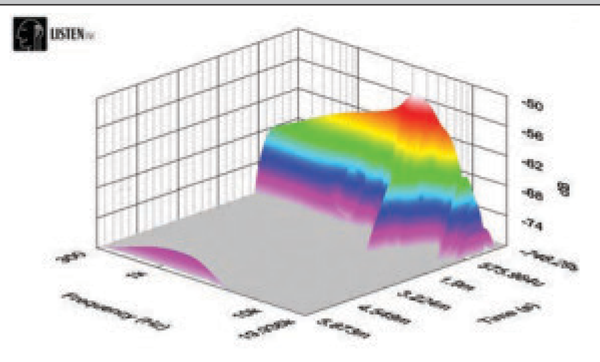


Figure 16: Wavecor WF182BD04-01 SoundCheck Wigner-Ville plot

SCAN-SPEAK 12W/4524G00

Next, I began collecting data on the Scan-Speak 4" diameter 12W/4524G00 (see **Photo 2**), the new mid-woofer addition to the price-sensitive Scan-Speak Discovery line. Small diameter midwoofers used in mini bookshelf speakers have been an important product with Scan-Speak. The 12W is built on a six-spoke cast aluminum frame that has three 25 mm × 6 mm "windows" for enhanced voice coil cooling. Powering this 4" device is a conventional 15 mm thick 72 mm diameter ferrite magnet sandwiched between the front and rear plates. Features include a fiberglass slightly curvilinear cone, fiberglass 1"-diameter dust cap, NBR rubber surround, 2.5"-diameter black flat cloth spider, 1" (25 mm) diam-

Photo 2: Scan-Speak 12W/4524G00

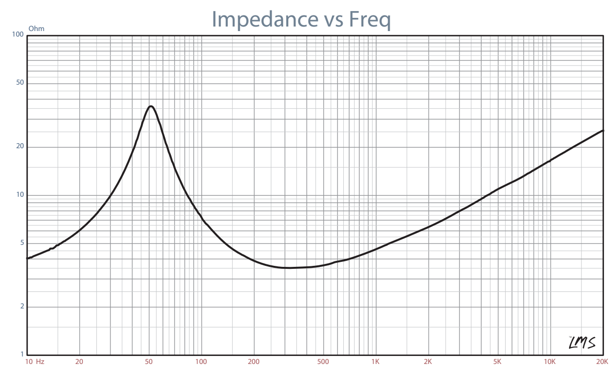


Photo 17: Scan-Speak 12W/4524G00 free-air impedance plot

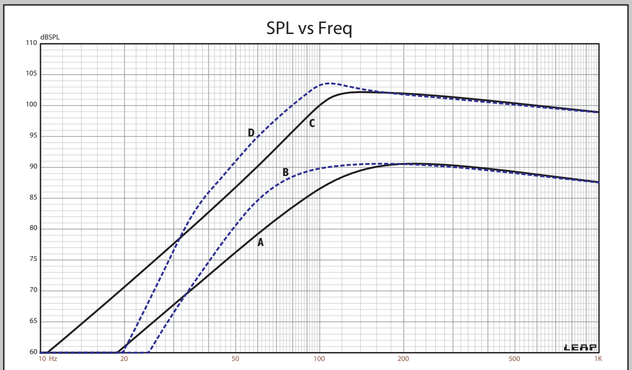


Figure 18: Scan-Speak 12W/4524G00 computer box simulations (A = sealed at 2.83 V; B = vented at 2.83 V; C = sealed at 12 V; D = vented at 12 V).

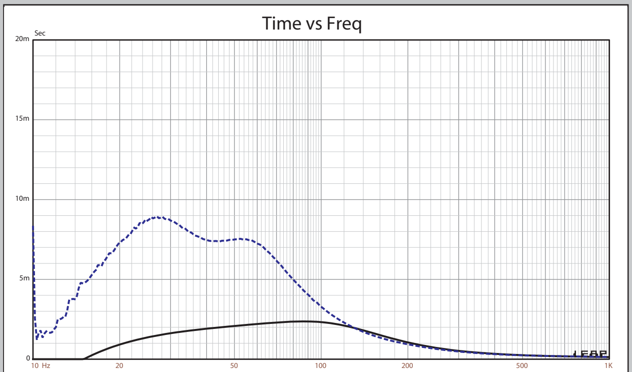


Figure 19: Group delay curves for the 2.83 V curves in Figure 18

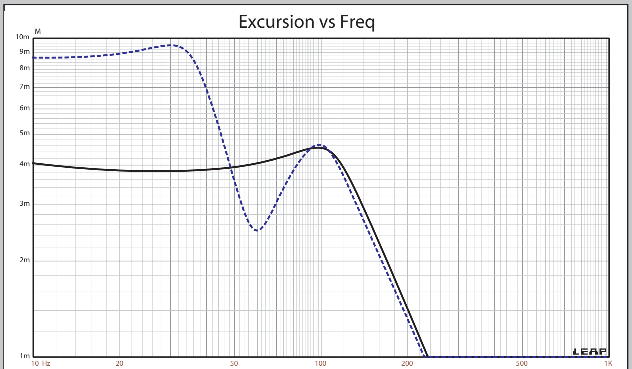


Figure 20: Cone excursion curves for the 12V curbes in Figure 18

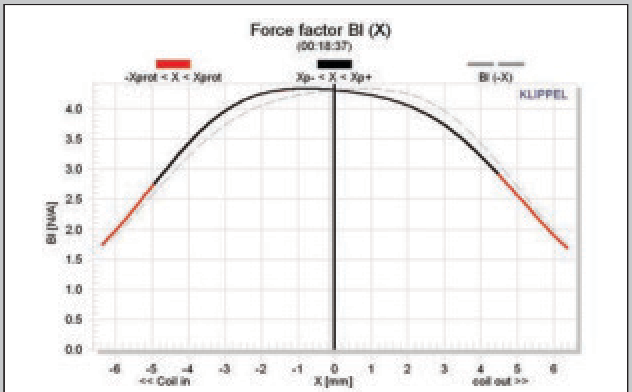


Figure 21: Klippel Analyzer Bl (X) curve for the 12W/4524G00

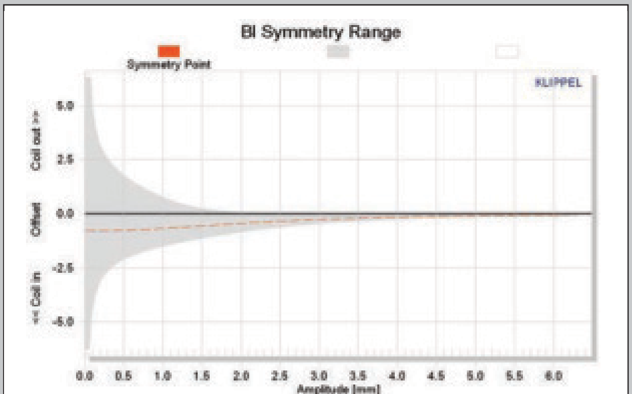


Figure 22: Klippel Analyzer Bl symmetry range curve

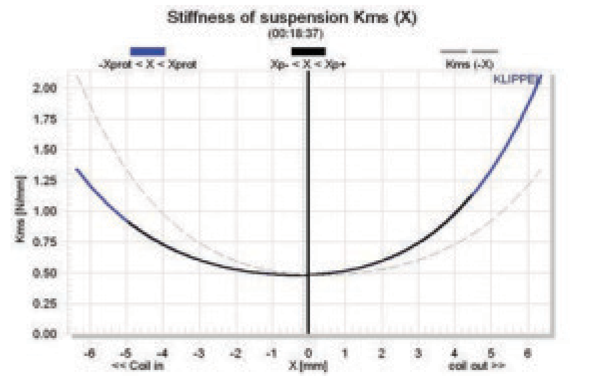


Figure 23: Klippel Analyzer mechanical stiffness of suspension Kms (X) curve for the Scan-Speak 12W/4524G00

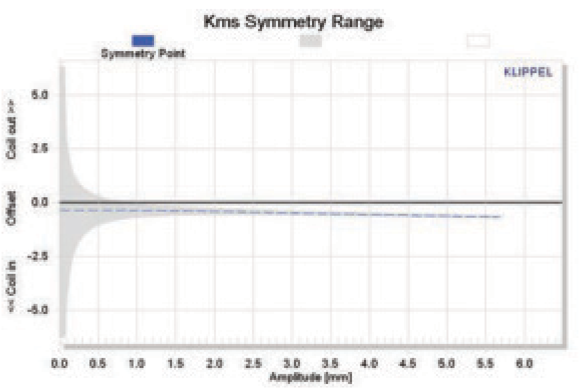


Figure 24: Klippel Analyzer Kms symmetry range curve

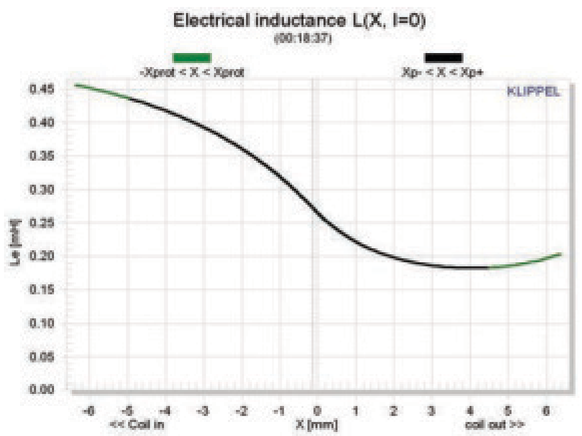


Figure 25: Klippel Analyzer L(X) curve for the Scan-Speak 12W/4524G00

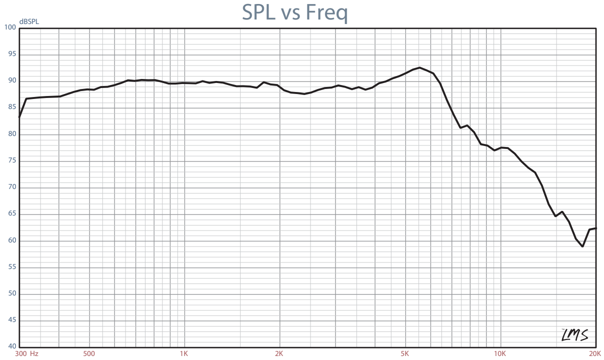


Figure 26: Scan-Speak 12W/4524G00 on-axis frequency response

eter voice coil (aluminum former wound with round copper wire), and gold-plated terminals.

Testing commenced with the driver clamped to a rigid test fixture in free-air and voltage and current sweeps taken at 0.3 V, 1 V, 3 V, and 6 V. Because this is a small diameter driver with only 3-mm Xmax, the 6-V data was too nonlinear for LEAP 5 to curve-fit, so I did not include it. I post-processed the six 550-point stepped sine wave sweeps for each 12W midwoofer sample and divided the voltage curves by the current curves (admittance) to produce impedance curves, phase added using LMS calculation method, and, along with the accompanying voltage curves, uploaded to the LEAP 5 Enclosure Shop software.

In addition to the LEAP 5 LTD model results, I also produced a LEAP 4 TSL model set of parameters using just the 1-V free-air curves. I selected the final data set, which includes the multiple voltage impedance curves for the LTD model (see **Figure 17** for the 1-V free-air impedance curve) and the 1-V impedance curve for the TSL model, and produced the parameters in order to perform the computer box simulations. **Table 2** compares the LEAP 5 LTD and TSL data and factory parameters for both Scan-Speak 4" samples.

LEAP parameter calculation results for the 12W midwoofer were close to the factory data, although the LEAP calculated sensitivity was about 2 dB lower. Given this, I proceeded to set up computer enclosure simulations using the LEAP LTD parameters for Sample 1. I programmed in two enclosures, one sealed and the other vented. For the first closed box Butterworth simulation I used a 120 in^3 enclosure with 50% fiberglass fill material, and for the second vented box, a larger volume of 251 in^3 QB3 with 15% fiberglass fill material tuned to 55.4 Hz.

Figure 18 displays the results for the 12W/4524G00 in the sealed and vented boxes at 2.83 V and at a voltage level high enough to increase cone excursion to Xmax +15% (4.5 mm). This resulted in a F^3 = 109 Hz with a box/driver Qtc of 0.69 for the 120 in^3 closed box design and a –3 dB = 72 Hz for the 251 in^3 vented simulation. Increasing the voltage input to the simulations until the maximum linear cone excursion was reached generated 102 dB at 12 V for the sealed enclosure simulation and 103.5 dB with same 12-V input level for the larger ported enclosure (see **Figures 19** and **20** for the 2.83-V group delay curves and the 12-V excursion curves). Very reasonable performance for a 4" woofer.

Klippel analysis for the 12-W midwoofer produced the Bl(X), Kms(X), and Bl and Kms symmetry range plots given in **Figures 21** to **24**. The Bl(X) curve (see **Figure 21**) is moderately broad and symmetrical, with a coil-in (rearward) offset. In the Bl symmetry range curve in **Figure 22**, there is a 0.8-mm, coil-in (rearward) offset that goes to 0.3 mm at the physical Xmax position (3 mm), so not too bad.

Figures 23 and **24** give the Kms(X) and Kms symmetry range curves for the 4" midwoofer. The Kms(X) curve is even more symmetrical. **Figure 24**, the Kms symmetry range plot, shows a 0.35-mm, coil-in offset at the rest position that increases somewhat to 0.5 mm at the physical Xmax of the driver. Displacement limiting numbers calculated by the Klippel analyzer for the midrange were XBl at 82% Bl = 3.4 mm and for XC at 75% Cms minimum was 2.4 mm, which means that for this 4" midwoofer, the suspension offset is the most limiting factor for prescribed distortion level of 10%.

Figure 25 gives the inductance curves L(X) for the 12W/4524G00, which shows a typical situation where the inductance increases and the voice coil travels inward covering more of the pole piece. Inductance swing from Xmax forward to Xmax rearward is about 0.21 mH inductance. Having Scan-Speak add a copper cap to the pole will decrease the inductive swing, plus you could incorporate the copper cap along with a non-conducting former and get some reasonable subjective improvement, but, of course, you just increased the cost of a cost-effective product!

With the Klippel testing finalized, I mounted the 12W midwoofer in an enclosure which had a 15"× 5" baffle and filled with foam damping material and proceeded to measure the driver frequency response both on- and off-axis from 300 Hz to 40 kHz at 2.83 V/1 m using a 100-point gated sine wave sweep. **Figure 26** depicts the on-axis response resulting in a very flat rising

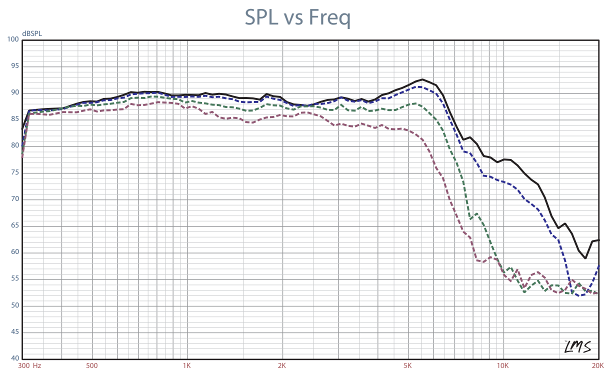


Figure 27: Scan-Speak 12W/4524G00 on- and off-axis frequency response

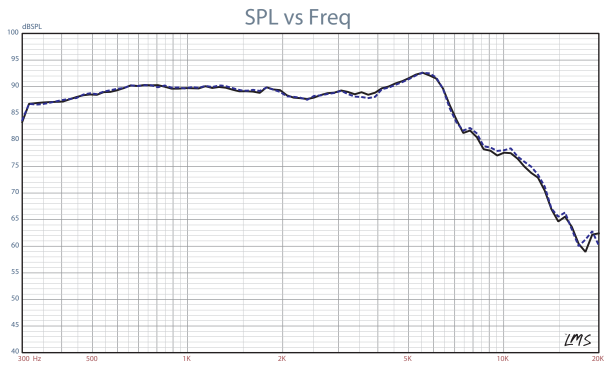


Figure 28: Scan-Speak 12W/4524G00 two-sample SPL comparison

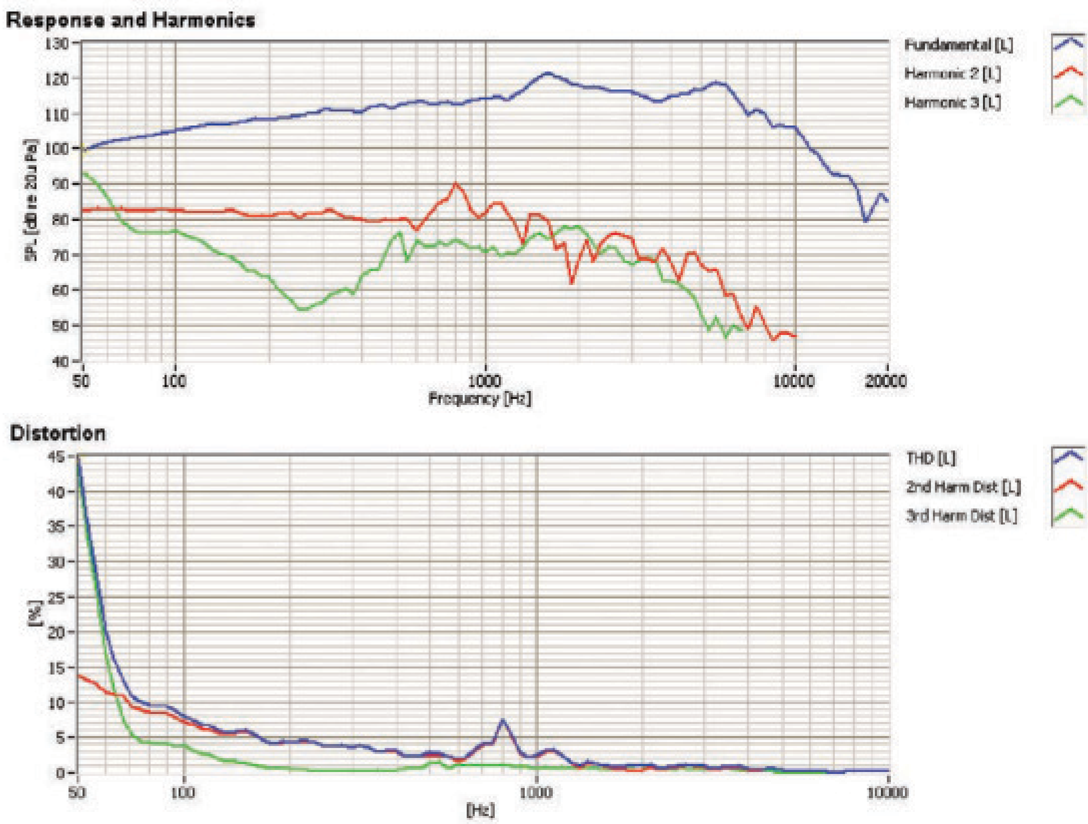


Figure 29: Scan-Speak 12W/4524G00 distortion plots

Affinity TECHNOLOGY

HIGH DEFINITION CINEMA

Sound Through Research

AN-501

5.1 Speaker System

1000 WATTS TOTAL POWER

• 500 Watt 10"Subwoofer + 10" Passive Radiator
• Hard Wood Cabinet Design for Accurate Low Bass Response
• Surround Speakers : Each Fetures (2) 2.5" H.D Full Range
• Drivers, 180 degree swivel cabinets.
• Center Channel : (2) 2.5" H.D Drivers

MSRP: $1995.00

AN-601

5.1 Speaker System

1000 watt dynamic peak power

• Low resonance mdf constructed subwoofer
• ABS 180 degree multi directional cube-surrounds
• 5 acoustic matched surrounds-center channel
• (2) 6.5 inch metallic finished passive front firing subwoofers
• (10) 2.5 inch shielded poly-coated midbass driver
• Frequency response: 60hz- 20khz • SPL 103db
• Nominal impedance 8ohm • Sensitivity 93db
• Easy hook up TV projector/ dvd/ blue ray

MSRP: $1999.00

AN-701

5.1 Speaker System

• 1500 watt dynamic peak power
• Pro studio subwoofer- purcussion design
• 4 acoustic matched surrounds
• ABS 180 degree multi directional cube-surrounds
• Full size shielded center channel
• (1) 8 inch long throw passive side firing subwoofer
• (10) 2.5 inch shielded poly-coated midbass driver
• Gold binding post • Easy hook up TV projector/ dvd/ blue ray

MSRP: $2495.00

*Model number and manufacturer's suggested prices are for identification purpose only. Manufacturer's suggested price do not represent a bona fide selling price in the metro trading area. Manufacturer reserve the right to make changes on specification and material.

Affinity TECHNOLOGY

www.AffinityHomeCinemas.com

response that is ±1.68 dB from 300 Hz to 4.5 kHz with a small peak just before the low-pass rolloff. **Figure 27** has the on- and off-axis frequency response at 0°, 15°, 30°, and 45°. At –3 dB and 30° with respect to the on-axis curve occurs at 4.5 kHz, so a 3 to 4.5 kHz crossover frequency would be appropriate for this Scan-Speak small woofer. And finally, **Figure 28** gives the two-sample SPL comparisons for the 4" 12W driver, showing a good match within the operating range.

For the last body of testing on the Scan-Speak 4" midwoofer, I again fired up the SoundCheck analyzer and SCM microphone and power supply to measure distortion and generate time frequency plots. Setting up for the distortion measurement again consisted of mounting the woofer rigidly in free-air, and the SPL set to 94 dB at 1 m (5.8 V) using a noise stimulus (SoundCheck has a software generator and SPL meter as two of its utilities), and then the distortion measured with the SCM microphone placed 10 cm from the dust cap. This produced the distortion curves shown in **Figure 29**.

For the last test on the 12W, I used the SoundCheck analyzer to get a 2.83 V/1m impulse response for this driver and imported the data into Listen, Inc.'s SoundMap Time/Frequency software. The resulting CSD waterfall plot is given in **Figure 30** and the Wigner-Ville (for its better low-frequency performance) plot in **Figure 31**. For more on this midrange and all the other great Scan-Speak drivers, visit www.scan-speak.dk. *aX*

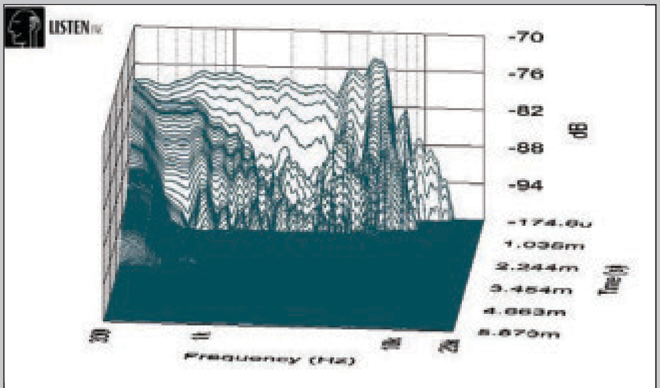


Figure 30: Scan-Speak 12W/4524G00 SoundCheck CSD waterfall plot

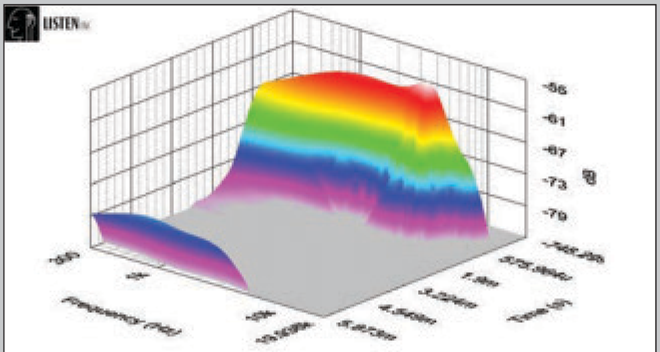


Figure 31: Scan-Speak 12W/4524G00 SoundCheck Wigner-Ville plot

By Robert Nance Dee (United States)

The Willow Pre-amp

A high slew-rate JFET amplifier

The curious thing about specifications is that there are none for musicality. I love the single-ended tube builders who stick to their guns about the musicality of Class A amps that run into single-digit distortion. Musicality must be paramount!

The first specification I always place a few notches down on the criteria list is distortion. Quite frankly, no one on the planet can hear the difference between 0.01 and 0.005 distortion. In fact, once you get under about 0.5%, if that, distortion is a non sequitur. What really counts to my ears is tracking, which I pay close attention to when I begin the design process. I always go for high slew rates because they bring clarity and definition to amps. My hat goes off to designers such as MOON who list slew rates in their specs.

Willow is just that type of project (see **Photo 1**). The PMBF4393 JFET has a rise time of 5 V/ns (200 V/µs), and the people who have listened to this amp can hear the difference. I built it as a companion to my 5002 power amp, both using the HA5002 buffer.

WHAT'S WRONG WITH OP-AMPS, ANYWAY?

Op-amps are slow (see **Figure 1**), most run 5 or 6 V/µs. Even the highly touted OPA627 (blue trace) at 40 V/µs and 50 times the cost of the 4393 can't beat it (see **Figure 2**).

The other thing about JFETs is their character. They clip like tubes, and I love tube distortion. Let's face it, we all love distortion. We're just picky about what kind!

DESIGN PHILOSOPHY

My goals with this amp were musicality, definition, and clarity. I wanted an amp that could give equal justice to large orchestrations, solo guitars, and the golden voices that fill a good part of my listening time. I played with several factors and pushed limits to see whether my "what if" theoretical ideas would make a difference empirically. They did, at least to these old ears, and after many SPICE hours, prototypes, and tweaks, this amp emerged. In fact, please look at this amp as modules from the power supply, input/output, gain stage to the anti-pop circuitry. You can build each independently or modify them to fit your specific interests.

I designed the power supply (see **Figure 3**) for a single toroidal transformer with a dual 18-V secondary. I isolated the power to each 38-V JFET

Photo 1: The Willow pre-amp

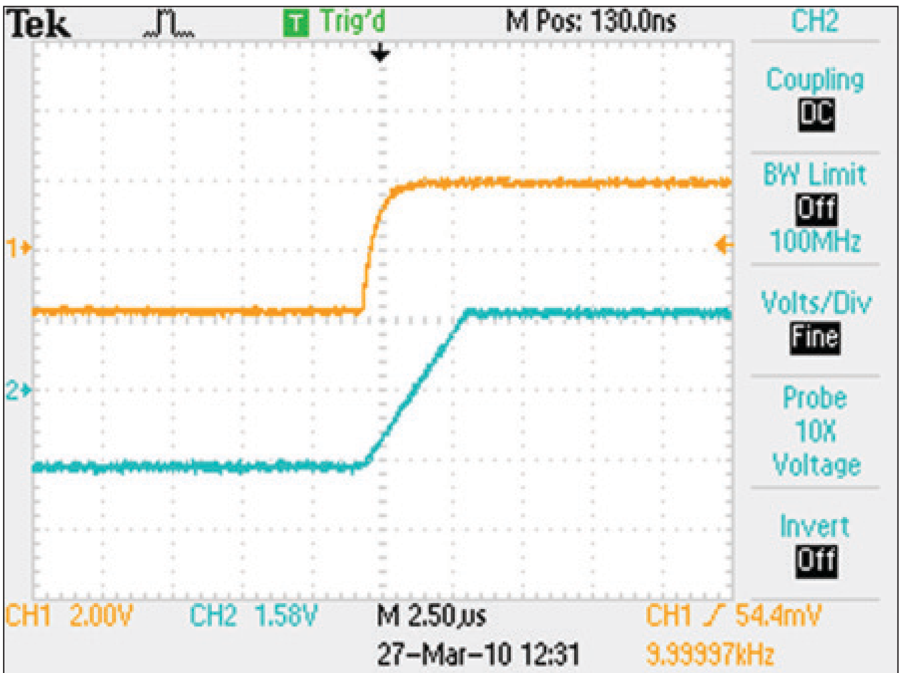


Figure 1: The PMBF4393 vs. the OPA6627

using one secondary for both, doubling it and tapping off the 18-V point for the positive side of the HA5002 buffers that run ±18 V (IC2, IC4).

The input/output board (see **Figure 4**) is designed for zero open contacts and to be highly ergonomic. This is no small design requisite. Ergonomics play an important part of my designs because I'm always turning pots and changing input channels. This amp sits at my work desk, and I flip between my CD player, FM tuner, and live stream—constantly!

What's special about this input/output is the optical encoder used to change from one input to the next. No contacts brush each other in the encoder, and its mean time between failures (MTBF) is extremely high. You can spin the dial all day and it won't wear out. It will never be noisy and never need to be cleaned. It runs four relays—one for each input—and one relay covers both the left and right channel. The relays are hermetically sealed, run on only 6 mA, have a MTBF in the millions, and cost under $4 each.

Another thing I don't like is printing on faceplates. Call me finicky, but lettering wears out over time and it's hard to read, especially in low light. Instead, I employed four different color LEDs, one for each input; changing from one input to the next lights a specific LED that sits behind the input knob. I used a black Lexan face on the amp and inserted a clear Lexan disk behind it. As I

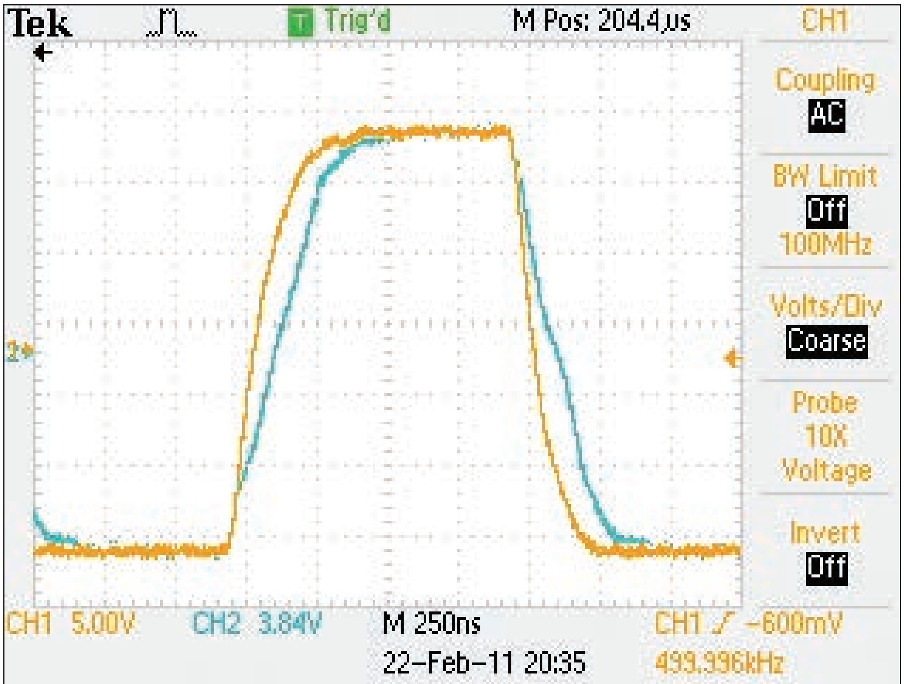


Figure 2: Top trace Agilent 33220A, bottom trace OPA227

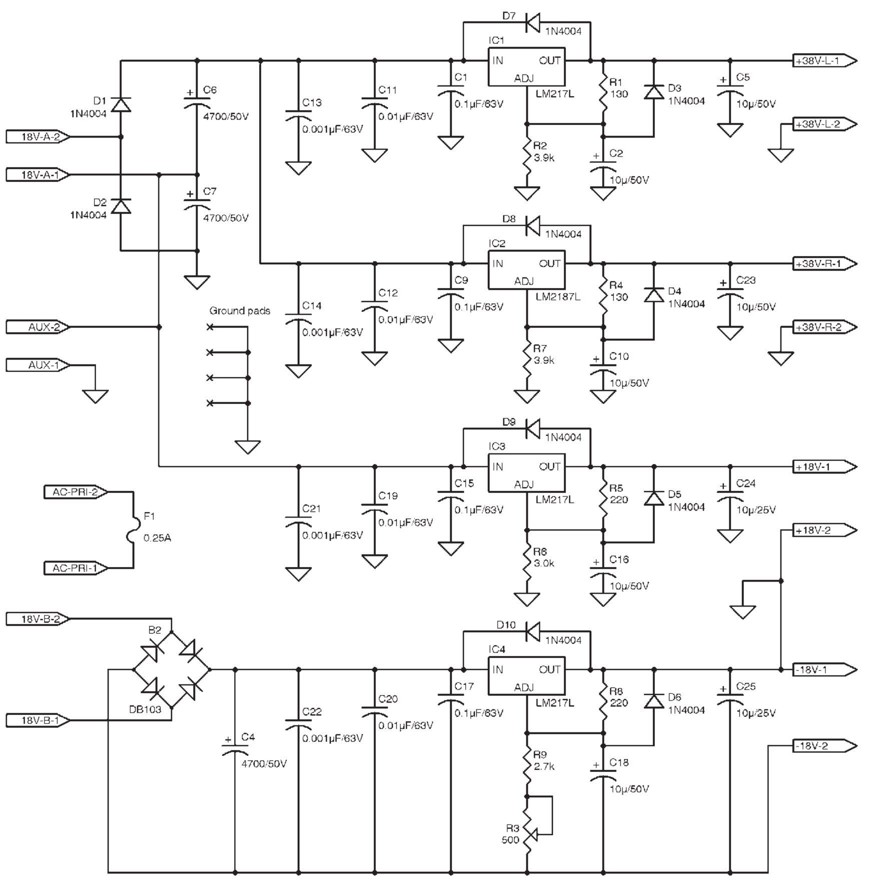


Figure 3: Power supply

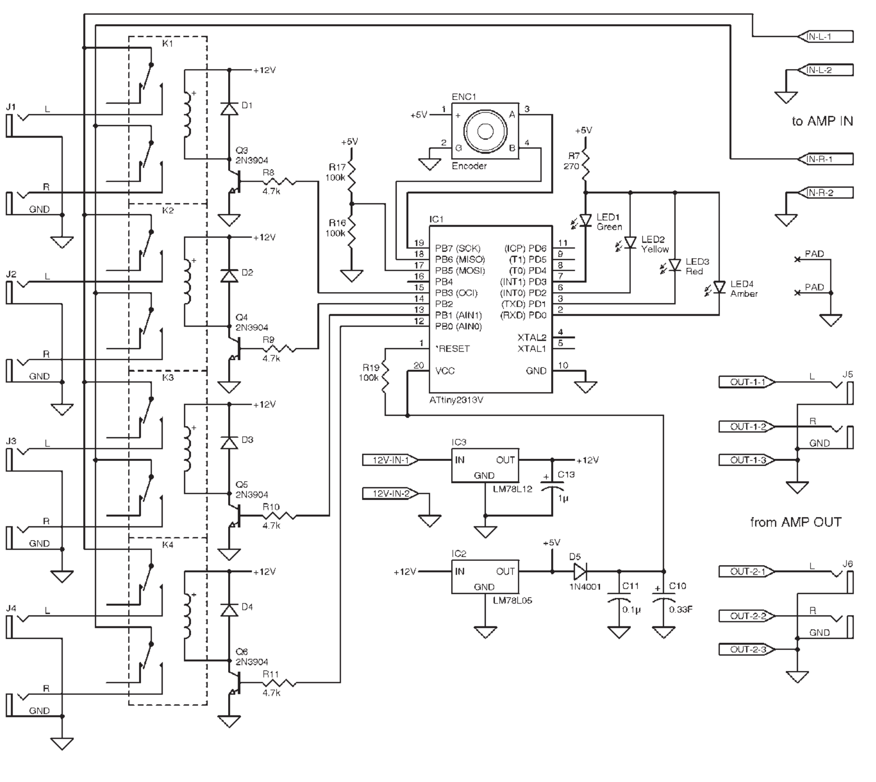


Figure 4: In-out

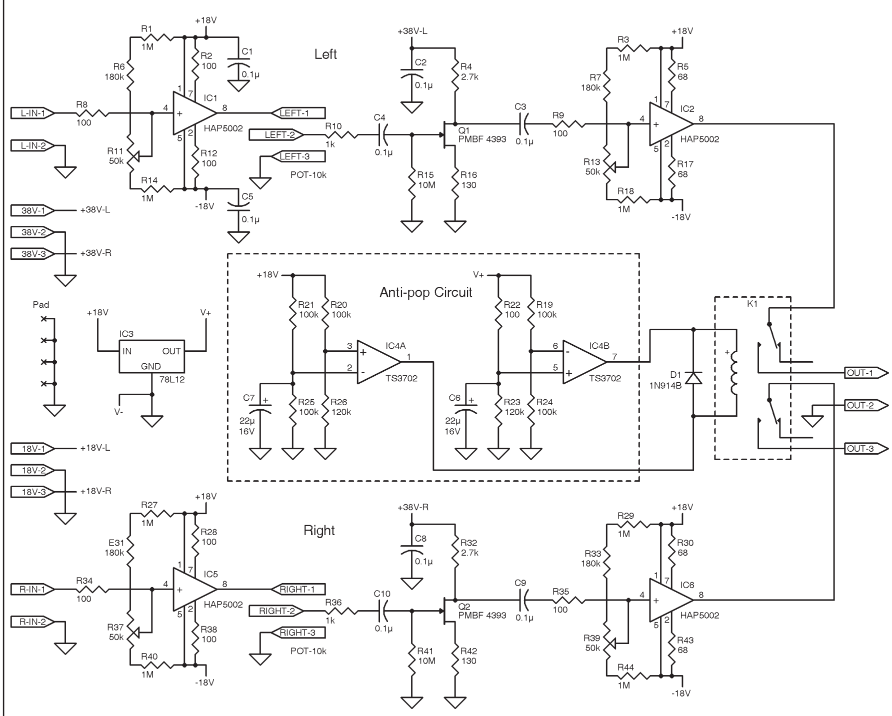


Figure 5: Gain stage

change from one input to the next, the white knob and clear Lexan background change color. Not only is it neat, but I can change which input I'm listening to in a completely dark room.

The gain board (see **Figure 5**) uses a PMBF4393 JFET running Class A and two HA5002 buffers from Burr-Brown/Intersil. I'd like to talk a little about buffers and how I use them in amps.

BUFFER BENEFITS

A good buffer such as the 5002 adds little, if any, coloration to the music. It's basically a transconductance device that gives great latitude and simplicity in connecting the JFET or tube. Yes, that's correct, I said tube because I use buffers to connect tube stages. You can also use the buffer to bias a tube or FET simply by biasing it just as I have done in this circuit. Here I use zero bias, but it is an easy task to turn a bias pot and in effect change a tube's bias.

Another benefit of buffers is their ability to place very high impedances on inputs and very low impedances on outputs. I don't like transformers in my circuits (to each his own), and buffers mean I don't have to use them. Capacitors are another story. They are a necessary evil but, like most of you, I have specific ideas about how I use them in circuits. I like to keep their values as low as possible. You will notice that the capacitors in this project are 0.1 µF.

Say you have an input impedance to your buffer that is 470 k, which is close to the actual value in this circuit. The bandwidth lead is:

$$f = \frac{1}{2\pi RC};$$

$$f = \frac{1}{6.28 \times 470 \times 10^{3} \times 0.1 \times 10^{-6}}$$

$$= 3.4 \text{ Hz}$$

You could have used a 0.47 µF capacitor with an input impedance of 100 k, which would give the same lead frequency, but I specifically wanted to keep the input capacitances low just to see how, if at all, it would affect the sound. The 0.47 µF capacitor 100-k resistor input has another benefit in that it reduces input noise, but this amp is very quiet and I especially like the sound so, for now, the 0.1 µF-capacitor stays.

If you have long input runs, you might want to consider using the 0.47 μF/100 kΩ option, but even with long runs I didn't have a problem. This is certainly an option, and I suggest you find your own answers. You will also notice that I didn't use an input capacitor before the first op-amp; you can if you like, especially if you have problem outputs from your external components. Remember that any bias on the input of the 5002 reflects on the output. The output is 3 Ω, so it is very convincing.

You cannot, however, eliminate the first 5002 without increasing the value of the volume pot. Here's another area where I like to keep things low. The pot for this amp is 10 kΩ—much too low for even a 0.47-μF input capacitor.

Lag (output frequency falloff) also benefits from the buffers. What I have done is place very little load on the JFET. It's just in the circuit for amplification; the buffers do all the rest of the work, and nicely. With 1300 V/μs rise time the 5002 slows nothing down in this high tracking amp.

One more thing: don't be tempted to use a through-hole PN4393 in place of the SMD PMBF4393. It tops out at 30 V and not 40 like the latter.

The anti-pop circuit is designed to catch any pops when the amp comes on (IC4a) as well as when it's turned off. It senses the slow drop of charged capacitors when shutting down and shuts the relay down quickly. It also gives the circuit a few seconds delay when powering up. It acts as an "and" gate because both conditions must be true—IC4a must be low and IC4b must be high for an output signal. One half the comparator controls turn on and the other turn off.

CONSTRUCTION

Boy, did I goof! If you look at my layout you'll see that I thought I could get away with the power transformer close to the digital pot I employed here. I couldn't, and I paid the price for noise—not bad, but why have it if you can avoid it? Actually, I like dedicated cases for my power supplies and power circuits so you might want to explore that avenue.

As I previously stated, think of this as a module project: you can use your own input stage volume stage (as I did here) or even power supply. As for the amp board, I built it before I did any drastic modding, but this is an open amp and I'd love to hear about everyone's ideas and tweaks!

Adjust all the pots for a null (zero volts) on pin 8 of all the 5002 buffers. This is an easy task, so do it a couple of times. Bias isn't that critical—as long as you're within ±50 mV things will be fine, although I easily got my circuits down to only a few millivolts.

Before inserting the ICs in their sockets (if you use sockets), check pins 1 and 7 for +18 V and pins 2 and 5 for –18 V. While the price of the 5002 is not prohibitive, there is no reason to blow chips!

I did add a headphone socket. This is your option, as are the screw terminals. They make it handy to troubleshoot prototypes, but they do encourage noise later on down the road.

The circuit boards (see **Photo 2**) are straightforward, and while I did use SMDs on the input board, I tried to avoid them on the gain board by utilizing as many through-hole devices as possible.

Finally, give this amp some time to break in—at least 50 to 100 hours. I have included a subwoofer output. My favorite configuration is a two-way speaker with a sub.

I favor building nice cases—do it right. Nothing looks worse than a great amp in a tin box. It's like washing your car—it always seems to run better after it's washed!

For more information on this project, visit www.dsgnspec.com. *aX*

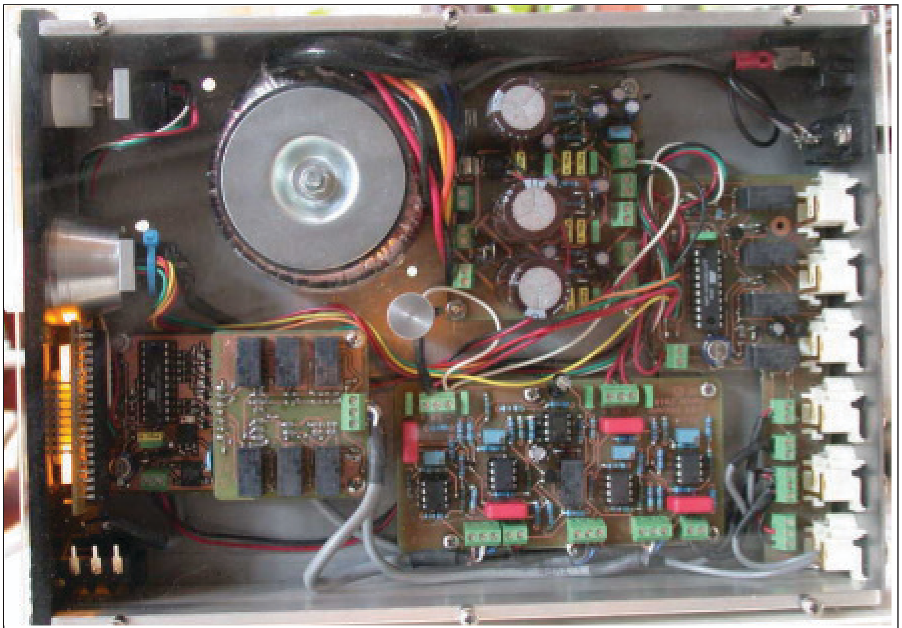


Photo 2: Circuit view

SPECIFICATIONS

• Gain	21.58 (12/1)
• Bandwidth	Less than 4 Hz to greater than 500 kHz
• Distortion & noise	Less than 0.2% (200 mV_{PP} input)
• Input impedance	Greater than 500 kΩ
• Output impedance	~3 Ω
• Inputs	4 (optically driven sealed relays)
Test instruments:	
• Agilent 33220A	AWG generator
• HP 334A	Distortion analyzer
• Agilent 34410A	Digital meter
• Tektronix TDS2012B	Digital scope
• HP6632B	Power supply
• Array 3710A	DC load

By Jan Szymanski (Australia)

Digital Audio Player

Build an embedded hardware/software system for 16-bit digital audio.

If you work on electronic designs, then you've probably had to make a decision about dividing your project between hardware and software. The duality of hardware and software has some similarities to dualities in physics such as mass/energy and wave/particles.

Back in 2003, I built myself a simple MP3 player ("Build an MP3 Player," *Circuit Cellar* 152, March 2003). The main part of the project was a specialized chip for performing MP3 decoding. Since then, significant progress has been made in audio and chip technology. Thus, it's possible to build a similar media player with more simple hardware and more powerful software. A great example of such a system is the digital audio player I'll describe in this article. The system is an embedded hardware/software solution for 16-bit digital audio.

PROGRESS IN DIGITAL AUDIO

During the past few years, a great deal of progress has been made in the field of digital audio system technology. Let's focus on the three main areas of development.

First, consider the quality of embedded ARM-based microcontrollers. These devices have enough processing power for the software decoding of MP3 and other compressed file formats in real time.

Progress has also been made in mass storage device technology. The affordable, easy-to-interface SD card has become one of the most popular (a de facto standard) types of memory. The USB flash disk widely used by PC users is the other popular technology.

Finally, it's clear that a lot of progress has been made in audio codec and DAC technology. More variety, functionality, and lower power consumption are commonly available today. Sorry, Steve, but "my best software tool is (no longer) a soldering iron."

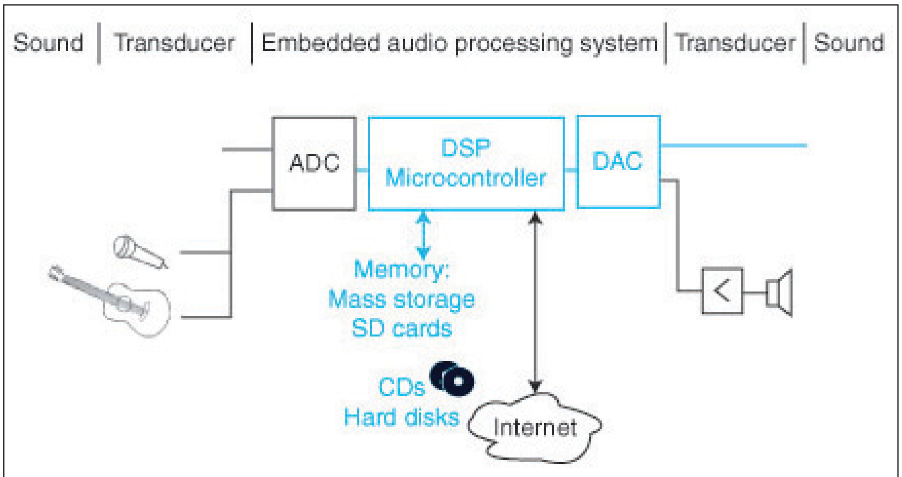


Figure 1: The input transducer (e.g., a microphone) converts sound from different sources into an analog signal. The ADC converts the analog signal into a digital signal. The DSP system with its resources does the processing, and it outputs the resulting signal into a DAC. To output sound, the DAC sends the analog signal into the speaker or headphones.

DIGITAL AUDIO PROCESSING

This is not a poor man's iPod. It's the real McCoy. My digital audio player is an example of digital audio processing in software.

There is often a need for audio in an embedded application, which can be controlled programmatically (by a command on a serial port that can make the system play a suitable tune). The block diagram for generic embedded audio processing is shown in **Figure 1**. The blocks used in the project are highlighted.

The user manual for the LPC214x family of NXP microcontrollers (section 1.3 Applications) doesn't list digital audio in its other applications, but you don't need to tell NXP Semiconductors what you're using it for. The SPI port on LPC214x microcontrollers doesn't support I^2S mode, but that's easy to fix.

DIGITAL MEDIA BASICS

Digital media technology is constantly evolving. To implement it, you need a good grasp of the terminology used in the field. There are several useful publications and web resources for broadening your knowledge of digital media technology. David Katz and Rick Gentile's *Embedded Media Processing* is one of my favorite resources. In this section, I'll briefly describe some basic terms.

Sound is a wave form of energy that propagates through air or some other medium. Amplitude and frequency are the two attributes that define sound. Music is an art form, which involves structured and audible sound. Although the definition is seemingly subjective, it's the most common. What's music to one person can be noise to another.

The audio signal can come from a transducer (e.g., a microphone) that converts sound waves into electrical signals, or it can be created (synthesized) electronically. On the other end, an analog audio signal coming to the speaker or an earphone is converted into a pressure change (sound wave).

To process sound, you need to convert both ways between analog and digital signals. An ADC and DAC do this. Devices combining one or more of each are called codecs. Note that the word codec is used for both the hardware devices and software algorithms.

To correctly represent the digital signal, refer to the Nyquist theorem. (The sampling frequency must be at least twice the highest frequency of the signal.) You can hear sounds between 20 and 20,000 Hz, so the sampling rate has to be at least 40 kHz. The commonly used sampling rate for CD quality audio is 44.1 kHz. The amplitude of sound is measured in decibels. The range for a human ear is from 0 (the threshold of hearing) to 120 dB SPL (the threshold of pain).

Dynamic range is a ratio of the maximum signal level to the minimum signal level (or the noise floor). For digital audio systems, the dynamic range depends on precision (number of bits representing the signal). For 16-bit codecs, the dynamic range is 96 dB.

There are standards for connecting physical devices in digital audio systems. The most common are I^2S and AC97. The former, which I used for my project, is a three-wire serial interface for the digital transmission of audio signals. It has a bit clock, data, and left/right synchronization lines. Intel created the popular AC97 standard for PC audio.

The popular audio file formats are wave and MP3. The latter is one of the most popular lossy audio compression codecs that can achieve a compression ratio of up to 12:1. Lossy codecs use a technique called perceptual encoding, which takes advantage of your ears' physiology. Although the data is lost, the decoded sound is close to the original.

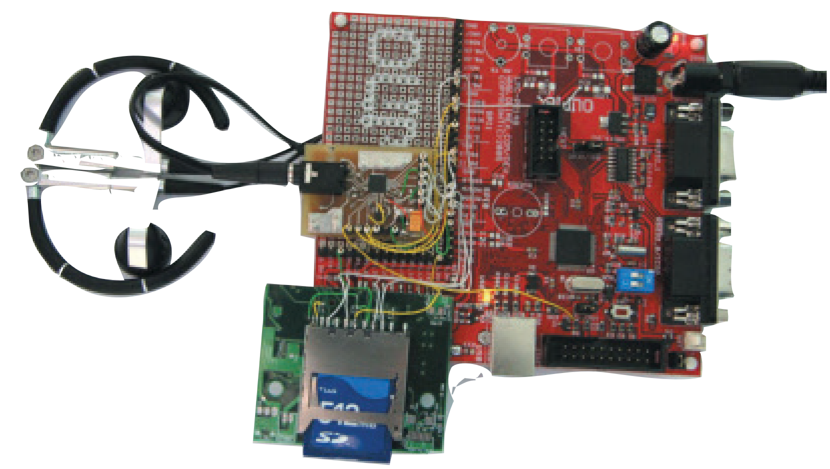


Photo 1: The prototype is built around an Olimex LPC2148 board. I added two extra prototype boards. One is for the SD card interface. The other is for the Texas Instruments audio DAC.

Crystal frequency	PCLK = MCLK	Audio sampling frequency Hz	Theoretical bit rate (Hz)	Theoretical clock divider	Real divider CSPDVSR	Error
11.2896 MHz	56.448 MHz	44,100 Hz	1,411,200	40	40	0%
12.0 MHz	60.0 MHz	44,100 Hz	1,411,200	42.517	42	1.20%
12.288 MHz	61.440 MHz	48,000 Hz	1,536.000	40	40	0%

Table 1: There are different SP11 clock settings for the various crystals and sampling rates.

AUDIO PLAYER PROJECT

My audio player is an embedded hardware/software solution for 16-bit digital audio. The purpose of the project is to achieve CD-quality audio, which means 16 bits (96 dB dynamic range) and a sampling rate of at least 44.1 kHz. If lower quality is satisfactory, you can use a simpler option. For example, you can use the onboard DAC (10-bit resolution and a maximum speed of 1 µs).

This project has three major functional blocks: a microcontroller, an SD card interface, and an audio DAC interface. The prototype is built around an LPC2148 evaluation board from Olimex and two extra prototype boards: one for audio DAC and one for the SD card (see **Photo 1** and **Figure 2**). Later on, I decided to integrate everything into one board to be a module/building block for use in embedded systems.

SYSTEM TIMING ANALYSIS

To achieve CD-quality audio (16-bit stereo with a 44.1 kHz sampling frequency), the required bit rate is 1,411,200 Hz (i.e., 16 bits (resolution) × two channels (stereo) × 44.1 kHz (Fs)). For 48 kHz, the bit rate is 1,536,000 Hz. This is the clock generated by the synchronous serial port (SSP), supplying data continuously to an audio DAC.

With a 12-MHz crystal (as used on Olimex and Embedded Artists evaluation boards), the timing is very close. The system clock generated by PLL from a 12-MHz crystal is CCLK = 60 MHz.

The other possible crystals are 11.2896 and 12.288 MHz. For NXP Semiconductors LPC21xx microcontrollers, the clock of the SSP is derived from peripheral clock PCLK, which is PCLK = CCLK/VPB. Its maximum value can equal to CCLK (when the VPB divider is 1). There is another divider hold in the CSPCPSR register to define the clock of the SSP (SCK1).

Table 1 shows some combinations of crystals and settings. If you want to use another clock or sampling rate, you can easily calculate your settings.

The data has to be read from the file on an SD card. The card is in FAT format, and it's compatible with a PC. The data has to be read and processed in real time. The processing required is very simple for uncompressed audio. It's more complex for compressed audio like MP3, which requires complex decoding.

HARDWARE

Failure is a new success. When I originally built the audio player, I didn't have enough RAM, and I failed to fully address the problem at the time. Fortunately, I eventually came up with a solution.

For this project, there are a few requirements for the microcontroller. You need at least 40 KB of flash memory and at least 40 KB of RAM. You also need one SPI for the SD card interface, a fast SPI capable of handling at least 16-bit data for the I²S interface, and one UART or other communication channel for connecting to an external system. A few different microcontrollers will work. NXP's LPC2148 and Atmel's AT-91SAM7S256 are the most popular. The latter has an SSI module with built-in I²S mode. The NXP microcontroller has a Texas Instruments DSP mode. It's no surprise that most audio codecs from Texas Instruments support its data interface with a glueless interface to the SSP on the LPC2148 microcontroller. For more generic solutions, you have to modify it to use the I²S. For that purpose, I've used some of the microcontroller's modules: Timer1 in Counter mode (with external input and output to provide LRCK for DAC) and PWM to generate MCLK for the audio DAC.

If you want to use an operating system in your application, then you'll need an extra timer. If your application has any other user interface, you might need some extra I/O ports.

Two extra prototype boards were

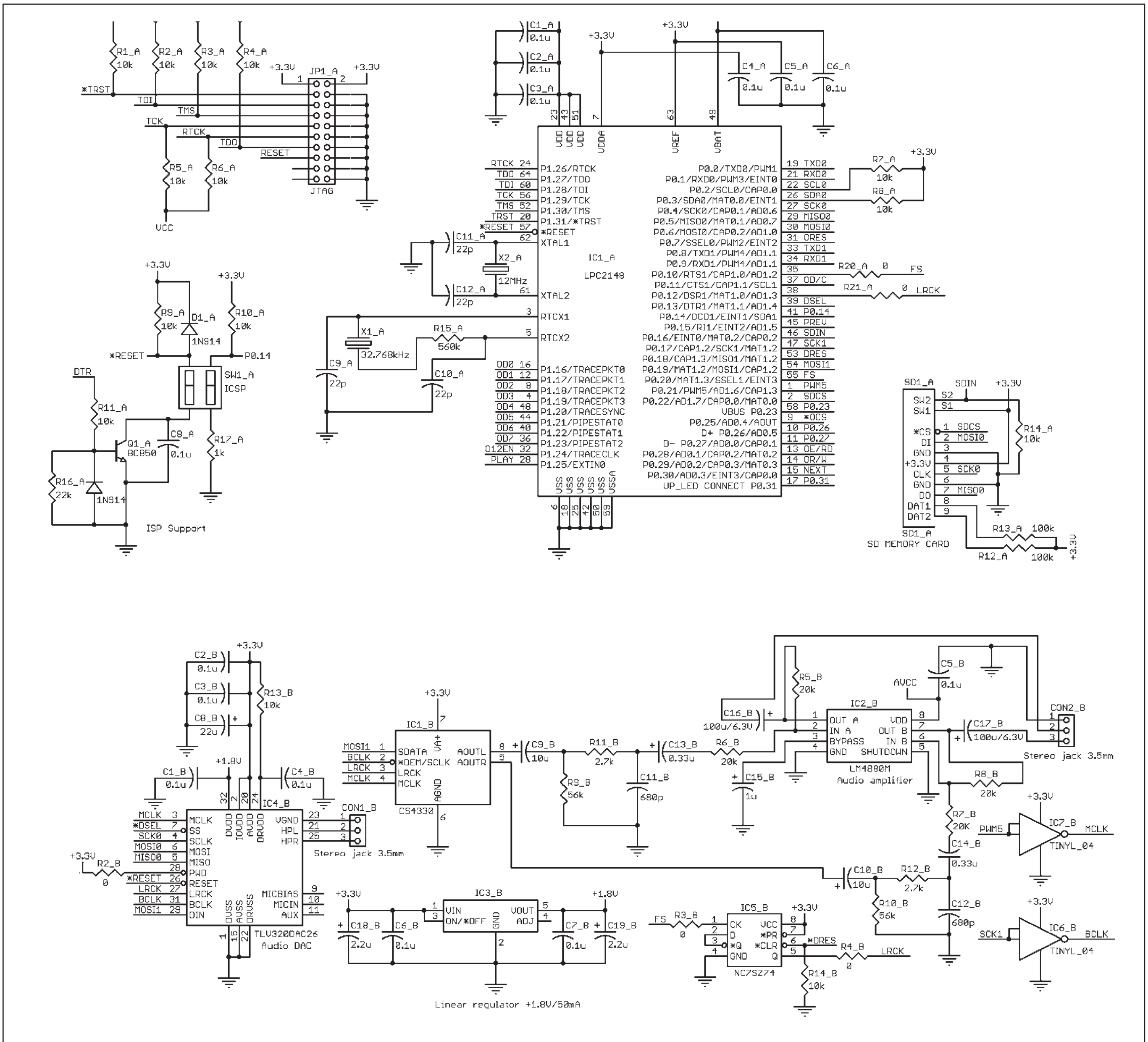
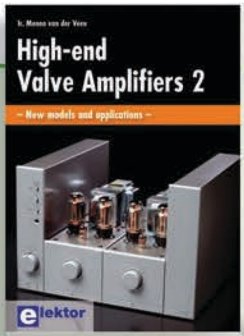


Figure 2: The system's main board is a modified Olimex LPC2148 evaluation board. Two alternative audio DAC circuits are shown. Use one of them or your own. The SD card interface is a standard SPI mode.

Books



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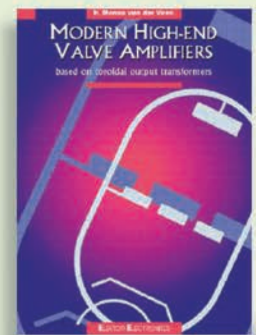

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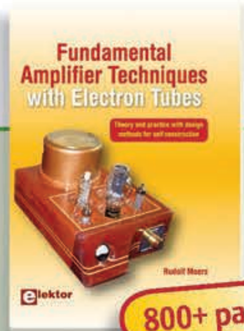

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DVD-ROMs

attached to the Olimex LPC2148 evaluation board. Let's take a closer look at each one.

SD CARD INTERFACE

The SD card operates in a standard SPI mode. The SPI0 has been used with its maximum clock of 7.5 MHz. It operates in 8-bit mode using SCK, MOSI, and MISO lines and another port line used as chip select (CS).

If you have any ARM7 evaluation boards from Keil, Olimex, IAR, Embedded Artists, or other manufacturers and they already have an SD card interface, you know that they all follow the same rule of using the SSP for that purpose. The idea is probably to give you a faster read/write time, but the better option for this project is to use SSP for the DAC interface. The SSP has an eight-frame buffer (FIFO) and corresponding interrupt source, both of which are not available on SPI0. This means the interrupt occurs more often when using SPI0 instead of SSP. On my prototype built around the LPC2148, I had to rewire it too, which wasn't difficult.

AUDIO DAC INTERFACE

I selected an I^{2}S standard interface for the audio interface because of its popularity (a number of codec chips available from different manufacturers). It's also easy to interface to the microcontroller.

Some audio DACs like the Texas Instruments TLV320DAC26 need extra control signals when initially setting up their internal registers in addition to I^{2}S signals and MCLK. With simple DACs like the Cirrus Logic CS4330-KS, for example, no setup is required. I have tried both of them. **Figure 2** shows the circuit for both options.

I used SPI0 lines shared with the SD card and an extra select line for the Texas Instruments DAC control interface. Although it needed 16-bit data, I used two consecutive transfers of 8 bits and it worked fine.

For some extra complexity of control interface, the chip has a built-in headphone amplifier, as well as programmable volume control and other controls. It also has an internal PLL, which gives you flexibility for selecting the external clock.

For an MCLK signal, you can use the microcontroller's crystal (directly or through a buffer) or generate it inside the microcontroller, as I did. The code within the `init_hardware()` function in the mp.c file on the *Circuit Cellar* FTP site is used for the BCLK signal to be generated on pin PWM5 (P0.21/PWM5 pin 1).

24-MINUTE I^{2}S INTERFACE

Most audio DACs require an I^{2}S or a similar (left- or right-justified) data interface. The LPC2148 microcontroller has two SPIs, but neither of them works in I^{2}S mode. Only one of them (the SPI1 or SSP) is capable of processing the 16-bit data required by the I^{2}S interface. That means you have to use it and do some modification to convert to I^{2}S.

The microcontroller's SSP is used in a 16-bit Texas Instruments DSP mode to make it easy to convert into I^{2}S. Refer to NXP's "LPC214x User Manual" for a description of the possible modes of operation.

A few steps are involved with converting from DSP mode to I^{2}S. The data output from SPI1 doesn't need any modifications to be used as I^{2}S data. The left/right clock (LRCK) can be derived from the FS signal in the straightforward manner. The FS signal, which is output on the SSEL1 pin (P0.20/SSEL1 pin55) in TI mode, is connected to the Timer1 input capture pin (P0.10/CAP1.0 pin 35). Using the

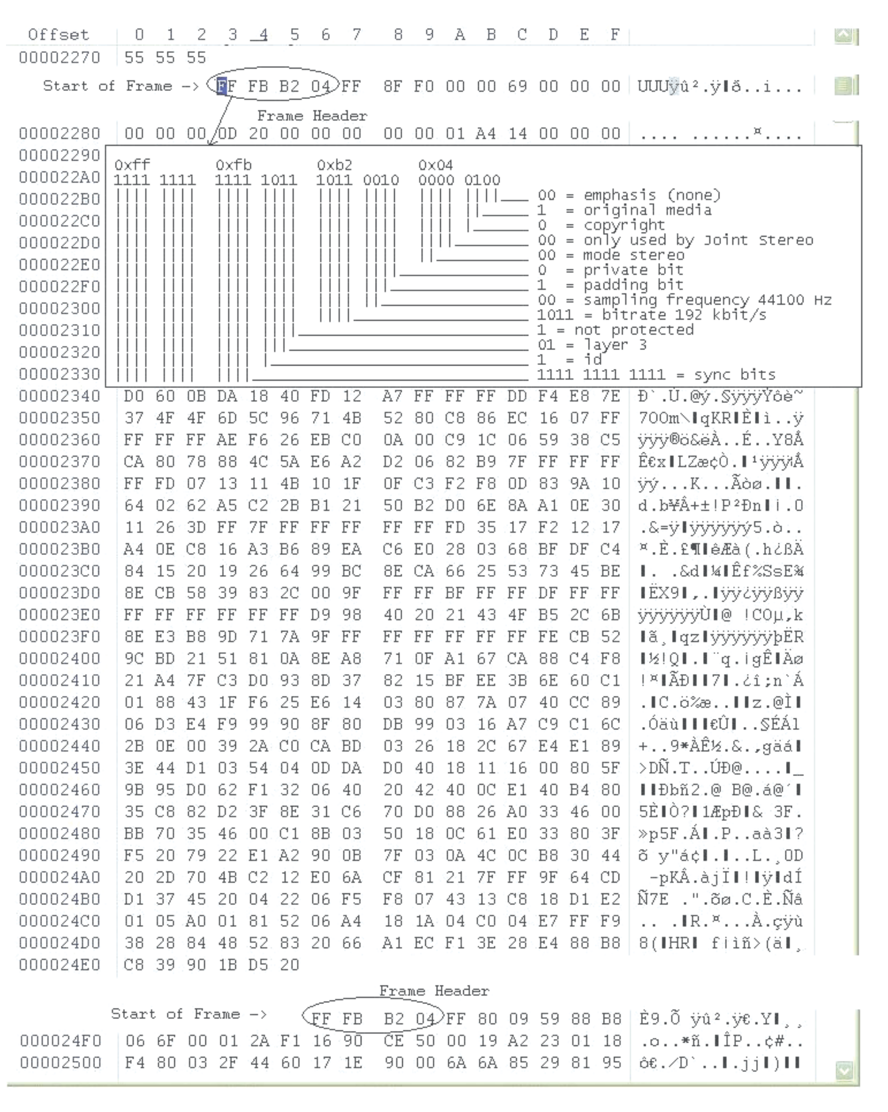


Figure 3: The MP3 file format is in the hex editor.

timer's one input capture and output match function, the FS signal on the input pin of Timer1 (P0.10/CAP1.0 pin 35) is converted into an output signal on pin MAT1.0 (P0.12/MAT1.0), which represents the LRCK signal of the I²S interface.

To complete the I²S interface, the clock must be inverted. This is achieved via the tiny logic inverter (e.g., a Fairchild Semiconductor NC7SZ04). Connect the inverter's input to SCK1 (P0.17/SCK1 pin 47) and its output to BCLK on the I2S.

If you use another DAC with a left- or right-justified format instead of I²S, you can achieve it by slightly changing the I²S modification because they are similar. All of the settings of the internal modules are done once after power-on reset. The code is in the `init_hardware()` function in the mp.c file. If you plan to use any of the low-power modes, you might need to stop some modules and activate them later during wakeup. If you want to save valuable resources like Timer1 and PWM5 for other purposes, you can use a simple D-latch (e.g., the TinyLogic NC7SZ74) to do the same in hardware.

```
void sspISR(void)
{
unsigned char c;
c = SSPMIS;
SSPICR = 0x03;  // Reset INT errors
left.b8[0] = *pcm_play_ptr++;
left.b8[1] = *pcm_play_ptr++;
right.b8[0] = *pcm_play_ptr++;
right.b8[1] = *pcm_play_ptr++;
left1.b8[0] = *pcm_play_ptr++;
left1.b8[1] = *pcm_play_ptr++;
right1.b8[0] = *pcm_play_ptr++;
right1.b8[1] = *pcm_play_ptr++;
SSPDR = left.b16[0];
SSPDR = right.b16[0];
SSPDR = left1.b16[0];
SSPDR = right1.b16[0];
if(pcm_play_ptr > (unsigned char *)PCM_BUFFER_END)
    pcm_play_ptr = (unsigned char *)PCM_BUFFER_START;
VICVectAddr = 0; // Update VIC priorities
}
```

Listing 1: Take a look at the interrupt service routine for the Tx FIFO half empty interrupt. The sequence of four 16-bit words is written into the transmit register from the PCM buffer. The data in the buffer comes either directly from the wave file or from the output of the MP3 decoding function.

SOFTWARE

I tried to make the code as simple and as minimal as possible so it would be easy to understand. The code currently supports WAV and MP3 formats. You can find information about both formats on the Internet if you're interested.

The WAV file format supports a variety of bit resolutions, sample rates, and channels of audio. A WAV file is a collection of a number of different types of chunks. There is a required format ("fmt") chunk, which contains important parameters describing the waveform, such as its sample rate. The data chunk, which contains the actual waveform data, is also required. The other chunks are optional.

The minimal WAV file consists of a single WAV containing the two required chunks: a format chunk and a data chunk. There are a number of variations on the WAV format, but if you convert from CD to WAV format, then most of the software will create the simplest WAV file.

MP3 is the other popular audio file format for compressed data. MP3 files are composed of a series of frames. Each frame of data is preceded by a header, which has extra information about the data to come. Extra information, which is called ID3 data, may be included at the beginning or end of an MP3 file. **Figure 3** shows the MP3 format with a hex representation of bytes and an explanation. You can investigate it by opening the file in any hex editor. I use WinHex because it has a disk utility as well.

HALF EMPTY

The SSP provides a continuous stream of 16-bit data for the audio DAC. The presence of the eight-frame FIFO and the Tx FIFO half empty interrupt makes for an easy data-loading scheme (see **Listing 1**).

I created a large buffer for storing PCM data that acts as a circular buffer. One pointer manages the writing into the SSP transmit FIFO (and later audio DAC). Another pointer manages the loading of new data from the SD card. For uncompressed audio files on input, the loading comes directly from the WAV file. For MP3 files, the data has to be decoded beforehand.

The simple synchronization is achieved by checking the distance between pointers with software. The software functions related to the SSP are the initial setup, ISR, and buffer loading functions. (The SSP's receive function isn't used at all, so the Rx of the at least half full interrupt is disabled.)

CARDS ON THE TABLE

Almost everyone knows how to use and program an SD card. There are a number of examples freely available. The embedded file system library (EFSL) is very popular. I used bits and pieces of my old code for the initialization and reading of files. My code covers only FAT16, but it's rather small and uses very little RAM.

I put the open-source software for MP3 decoding into a separate directory (\fixpt) as part of CrossWorks for an ARM project. The compiler is based on GNU. There is a nice debugger in the package for easy development. Rowley offers a fully functional, time-limited evaluation version.

I was positively surprised that I had only a few compilation errors. After commenting out some unnecessary statements, everything compiled smoothly.

I haven't used the dynamic memory allocation, but I have a rather static memory assignment. As a result, my `mp3_init` function is different, but you can do it either way. A dynamic allocation is more elegant because you can free up the memory for other uses.

My `SD_play_mp3()` function is based on the open source example. Only three functions from the package are called: `MP3FindSyncWord()`, `MP3Decode()`, and `MP3GetLastFrameInfo()`. I created the `myFillReadBuffer()` function to handle the readings from the SD card and filling the input buffer.

I usually use the "It's not you, it's me" approach when something doesn't work. After fixing some bugs in my code, everything worked fine and I didn't have a need to go any deeper into the MP3 code. If you're curious about how it's done, browse through the code and read the comments.

If it doesn't work, you're not hitting it with a big enough hammer. If your hardware is identical or similar to mine, you might first try to load the hex code using the NXP ISP utility. If it works and you need to make changes, you can rebuild the sources. Remember to turn on the optimization for the MP3 code in the GNU compiler. The "–Os" option worked for me.

There is no embedded video included in the project yet, but don't be surprised if it's as easy to do in the near future as it is to do today with embedded audio (that's if Moore's law still applies).

I've always admired people with artistic ability. Although embedded system design is more of a craft than an art, you can still make everything work well and look good.

FACE THE MUSIC

There are limitations in the current version, but you can extend or modify it to suit your application. There might be bugs in my code, so feel free to modify that too. The music can be good or bad, but embedded systems come in a variety of flavors.

Set the volume control below the threshold of pain and let the electrons and holes inside the silicon do the job. Here in Australia, I typically tune in to the Triple J radio station (105.7 FM), where the music is not only good, it's fresh. *aX*

Author's note: I would like to thank my son Jakub and daughter Jola for their support and unbiased criticism.

Editor's note: This article originally appeared in *Circuit Cellar* 194, 2006.

RESOURCES

Embedded Filesystems Library (EFSL), www.efsl.be and http://sourceforge.net/projects/efsl/.

Helix DNA Downloads, http://forms.helixcommunity.org/helix/builds/.

D. Katz and R. Gentile, Embedded Media Processing, Elsevier, New York, NY, 2006.

A. Sloss, D. Symes, C. Wright, ARM System Developer's Guide, Elsevier, New York, NY, 2004.

Texas Instruments, Inc., "Low Power Stereo Audio DAC with Headphone/Speaker Amplifier," SLAS428, 2004.

M. Thomas, "Interfacing ARM Controllers with Memory Cards," 2006.

SOURCES

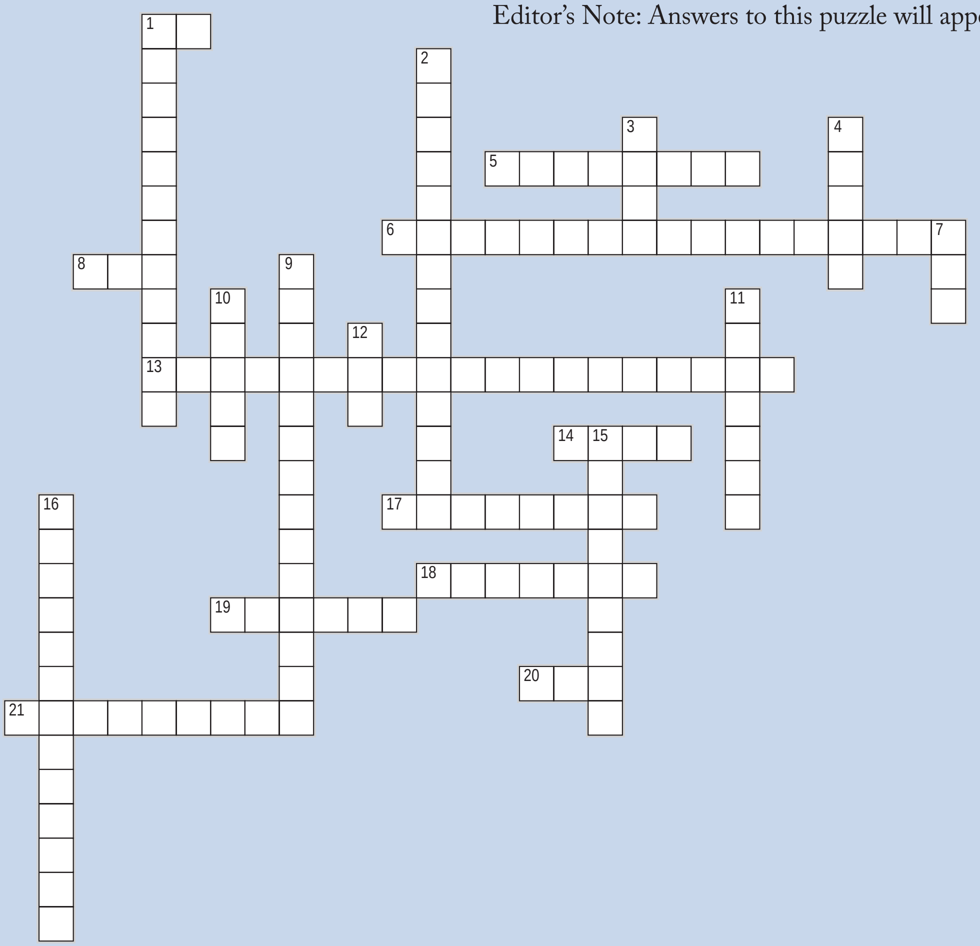
LPC2148 Evaluation board
Olimex
www.olimex.com

LPC2138 Microcontroller
NXP Semiconductors
www.nxp.com

CrossWorks for ARM
Rowley Associates
www.rowley.co.uk

WinHex Hex Editor
X-Ways Software Technology AG
www.x-ways.net/winhex/index-m.html

Editor's Note: Answers to this puzzle will appear in the next issue.



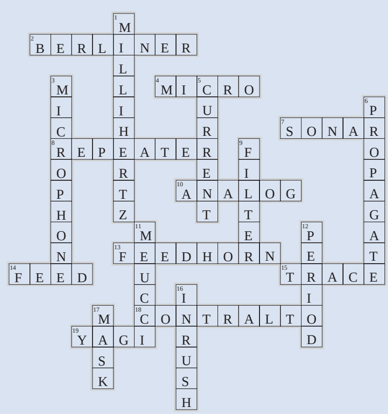
ACROSS

1. Abbreviation for decibel
5. Pertaining to any undesirable signal components not present in the original signal
6. A measure of data transmission speed often used for MP3s and other streaming file transfers (three words)
8. Pulse-length modulation
13. A power supply for a device that is not integrated inside the device itself (three words)
14. Kilobits per second
17. Erase all data that was stored on the drive
18. An improved version
19. An interconnected group of components that work together
20. Digital signal processing
21. A passive electronic component consisting of two foil conductors separated by a thin insulating layer

DOWN

1. A mixing board or console that uses DSP to combine and process audio signals (two words)
2. A microphone that features tube-based internal pre-amplification electronics (two words)
3. High-frequency noise, as in the sound of the letter "S"
4. Short for specifications
7. Digital-to-analog converter
9. A digital integrated circuit which performs one or more specific pre-programmed functions
10. To re-route a signal to a different circuit
11. Amount of computer data generally assumed to equal 1,024 bits
12. A "wave" format audio file
15. Live recording made directly from the outputs of a PA system's mixing console (two words)
16. A type of magnetizable substance based on iron

Answers to the November 2011 Puzzle



Elysia's 19" rack-mounted Xpressor stereo compressor

19" RACK-MOUNTED STEREO COMPRESSOR

High-end studio signal processing company Elysia now offers a 19" rack-mounted Xpressor stereo compressor ($1,499). Based on the extremely successful Xpressor 500 module for API 500 series, the new Xpressor rack adds a side chain circuit along with a superior linear power supply with a shielded toroidal transformer.

Its discrete circuitry runs in constant Class A mode, and was designed to provide superior audio quality, combining a clear and open sound with a flexible amount of punch. Many of the unique features of the Elysia flagship products have been adopted by the Xpressor, such as Auto Fast attack mode, switchable release characteristics, and Warm Mode, which adds character and color to the signal. The Xpressor also provides a built-in side chain filter, Elysia's gain reduction limiter and a mix control knob for onboard parallel compression.

The unit is perfect for stereo buss compression as well as processing individual signals, and is an ideal tool for approaching dynamics processing in creative ways. For more information visit, www.elysia.com

SCARLETT 2i2 RECORDING INTERFACE

Focusrite now offers the Scarlett 2i2 audio interface—a 2-in/2-out USB 2.0 audio interface with the highest specifications.

With Focusrite mic/instrument preamps and high-quality 24-bit/96 kHz digital conversion and flexible monitor control, it features a new unibody industrial design, which is visually striking, functional, and robust. Its key features include:

- High-quality, Focusrite mic preamps
- Anodised aluminium unibody chassis
- Unique LED-halo signal indicators
- Professional-standard 24-bit/96-kHz digital conversion
- Direct Monitor function for zero-latency tracking
- Powered by USB connection
- Focusrite Scarlett plug-in suite and Ableton Live Lite included

At the heart of the Scarlett 2i2, two Focusrite microphone preamplifiers provide an exceptionally clean signal path for mic-, instrument- or line-level sources, and supply phantom power to condenser mics. Best-in-class digital converters capture pristine-quality audio for professional-sounding results, and a new unibody chassis—a single piece of red anodised aluminium—surrounds the 2i2's internal circuitry and controls. Not only does it sound and look fantastic, the Scarlett 2i2 is also rugged enough to take knocks on stage and can be thrown into a laptop bag to make high-quality recordings anywhere. Other innovations on the Scarlett 2i2 include unique LED-halo signal indicators, which encircle each of the gain controls. These glow green when an input signal is detected and turn red if clipping occurs, to provide a quick and easy way of ensuring that an appropriate signal level is achieved.The Scarlett 2i2 has a pair of quarter-inch TRS jack outputs for feeding powered studio monitors or an external audio device, such as a mixer. A large knob provides accurate control of the output level, and direct monitoring allows for zero-latency tracking. There's also a quarter-inch headphones output with dedicated level control. Since the interface is bus powered, an additional power supply is not needed, meaning the interface is truly portable. What's more, it's bundled with the Focusrite Scarlett Plug-in Suite as well as Ableton Live Lite: all that's needed to get recording right away.Combining professional-sounding mic preamps with the very best in USB 2.0 interfacing technology, the Scarlett 2i2 is perfect for those new to recording, and for professionals who want the Focusrite quality in a compact, mobile and robust package. The Focusrite Scarlett 2i2 joins the existing line of Focusrite USB and Firewire interfaces, and will not supersede any current Focusrite products. It started shipping worldwide in October 2011 for $199.99. For more information, visit Focusrite at www.focusrite.com

PERFORMANCE MANAGER SOFTWARE

At PLASA 2011, Harman showcased its JBL HiQnet Performance Manager software, which has undergone an extensive beta test period in advance of its imminent shipping. JBL HiQnet Performance Manager is a highly refined user interface that facilitates the design of touring and live performance venue sound reinforcement systems. HiQnet Performance Manager is an application-specific iteration of the category-leading Harman HiQnet System Architect connectivity and control software application for professional-grade audio system integration.

Since its successful launch at Prolight + Sound 2011 and live demonstrations at InfoComm 2011, the software has been deployed to a group of 45 online beta testers in all parts of the world. The beta tester community has examined software functionality and performance by putting pre-release versions of Performance Manager through their paces on a host of varied system applications. With the experiences gained through these Performance Manager systems, the beta team has been able to provide considerable real-world feedback to ensure the highest level of usability in the field, openly discussing operational enhancements and refinements with the Performance Manager team and with each other.

Performance Manager's Run Show mode is geared toward live performance. It provides real-time monitoring and adjustment for EQ and dedicated monitoring interfaces for levels, speaker loads, thermal conditions, and more.

HiQnet Performance Manager provides a comprehensive, step-by-step workflow that directly corresponds to real-world system configuration, taking the workflow paradigm introduced in System Architect to a higher level of functionality for any live performance audio application. It is fully integrated with JBL's Line Array Calculator II loudspeaker configuration and acoustic modeling software. The user begins by loading templates of the speaker arrays used in the system, and then runs Line Array Calculator II for each array as part of the initial sound design task of determining how many and which type of loudspeakers are required to cover a given venue. For each array, Performance Manager automatically loads the passive VERTEC or powered VERTEC DrivePack DPDA line array configuration into the main application workspace—the first of many automated design processes native to the software application. Loudspeakers can also be manually loaded into the templates if desired.

Once the user defines the required amplifier parameters for the passive loudspeakers within the arrays, Performance Manager automatically loads the correct number of Crown Audio VRACK or other user-determined amplifier racks into the audio system. The software then associates the amplifier outputs with the bandpass crossover inputs for the selected array and programs the amplifiers with the correct JBL preset data, as well as gain shading and JBL Line Array Control Panel equalization parameters that are determined in JBL's Line Array Calculator II as part of the modeling process to optimize sound pressure level and frequency response over the defined audience geometry.

Representations of the bandpass inputs for each loudspeaker section are overlaid onto the arrays, enabling the user to easily visualize the array configuration, whether JBL DrivePack-powered or driven by external Crown power amplifiers. Performance Manager software also significantly simplifies system-networking configuration. The user can simply drag and drop devices discovered on the network onto the pre-configured devices within the Performance Manager workspace to synchronize all addressing and parameter values.

In addition, the Performance Manager graphical interface provides embedded control panels for array calibration, time alignment and system EQ which utilize input section digital signal processing resources available in either Crown I-Tech HD power amplifiers or JBL DrivePack-powered loudspeakers with DPDA digital audio input modules. Input attenuation, equalization, delay settings and bandpass controls are all readily accessible directly within the main application workspace along with flexible grouping and comprehensive solo/mute functionality for system testing.

Once system tuning is complete, Performance Manager's Show Mode display is optimized for the actual live performance, offering appropriate adjustment control ballistics for equalization and dedicated monitoring inter-

Harman's JBL HiQnet Performance Manager software faces for levels, speaker loads, thermal conditions and AC power requirements. The workspace for all stages of Performance Manager's workflow has a common design motif, with monitoring functions overlaid on top of the same loudspeaker bandpass representations within the workspace, making visualization easier and more consistent across various workflow screens.

For more information, visit Harman's website at www.harman.com.

REASON 6 AND BALANCE AUDIO INTERFACE

One day after its official launch, Swedish producer and Propellerhead product specialist Mattias Häggström Gerdt showcased the new Reason 6 and Balance audio interface at the BPM Show October 1–3, 2011 in Birmingham, United Kingdom.

Balance is Propellerhead's new 2-in 2-out audio interface 9. Balance sounds as good as it looks and works as great as it feels. Users don't need to worry about calibration or setup. You can concentrate on having fun instead: making music. With two channels of pristine audio recording, and with inputs for all your gear, your instruments are always connected and you are always ready to record. Balance is perfectly integrated with Reason and Reason Essentials. With Clip

Propellerhead's new Reason 6

Propellerhead Balance audio interface

Safe, you'll never lose a good take to distortion again. Big Meter lets you tune your guitar and set the levels from across the room. And Balance has the Propellerhead Ignition Key built in. Use it with Propellerhead software, other pro audio apps, or even iTunes Balance works with everything. www.propellerheads.se/products/balance/index.cfm.

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Contributors

Robert Nance Dee ("The Willow Pre-amp," p. 27) is a retired electronics engineer. He worked with the first military computers, has several patents, and has written many magazine articles. He enjoys machining, watch making, bicycling, cabinet making, and philosophy. He is an amateur arborist, an environmentalist, and a vegetarian who thinks his life is a hobby. His passion for audio, in every form, spans 50 years and started in high school while working part time in a radio/television repair shop. He lives off the grid in a house he built surrounded by animals and fruit trees in the Catskill Mountains. His audio projects are named after is animals and friends.

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Jan Szymanski ("Digital Audio Player," p. 33) is an electronics engineer who specializes in embedded systems design. He runs Cherry Microsystems, a consulting company in Australia. In his spare time, Jan enjoys music and sports.

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