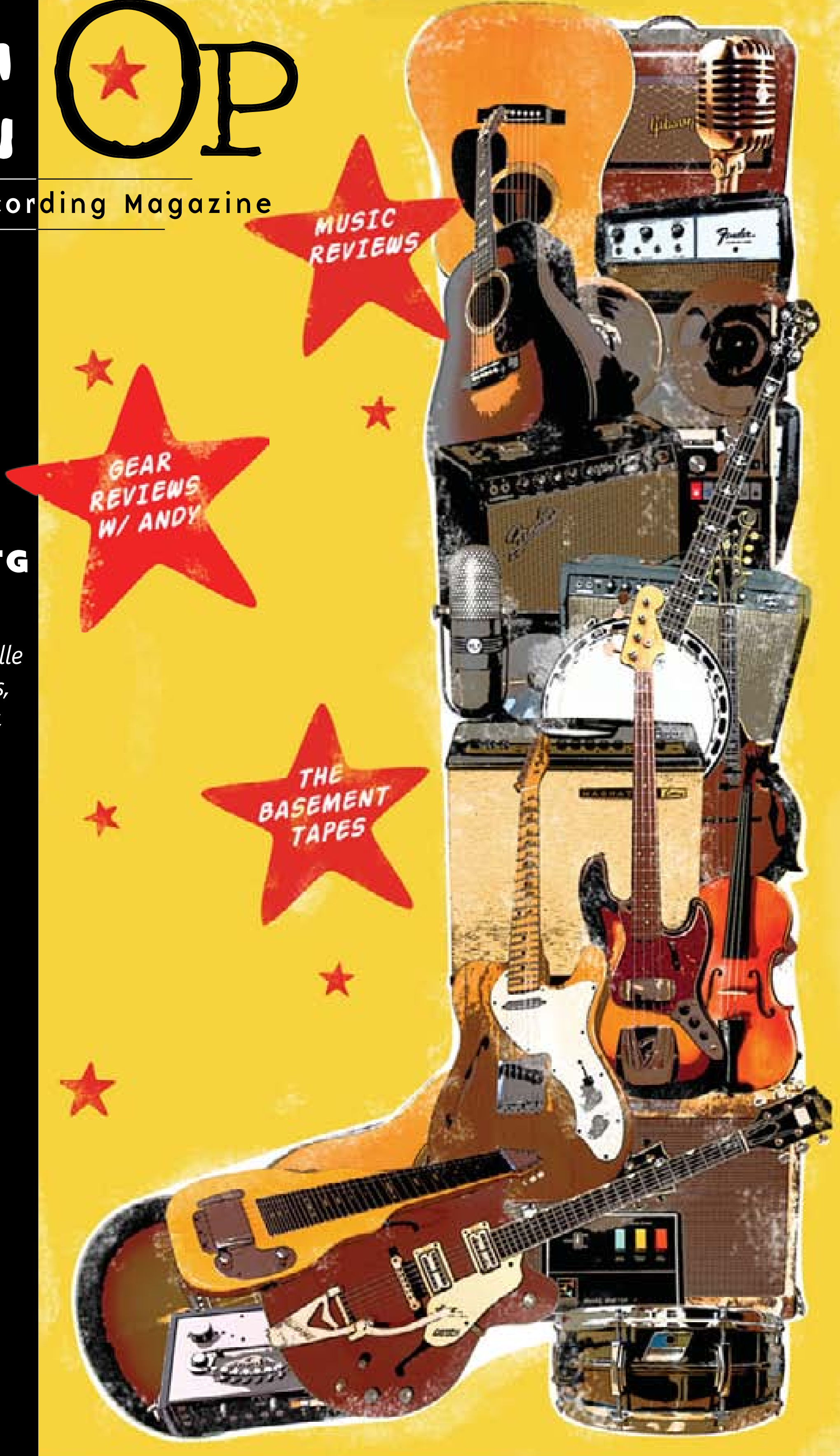
Tape Op

The Creative Music Recording Magazine



Kyle & Jason Lehning

Gary Paczosa

Three engineer/producers based in Nashville discuss working w/ Nickel Creek, Alison Krauss, Waylon Jennings, Randy Travis, Good Old War, and Mat Kearney.

Mew

Copenhagen's Finest

Kim Rosen

of Knack Mastering

James Demeter

in Behind the Gear

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TAPE OP GOES TO NASHVILLE!

Tape Op GEAR ReVIEWs

Audio-Technica

ATH-R70x open-back reference headphone

I love this new model from Audio-Technica's growing line of professional studio headphones. It features open-back earcups housing proprietary 45 mm drivers, and a unique headband with spring-loaded "wings" that self-adjust the fit. It's by far the lightest and most comfortable circumaural headphone I have ever worn. The earpads are covered in a breathable microfiber fabric that remains plush and cool, and there's just enough cushion to keep the drivers close to my ears for maximum fidelity, without earlobe contact. Being the gear geek, I appreciate the design of the Y-cable that plugs into the earcups. Each of the cable's earcup ends has a turn-to-lock, three-conductor TRS plug, so it doesn't matter which plug you use in which earcup jack; each receiving jack connects to the correct conductors for its respective left or right signal. Ironically, the headphone looks perfectly symmetrical, so the only way to tell the L/R assignment of the earcups is to look carefully at the inside of the headband for near-invisible markings identifying the sides. (I ended up placing a sticker on the right-hand side of the headband.) Also, only one Y-cable is included, and it's over 10 ft long, which is great if you're sitting in your favorite La-Z-Boy while listening to your hi-fi, or adjusting a compressor in a far-away rack against the side wall of your control room, but my preference is for a cable half that length. The only included accessory is a polyester-fleece carrying bag.

Overall, I would characterize the sound of the *ATH-R70x* as warm and inviting, but with plenty of honest detail. Low-frequency extension is exemplary for an open-back design, and I can clearly discern fundamental notes down to 22 Hz. Some distortion is audible at these extremely low frequencies, but I hear far less distortion in the *ATH-R70x* than in my open-back Shure SRH1840 headphone [*Tape Op* #89], particularly below 34 Hz. Moreover, the *ATH-R70x*'s distortion is devoid of high-order harmonics, so it's less distracting, and deep bass notes don't sound falsely clicky. Moving up the spectrum from there, the volume takes a shallow slope up to 83 Hz, then tilts smoothly downward, contributing to the warmth I hear. Importantly, driver damping is well controlled, so there's very little distortion, ringing, or time-domain smearing in the lows and lower mids. A 3 dB peak at 3.5 kHz is followed by a –3 dB trough at 4–7 kHz, with some ringing at both 3.5 kHz and 8 kHz. Above 12 kHz, the extreme highs are gently tamed, but high-frequency detail is still there, thanks to good transient response and very quick settling time. Imaging is precise, with a strong phantom center surrounded by a wide soundstage.

Keep in mind that our individual head and ear shapes will affect how headphones sound, so what I hear in the *ATH-R70x* may not be exactly what you hear; but my listening notes should give you a general idea of what to expect. Also, don't let any of the numbers above scare you, because the peaks, dips, resonances, and distortions you might hear in high-quality headphones are far less consequential than the substantial blurring and inaccuracy caused by speaker-room interactions in all but the most perfectly treated rooms. My tests were done over a month's time — not only performing mixes and listening critically to familiar music (including songs I had mixed prior), but also listening to individual instrument sounds as well as test tones and impulses.

I'm not a fan of lifestyle headphones that emphasize bass and hype the highs. In contrast, the sound of the *ATH-R70x* is relatively neutral and absolutely unfatiguing, even for hours at a time. At $350 street, the *ATH-R70x* is not an impulse buy, but it's more affordable than many "luxury" headphones. I really enjoy listening to music on "open-air" headphones, and I own many models, going all the way back to several Sennheisers that I purchased in the '80s. But the *ATH-R70x* is the first open-back headphone in my collection that I would be confident using alone as a mixing reference, without a second closed-back headphone or a well-tuned subwoofer to inform me of the lows that would otherwise be missed or misrepresented. Consequently, this is the headphone I take with me between studio and house.

($349 street; www.audio-technica.com) –AH

Harrison

Lineage 8-channel mic preamp

I haven't spent any time on a Harrison console, but I do know that many important albums and films were recorded or mixed on Harrison consoles over the decades, and company founder Dave Harrison kickstarted his console-designing career with the MCI JH-400 series, the first production desks with an inline layout. The *Harrison Lineage* pays homage to 40 years of Harrison consoles by assembling pairs of four different mic preamp designs (a total of eight channels) in a 1RU-height rackmount chassis. Opening the unit reveals lots of discrete through-hole components, some op-amps, and a couple transformers — all carefully laid out to prevent crosstalk between unrelated functions. Moreover, an internal power supply is fenced off from the rest of the circuitry to further prevent noise.

Over the course of months, I used the *Lineage* preamps on various sources, but I did so with no prior experience with Harrison products, so I had no expectations of sound or performance between the various designs. Here's what I discovered:

The two *32c/MR Series* preamps recreate the circuitry from Harrison's '70s and '80s–era 32c and MR consoles. I love how these push vocals, especially female vocals, forward in the mix. The same goes for electric guitar and horn solos. Additionally, extreme lows, particularly frequencies below 35 Hz, are emphasized, whether you're using ribbon, dynamic, or condenser mics. For example, kick drums recorded through these have a ton of power — sometimes too much. With great power comes great responsibility; and restraint is required when EQ'ing lows. I hear transistor distortion ramping up earliest on this design, and it's easy to capture a big, rock and roll sound, with a nice bit of "compression" from the discrete new-old-stock 2SD786 transistors saturating, especially on drums. But you do have to be careful — too much of a good thing can put you in dark and wooly territory.

The *Series 10/950* inputs recreate the preamps from the SeriesTen, introduced in 1985 as the first fully automated desk. With these, I can hear a tiny bit of crunchiness even on gently played transients like high-hat hits and acoustic guitar notes. Also, distortion ramps up quickly in the form of lower-midrange "bloom" (or "mud") when overdriven. This design stands out as being most different sounding among the four.

The *Series 12/LPC* inputs recreate the preamps from the SeriesTwelve multiformat console introduced in 1992. To my ears, these are the most natural sounding and most true to picture, with the clearest midrange. Not surprisingly, they respond best to EQ — nothing gets too brash or feeble, even after heavy-handed boosts or cuts. These are my favorite for drum overheads; a Royer SF-12 stereo ribbon mic [*Tape Op* #25] above the kit through the *Series 12/LPC* results in perfect imaging, with natural transients. It's near-impossible to make these preamps sound harsh, even with extreme processing.

The *Trion* channels feature the analog preamps first seen in the Trion digital mixer in 2005. These sound closest to the *Series 12/LPC* preamps, with a very natural sound. On the other hand, they have more character due to the Lundahl transformers on their inputs. For example, toms and kick drums tend to sound bigger than life through the *Series 12/LPC*, without the overt distortion from the *Series 10/950*; and electric guitars take on more weight, without getting blurry. You do have to be careful EQ'ing lower mids, especially in the 250 Hz range, and these have the least amount of infrasonics below 30 Hz. Overall, I think the *Trion* preamps are the most versatile preamps in this box. They're clean, but not clinical. You can turn them up for more transformer saturation to bring out more character, which is great for drums and anything overdriven, including guitars and vocals. Moreover, the *Trion* preamps offer the most gain (70 dB), and you can also use them as FET-based instrument DIs via the front panel Neutrik combo jacks. Each *Trion* channel has a switch to choose between front/back inputs, and additionally, a "Fix" switch to swap out the input gain knob for a separate, recessed gain trim on the front panel. What's cool about this feature is that you can leave two pairs of mics attached to the *Trion* channels and switch between the two; or you can do what I did and use one pair of inputs for a passive summing box leaving the other pair open for mics. The neat thing here is that I can enable the "Fix" trims when selecting the summing box — which allows me to adjust the amount of "analog goodness" to taste, then recall at the touch of a button!

All eight preamps include switches for phantom power and –20 dB pad, as well as signal-overload LEDs that indicate +18 dBu level. All four preamp designs are quiet, with plenty of gain (*Trion* having the most).

If you're looking for a quick and easy way to add different preamp flavors to your recording kit, the *Harrison Lineage* is a great way to go. You get eight high-class mic preamps with four different sounds, taking up only one rackspace. Moreover, two instrument DIs are included, and you can set up a pair of alternate, trimmable-gain inputs, allowing you to assign the *Trion* preamps to a summing box or other recording/processing flow with painless recall. Although the price of the Lineage may seem high at first glance, if you divide it up, $350 per preamp channel is a steal. Or another way to look at it — for the cost of a 500-series chassis filled with eight channels of boutique preamps, you can buy two *Lineage* units, and you'd still have room left over in your rack!

($2,795 street; www.harrisonconsoles.com) –AH

Focal Professional
SM9 active studio monitor

The quest for the right studio monitor can be long and frustrating. Or, the first pair you audition can instantly become the pair you know and trust, allowing you to do your best work right from the get-go. Focal Professional's flagship French-made *SM9* monitor was the latter for me — and for everybody at Figure 8 Recording in Brooklyn. My initial plan was to audition a few models to find the right set of monitors for our main control room. As it happened, I set up the *SM9* pair for the inaugural session at the studio last November — overdubs with guitarist Nels Cline and bassist Trevor Dunn for composer/drummer Scott Amendola's ambitious and amazing orchestral record *Fade to Orange*. And a few months later, I remembered that I wanted to try out some other speakers. In the meantime, dozens of projects had come through the studio, and every single one of the six engineers working in the space had effused to me how happy they were tracking and mixing on the *SM9*. This is just clearly a next-level speaker.

Focal markets the *SM9* as "two monitors in one," due to the fact that in its default state, it's an active 3-way speaker, with a 400 W amp for the forward-facing 8" subwoofer, and one 100 W amp each for the 6.5" woofer and 1" tweeter. With a push of the (vaguely-named) Focus switch, the amp for the subwoofer is turned off and the crossover is altered, so the frequency response changes from 30 Hz – 40 kHz to 90 Hz – 20 kHz (both ±3 dB). Focal claims in its literature that this is to check how the mix will sound on your TV or "multimedia systems." I got a chuckle thinking that any TV speaker could sound as good as the *SM9*, even with the Focus button on. This feature does not eliminate the need for Auratones, boomboxes, or other "real-world" reference speakers. But, point taken.

In addition to the subwoofer, the *SM9* has an 11" passive radiator on its upper surface, which gives it a definitive look and also happens to preclude stacking a second speaker on top of it. I find the sub range of the *SM9* very even and trustworthy, and I have to credit the design of the passive radiator with some of that evenness. It functions in lieu of a bass port and couples with the subwoofer to help reproduce those low frequencies without the distortion and turbulence that ports can cause. The one frequency range of the *SM9* that sounded a little weak in our control room was the lower midrange. Initial mixes were coming out of our room needing a slight tuck in the 120–180 Hz range at mastering. A 1 dB boost on each speaker on the LMF control, which is centered at 160 Hz, has really helped us hear that range better, and mixes have been translating excellently since that small adjustment. Other EQ points on the rear of the *SM9* are LF (50 Hz), and MF (1 kHz). There are also shelving filters from 250 Hz downward and 4.5 kHz upward, and an HPF with three settings (45, 60, and 90 Hz), each at 12 dB/octave, allowing you to run the *SM9* with an external subwoofer if desired. All EQ settings, aside from the HPF, are conveniently bypassed with the single push of the (also vaguely named) Direct Input button.

Other than that small boost in the low-mids, the rest of the EQ has stayed flat in our setup, because the interaction of the three drivers just works so well and sounds so good — at all listening levels. A hallmark of an excellent speaker for me is being able to trust it for quiet monitoring, and the *SM9* comes through in this regard fantastically. The stereo imaging, which I find absolutely exquisite on the *SM9* pair, maintains its separation and depth at low listening levels, while the "phantom" center channel still sounds robust and forward. At louder levels, the frequency response also sounds very even; I hear no unwanted resonances or discernible crossover distortions, and the high end doesn't get shrill. The tweeter is made of beryllium and is an inverted dome construction, which is one of Focal's hallmarks. I have experience using this tweeter on the smaller Solo6 Be speaker [*Tape Op* #60] over the past decade, finding it trustworthy and easy to listen to for long stretches of time.

Once we did get around to comparing the *SM9* to a few other models, we spent a full day leisurely listening to four sets of speakers. We wanted all of them set up on the same stands with the same room placement, so we did no quick A/B'ing, only unscientific full-song listening to a variety of material spanning fifty years of recording, and cycled each set of speakers through twice. There were notes like this regarding the *SM9* from the other engineers: "clarity in all frequency ranges," "visible ultra-harmonics," and "low end holds together in ways it doesn't normally — subs are sick!" Out of the five of us doing the listening test, four of us chose the *SM9* as our favorite model, and the fifth person had it as second favorite. I kept the prices a secret, since I wanted to eliminate that as a factor as much as possible, although I'm sure size was a clue. Only one other speaker was in the *SM9*'s price range, and the other two were significantly cheaper, but it was clear to everyone in the room which speaker was going to stay in place after the tests.

Even though I enjoy using the *SM9* immensely, it is not without one small fault; the buttons for putting it into standby, engaging the EQ controls, and turning off the subwoofer (and passive radiator) are on the inner sides of the enclosure if positioned as recommended (subwoofer on the outside). So if you have a pair of these speakers set up on stands behind a console (as we do behind our Neve 5316, which is relatively deep), the buttons are quite hard to reach, since they are pivoted away from you. Frankly, I would utilize the Focus feature a lot more with easier access to it. I would love to see these controls on a remote, considering the size (and cost) of the speaker. But really, that's the only thing I have come across in over six months of use that I don't absolutely love about the Focal *SM9*.

Visually, the *SM9* is impressive and sleek. It's built incredibly solidly and weighs 75 lbs, so you will want to either buy a very substantial set of monitor stands, or build your own heavy-duty ones, like we did. Following instructions we found online, we constructed extremely stable, tripod monitor stands out of 2" PVC pipe, playground sand (laboriously dried by hand with a heatgun by our intern Dylan Guidry — thanks Dylan!), and 1" boards sandwiched together — all spray-painted black to match the *SM9*. We built them to be the perfect dimensions both for the speakers and for the height of our desk. They are dense enough to keep their resonance to an absolute minimum, especially utilizing Primacoustic Recoil Stabilizers (two 10.5" × 13" pads per speaker) [*Tape Op* #62]. The cost of the homemade stands was negligible, which made us feel better about spending $400 on the Stabilizers. But for a pair of $7,600 speakers, you want to make sure you're doing everything you can to hear the speaker itself without mechanical resonance or movement. Yep, you saw that price tag right, and that will certainly make some of you dear readers grimace and flip quickly to the next review, and understandably so. At this price point, it is not a decision one makes lightly, and our lack of hesitation to shell out that kind of dough on our "windows to our mixes" speaks volumes about the quality of these speakers. I am definitely not of the mindset that you really need $7,600 speakers to make good recordings and solid mixes, but I will hereby report that I am finding that it helps.

($3795 street each; www.focalprofessional.com)

–Eli Crews <www.elicrews.com>

PreSonus
AudioBox iTwo Studio & AudioBox Stereo bundles

I'm always impressed with the sheer value that PreSonus is able to offer across its product lines. For decades, the company has been selling affordable audio gear and software, while still managing to include features and deliver sound quality beyond what you'd expect for the price. Take for example *AudioBox iTwo Studio*. This "recording kit" includes an *AudioBox iTwo* USB 2×2 interface, an *HD7* semi-open headphone, an *M7* side-address mic, cables, and a license for *Studio One Artist* DAW software — at a street price of $260. If you're a singer-songwriter recording at home or a musician who wants to put down ideas while on the road, this is a great deal. Likewise, *AudioBox Stereo*, a similar kit which includes the previous-generation *AudioBox USB* interface, an *HD7* headphone, a pair of *SD7* pencil condenser mics with shockmounts, a stereo mic'ing bar, and *Studio One Artist* — all for $250 — is a great starter pack for stereo recording of instruments, bands, choirs, ensembles, and stage productions.

Let's start with the *AudioBox iTwo*. First of all, it looks great. Its brushed aluminum chassis wraps around from top to bottom, book-end style. Two combo jacks on the front operate as XLR mic or 1/4" instrument/line–level inputs. Two input gain knobs are next to switches that choose instrument or line–level for the 1/4" inputs. A single 48V switch toggles phantom power for the XLR inputs, and a 1/4" headphone jack is paired with a volume pot. Importantly, there's also a mix knob for analog zero-latency monitoring; it allows you to vary the relative levels of what you're recording versus what's being played back from software. Therefore, there's no need to worry about driver latency, software buffer size, or a separate virtual monitor mixer. A big knob in front controls the volume of the main output, accessible via a stereo pair of balanced 1/4" TRS line-level jacks in back. Also in back are MIDI I/O and two USB ports. One USB port is for connecting the *AudioBox iTwo* to a host computer, while the second is for connecting to an iOS device. Interestingly, because the *AudioBox iTwo* is bus-powered, it requires USB power supplied to the first port when you're using it with an iOS device connected to the second port. An iPad power supply or an external USB battery unit works fine for this purpose.

Due to the *AudioBox iTwo* being Core Audio class-compliant, installation on Mac OS or iOS is a no-brainer. Plug it in, and you're good to go. On Windows, driver installation is required, as you would expect. Setting up the driver is explained clearly in the user manual. After installation on my Windows 8.1 Pro laptop, I chose to run the *AudioBox iTwo* as an ASIO device in Cubase [*Tape Op* #90], which required tweaking a few Device, VST, and MIDI setup panels. Later, I installed and ran the bundled *Studio One Artist* software. On its opening screen, *Studio One* confirmed that the *AudioBox iTwo* was connected, even displaying clickable links into the Audio Device and External Devices panels. Seconds later, I was up and running. Clearly, PreSonus thought out the hardware/software integration here, because this process was seamless and void of needless noodling within I/O configuration options.

Upon first use of the *AudioBox iTwo*, I was pleasantly surprised with its sound. The Class A mic preamps sound clean and they have plenty of gain for most dynamic and condenser mics, including the bundled M7. (On the other hand, there isn't enough gain to record a quiet source with a passive ribbon mic, but I wouldn't consider that a typical use case of

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Gear Geeking w/ Andy...

I've mentioned before that one of the goals of *Tape Op* Gear Reviews is to educate our readers. In other words, we strive to publish reviews that are just as informative about recording techniques as they are about the specific products being reviewed. Therefore, even if you're not considering a purchase of any products on these pages, I'm hoping that you'll find value in reading what's here. I know that I learn something new from the reviews in each issue, and I love turning that knowledge into action. For example, in the **Flexiguy FG500 mic preamp** review [*Tape Op* #105], Mavericks Studio owner Bobby Lurie opined about how recording through a single preamp type simplifies the workflow, puts greater focus on the music (instead of the gear), and importantly, results in a more cohesive sound across instruments. Subsequently, the mix comes together naturally, with less bus compression and processing needed to "glue" it together. Over the years, I've built up a decent collection of outboard mic preamps, but for my most recent recording project, I decided to take Bobby's advice and go with one preamp type as much as I could. More specifically, I went with one preamp "family" — six vintage **API 312** preamp cards that were modified and racked by Brent Averill Enterprises (now BAE) and five (non-vintage) **BAE 312A** preamps [#45]. The artist and I also decided to record the band live, including the lead vocal. (Because I was recording more than eleven channels at once, I had to supplement the BAE preamps with a few channels of Hamptone and Neve.) Quoting Bobby from his review, the various tracks ended up summing in a "seamless" way and sounded "wonderfully glued together." The final mixes were the easiest to complete in recent memory, and my notes from that session fill less than one page — a testament to how little processing was used in both tracking and mixing. I'm not saying you should sell off your racks of outboard preamps, but if it's been a while since you last recorded through a single preamp model (or through the preamps in your console), consider giving it a try, because you might have as much fun as I had. In the meantime, check out **Hayley Thompson-King** <www.hayleythompsonking.com> to hear songs from this session. In particular, watch the video of Hayley and her band performing "Dopesick" <goo.gl/bnsz4Q>, which was recorded at 11 AM (first take of the day) with no overdubs other than background vocals and second guitar. ••• On a related note, I learned something else while recording Hayley's band: Flipping the polarity of the headphone feed can significantly change what the musicians are hearing in the tracking room. For many of you, that's a "No Duh" statement. For me, it's "Doh!" Sure, like every other engineer who's recorded more than two mics at once, checking polarity at the mic preamp is something I do naturally during soundcheck. (And actually, I get even geekier. I adjust timing (phase) between mics, especially when tracking and mixing drums, but that's a whole other topic.) But I never thought to flip the polarity of the signal going to the artists' headphones! Try it yourself — put on headphones and monitor yourself singing into a mic. Flip the polarity anywhere in the signal chain. You'll hear a dramatic difference. What you're experiencing is the sound going through the air (and through your bones) combining with the sound emanating from the headphone drivers, with different amounts of delay — yup, phase cancellations. This makes me wish that personal headphone mixers had a polarity button on each fader channel, so that artists could optimize their headphone mixes accordingly. For now, I'll have to remember to test the polarity of the singers' mics during soundcheck while they're setting up their headphone mixes. *–AH*

this bundle.) The instrument-level DI has an input impedance of 1 MΩ (standard for an active DI), and it works fine with magnetic pickups. (I didn't have any instruments with piezo pickups that I was able to record with the *AudioBox iTwo*.) I didn't spend as much time with the *AudioBox USB*, which is housed in the "classic" PreSonus chassis, but it has a similar feature set to the *iTwo*, minus the iOS compatibility and line-level inputs. Its mic preamps (also Class A) have more gain than the *iTwo*'s, but they also have more self-noise, so signal-to-noise ratio is about even between the two preamp implementations.

I was also impressed with the bundled *M7* condenser mic. Although it's made to look like a large-diaphragm mic, if you shine a bright light into its basket, you can see the outline of its 0.5" small-diaphragm capsule. The mic is voiced for close-in work, and its multilayer screen does a good job of avoiding pops, as long as the mic isn't being hit with a direct blast of air. If you have good singing technique, you could use this mic without a pop filter. Frequency response is fairly flat, with a slight emphasis in the midrange between 900 Hz and 4 kHz, and the highs aren't hyped like they are on countless other low-cost condenser mics. There is a tiny bit of extra "air" from 13 kHz and above. The *M7* does exhibit some stridency in the form of harmonic distortion at 7 kHz when presented with high SPL, but when recording low to medium volumes, the *M7* sounds refreshingly neutral. I especially like how female vocals sound through the *M7* — lots of midrange presence, very little sibilance. You do have to be careful positioning the mic, as its sweet spot is quite small. For most sources, including voice, it sounds fullest 3"–8" out, on-axis within 30° of center. At greater distances, the low-frequency response thins quickly below 200 Hz — helpful for recording a boomy acoustic guitar but not ideal for tracking a drum kit sans close mics. Conversely, at distances less than 3", proximity effect ramps up steeply.

The *SD7* pencil condenser mics bundled in the *AudioBox Stereo* kit are also impressive for their cost. These come with screw-on cardioid capsules, but I wasn't able to look into the tiny holes of the 0.5" diameter front screen to determine actual diaphragm size. Because it's voiced for distance mic'ing, the *SD7* has less low-frequency roll-off than the *M7* does at distances greater than 8", and it exhibits a few dB of high-frequency lift in the range of 5–15 kHz. The included shockmounts are actually pretty nice. They're lightweight (being plastic), but they keep their set angle without slipping, and a bunch of extra rubber elastics are included. The stereo bar is quite heavy, as it's made of steel and brass. Together with the shockmounts, the stereo bar can be set up for X-Y or NOS recording (but not ORTF). The substantial thumbscrew bolts that hold the shockmounts can be threaded out of the bar for reinsertion from the other side of the bar, giving you more mic placement options. I tried the *SD7* mics on acoustic guitar, drum overheads, background vocals, and room ambience. The extra crispness up top means the *SD7* sounds more "modern" to my ears than the *M7*, but it's easily subdued with subtractive EQ.

The bundled *HD7* headphone looks like the Superlux HD681, a low-cost clone of the venerable AKG K 240 line. Like its inspiration, the *HD7* has a self-adjusting headband and earcups that are semi-open. Comfort-wise, I could wear the *HD7* all day. Given the *HD7*'s semi-open design, I was surprised to hear commendable low-frequency extension. With the *HD7* donned, my ears can make out fundamental tones down to 22 Hz, with low-order harmonic distortion becoming audible from 32 Hz on down. Thankfully, the lows aren't exaggerated like they are on headphones from lifestyle brands. On the other end of the spectrum, I hear a dip at 3–4.5 kHz, followed by a peak that extends to 8 kHz. At higher volumes, this peak is augmented by harmonic distortion, highlighting any sibilance that might be in the playback. I wouldn't rush out to buy this headphone on its own, but it's a great addition to each of these bundles. The *HD7* feels and sounds expansive, and importantly, you can trust it to honestly convey the low-frequencies in your mix, in a way that small speakers in an untreated bedroom or practice space can't.

Admittedly, I didn't spend a lot of time using *Studio One Artist*, but I am amazed with how far this DAW has come since its initial release [*Tape Op* #76]. Thankfully, it's remained free of the bloatware that plagues the more established systems, so it still feels streamlined despite it being rich in features. Nice workflow enhancements, like a truly useful right-click contextual menu that not only "learns" your most recent actions, but also allows you to edit event names, tempos, tuning, and other parameters directly in the menu, make *Studio One* a pleasure to use. Many plug-ins, virtual instruments, and sound libraries are included. Check out the PreSonus website for details. On a related front, I did try the wireless-transfer feature from iPad. After you record in the free *Capture Duo* app (or the full-featured *Capture*), you can transfer your audio files to *Studio One* via WiFi — very cool! I also tried the *AudioBox iTwo* with WaveMachineLabs Auria [#92] on my iPad, and it worked flawlessly. Note that the zero-latency Mix knob on the *AudioBox iTwo* is pretty much a necessity when overdubbing on an iPad.

One last thing I want to point out — both *AudioBox* manuals are very well written, and not only do they each include a "Quick Start" section for *Studio One Artist*, but they also include a whole chapter of tutorials covering mic'ing technique, dynamics processing, and EQ'ing. A ton of useful information is presented, and it's definitely worth reading. Overall, PreSonus has done a wonderful job of integrating several products with high value for money into the affordable and easy to use *AudioBox iTwo Studio* and *AudioBox USB* recording kits.

($259.95 and $249.95 street; www.presonus.com) –AH

RG-Recording

Cassette tape splicing block

Many of us remember the days of cassette-based 4-track machines. Likewise, we can recall the days of receiving mixtapes on actual cassette tapes, not over email. I have found that the problem with playing old cassettes is usually mechanical in nature — crinkled tape, broken tape, or the worst-case scenario, a deck eats the tape. Consumers would assume there is no hope, but audio engineers are not afraid to roll up our sleeves and attack the problem. Removing the tape reels and transplanting them into a new shell often solves the issue. However, if the tape is torn or damaged, a splice may be in order. Trying to splice 1/8" tape on a 1/2" block is not the way to live. Here is where the RG-Recording splicing block comes in. The block is made of hardwood (oak, ash, or birch), has a channel sized specifically for cassette tape, and comes standard with 90° and 45° cut guides. RG-Recording can even make a block with custom cut angles for no additional cost. The block is about 7" long and has a non-slip rubber base to keep things steady. Given its price, it's a no brainer.

($12.99 w/ free shipping; stores.ebay.com/rg-recording)
–Garrett Haines <www.treelady.com>

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Sonarworks

Reference 3 speaker & headphone calibration software

It goes without saying that our studio monitors, be they speakers or headphones, influence the sound of our productions more than any other processor, instrument, or device. Recently, the technologies of speaker system design and advanced materials have allowed manufacturers to produce accurate monitors at every price-point. Once our speakers have been chosen and the room has been acoustically treated, we listen, adjust, tune, equalize, re-treat, and repeat until our mixes translate (reasonably) well to other playback environments. For most of us, however, there are practical limitations to the type of monitors we use and the amount of room treatment we can employ. Even in world-class rooms with the most esoteric systems, providing the most accurate playback requires lots of expertise and time spent adjusting speaker placement, acoustic treatments, and possibly room EQ. Sonarworks, a new company from Latvia, has developed a product to simplify the process of bringing our playback system to the highest level of accurate playback. *Reference 3* allows novice and seasoned users to calibrate their speakers and headphones.

Sonarworks Reference 3 software consists of three components: speaker and room calibration software; speaker correction plug-in; and headphone correction plug-in. Additionally, the full kit includes a Sonarworks calibrated measurement mic. The software walks the user through room analysis and then generates a unique plug-in preset which provides correction for the specific monitors as measured in the room. The headphone plug-in will correct the response of many commonly used professional headphones. Headphones with individually measured response curves may be purchased directly from Sonarworks. Each headphone comes with a unique plug-in preset which matches that exact headphone. Alternatively, an "average" curve, derived from many sets of a particular model of headphone, may be applied to match your existing headphone.

In use, the *Reference 3* software (Windows and Mac OS) walks the user through the steps of setting up the reference mic, choosing appropriate monitor levels, and measuring the room response at 24 positions around the main listening area. The software produces bursts of clicks, which are used to calculate the mic's exact position in the room, along with short frequency sweeps, which measure the perceived acoustic power, or apparent frequency response of the speaker in that room. After completing the measurement steps, which takes 5 to 10 minutes, the software generates a user-named plug-in preset. The *Reference 3* plug-in running in your DAW uses this preset to correct the frequency response of your speakers. Multiple plug-in presets may be generated, for different monitor sets or listening positions. The plug-in itself, which is available for AU, VST, RTAS, and AAX Native hosts, lives on the master monitor fader in the DAW. This may be a bit confusing at first — you don't want to bounce your mix through the *Reference 3* plug-in; you just want to listen through it to hear your corrected room response. Each DAW provides a different method for routing master and monitor faders, so you'll have to determine the best way to instantiate the plug-in for your system. Alternatively, the plug-in may simply be bypassed during the final bounce to avoid printing the mix with your room EQ. I found a simple routing setup in Pro Tools to insert the *Reference 3* plug-in on my monitor output while my mix bus (bounce path) remains unaffected by the plug-in. Unfortunately, other sources besides your DAW, like an MP3 player or iTunes, will not be heard through the room correction software — that would require an external processor (running Sonarworks) between your monitor controller and speakers (hint-hint, Sonarworks!), or routing your playback device through your DAW.

Sonarworks Reference 3 software separates itself from other speaker calibration software in a few different ways. First, Sonarworks provides linear-phase equalization, which minimizes the phase distortion and artifacts that traditional equalizers produce. Second, Sonarworks provides up to 16,000 EQ points for precise correction. Third, the software lets the user adjust or "voice" the EQ preset to tailor settings for a desired response. This voicing may simply turn up the bass (or treble) because you enjoy it, or you can apply stock presets which allow your measured speakers to emulate the frequency response of some other typical studio monitors. This, to a large degree, lets you hear what your mix may sound like on other common speakers. These same principles apply to the headphone plug-in, except that users aren't able to measure their own headphones — Sonarworks can do this for you, or you can choose a typical response curve for your specific model of headphone.

I have spent many hours tuning my room with real-time analyzers and room analysis software, adjusting my crossovers, subwoofer level, speaker positioning, and acoustic treatment until I was most satisfied with my monitor system. I typically try to avoid room equalizers, except for gently tweaking soffit-mounted main speakers in well-tuned rooms. After installing *Sonarworks Reference 3*, I can honestly say that my room sounds noticeably more accurate. I hear better stereo separation and imaging, smoother overall frequency response (especially at the subwoofer crossover point), and even some additional perceived height information. My long-time clients have commented on how well my recent mixes have translated to their playback systems, which bolsters my confidence during mixing and mastering. I believe it is important to fix acoustic and electronic issues in the room and playback system first; and then adding *Sonarworks Reference 3* provides a very noticeable improvement in an already well-balanced system. I have applied *Reference 3* to more than five different rooms, including low, mid, and high-priced monitor systems, and every room's accuracy benefitted from the software. After several weeks of using the plug-in on my own mixes and masters, I can't find any noticeable artifacts or limitations that I have previously found in other software-based room correction.

As with any plug-in, latency is possible, and Sonarworks provides three modes of accuracy versus latency. The lowest latency setting produces a monitoring delay of around 1 ms, while the most accurate setting produces about 60 ms of delay. Obviously, for tracking and programming, the lowest latency setting is essential, but for mixing and mastering, the greater latency is not a problem. I found all three latency/accuracy settings provided similar improvement to the sound of my monitors, though each of the three settings produces its own extremely subtle tradeoff between frequency and phase–response accuracy.

The *Reference 3* plug-in provides many useful features, like graphs of the before and after frequency responses, the correction curve, as well as more esoteric information like dynamic range limits and phase-response graphs for those who desire to view such information. This effective plug-in may be used in a very simple plug-and-play way, or you can delve under the hood for finer tweaking, experimentation, and useful feedback about your system. A professional room tuning can cost hundreds up to thousands of dollars, while this simple-to-use tool from Sonarworks brings a high level of accuracy to your room for much less. I highly recommend this software as a finishing touch in any room. *($49–$299 direct; www.sonarworks.com)*

–Adam Kagan <adamkagan@mac.com>

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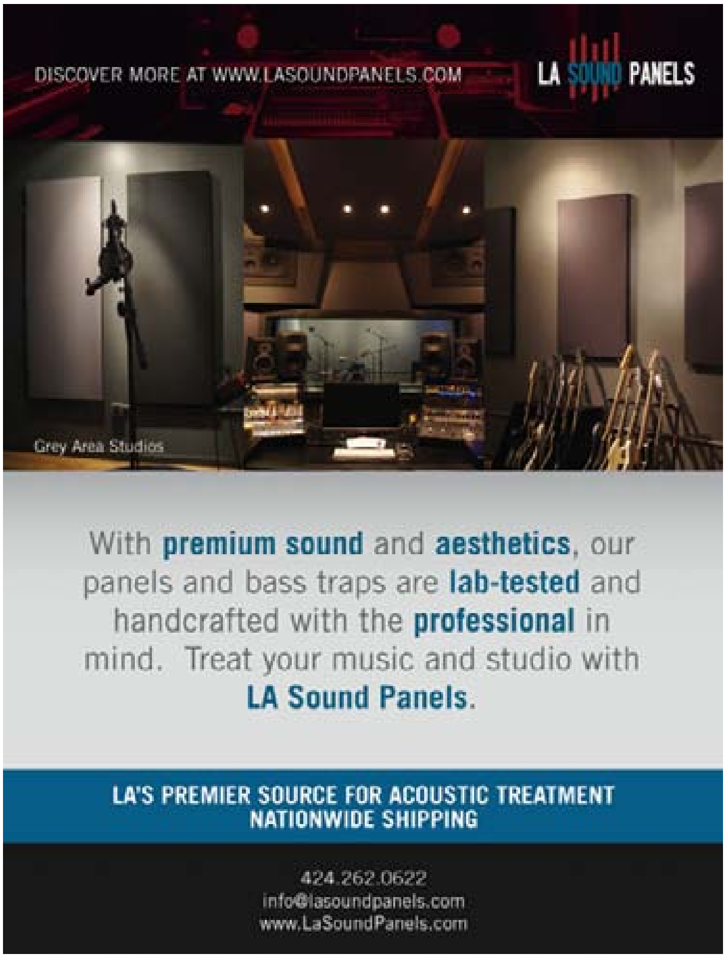
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Amphion Loudspeakers

Two18 studio monitors

A few months ago, I took on a project that pushed me into the dreaded realm of needing new studio monitors. After about ten years of using a pair of Behringer Truth B2031A passive monitors (don't turn up your nose or ears until you try them; they blew away all similarly-sized speakers I tried until now), I began the process of remastering Marcel Dupré's pipe organ recordings for Mercury Living Presence and Philips. I needed reliable deep bass from my monitors, and the Behringers couldn't go low enough.

I briefly tried adding a subwoofer, but it didn't work in my room, and I also didn't like the centered bass, since different bass frequencies sound different in different parts of the stereo sound-picture of a highly reverberant space like Paris's Church of Saint-Sulpice and New York's St. Thomas Church, two of the three locations where these recordings were made. In short, this kind of bass doesn't center itself, and it sounds strange when all bass frequencies come from the same center channel below the desk.

After reading Adam Monk's positive review of the Amphion One15 and One18 [*Tape Op* #105], I asked to demo a pair of the company's largest studio monitors, the *Two18*. "Largest" is a relative term, since this model is still only 21.7″ × 7.5″ × 12.4″ in size. It utilizes a waveguide-mounted 1″ titanium dome tweeter and two 6.5″ SEAS aluminum woofers in a sealed cabinet with twin bass resonators on the rear panel. Amphion claims usable frequency response of 39 Hz – 20 kHz (±3dB). My experience is that the usable bass extends a bit further down, far lower than I've heard before from speakers this size.

As Adam reported in his review, part of the pleasure of the Amphion experience is dealing with founder/owner Anssi Hyvönen, who has recently started working with U.S. rep Dave Bryce. The loaner pair was shipped to me from a major studio that shall remain nameless, where there had apparently been experiments in maximum viable sound pressure levels. Net-net, one of the woofers was blown and another one had some "hair on the edge" in some playback situations. Dave quickly sent me replacement drivers, and the repair was as easy as undoing a half-dozen Torx screws, putting the right wires on the clearly marked speaker terminals, re-seating the driver, and tightening the screws. Voila, good as new.

Anssi also gave me good advice about where to position the speakers in my somewhat cramped studio space. It ended up that placing them about a foot out from the walls, and moving my fiberglass panels around a bit, yielded the fewest and least annoying upper-bass nodes while preserving plenty of deep bass level, without producing an artificially dark sound quality. I verified this through several hours-long listening sessions — a treat since I got to hear some of my favorite music with new clarity and punch. Especially enjoyable were my favorite LPs; without the typical harsh studio monitor midrange, vintage vinyl sounded all the more sweet.

From the get-go, I liked the highly focused but non-screaming midrange and the accurate treble (meaning it would be described as "reserved" compared to most other studio monitors, which I find over-harsh on top). I was surprised how much the mid and top resembled the underrated Behringers, but with more evenness and less "nasal" sound qualities (this is high praise because one of the things I like about the Behringers is the absence of "honking" midrange that is so typical of nearfield monitors, especially the self-amplified kind).

Once I got down to business mixing the 3-track Dupré tapes to stereo masters, I immediately noticed that I could work at lower SPLs than my old system required, and that anything I did as far as channel mix on the Amphions translated very well to my big B&W 808 speakers upstairs and also to my Sennheiser HD 650 headphones [*Tape Op* #43]. So, the stereo image and frequency spectrum decisions I made with the Amphions worked on big speakers in a big room and on high-fidelity headphones. The Amphions were especially helpful in sorting out how to balance musical detail against room reverberation, and how best to spotlight Dupré's quick and complex playing. For each mix I turned in, the client feedback was very positive, and the client and I both agreed that the remasters compare very favorably to the original LP issues. Working on the Amphions, I didn't find myself regretting decisions or going back and listening again and again because I was doubtful of what I was hearing. These speakers speak the truth!

I asked Anssi to explain his design philosophy leading to his company's series of professional monitors. First and foremost, he said, "We never wanted to make yet another box. The goal was to come up with something which would hopefully indirectly contribute into putting emotion back to music again."

He explained how he accomplished that goal: "Speaker building is always a balancing act. The larger the driver, the more it can move air. The larger it is, the slower it gets. The trick is to keep the drivers fast, but still come up with reasonable ability to move air. A passive radiator helps in this respect, but that is not all it does. One of the nicest additional benefits of using a closed-type construction is to be able to better control what happens inside the box in terms of air flow and pressure changes, which increases the midrange resolution by allowing the active drivers to work better."

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In a nearfield situation, working at reasonable SPLs, I think these speakers can reliably tell an engineer about everything but the very bottom octave. The difference in the bass of a pipe organ between the Amphions and the big B&Ws is that the Amphions produced the sound of the low note attacks, but the B&Ws moved the floor when the bass pipes really let go. I don't think you need the floor shaking to make reasonable mix and EQ decisions, but that's open to disagreement. With a fast run down the organ console, where the frequencies quickly drop, I hear each descending note sound distinctly through the Amphions, whereas the B&Ws sound more like a downward-sliding tone, likely because the woofers can't piston-fire as quickly as the notes are sounding. Translating this to a modern pop or sound-for-picture mixing situation, you'll get fast bass transients out of the Amphions, but not enough bass energy to make the walls and floor shake.

Taking breaks here and there from the remastering and other studio work, I listened to CDs of a few 2014 Grammy winners, the kind of great music that doesn't get featured on the evening telecast. I wanted to hear a variety of styles and production techniques through these speakers. My listening notes follow.

St. Vincent – *St. Vincent* (Best Alternative Rock Album): The *Two18*s brought out the jagged, "pointy" qualities of the music and sound, and the surprisingly wide and crisp dynamics. Also clear, beyond and above all the interesting sounds and textures, was Annie Clark's very fine voice. The album's overall sound quality was loud but not overwhelming, and many interesting sounds, riffs, and hooks emerged from the dense mix. This album is worth hearing on good speakers or good headphones, preferably from a real CD or high-resolution file, rather than a lossy stream, because it's an ear treat.

Chris Thile and Edgar Meyer – *Bass & Mandolin* (Best Contemporary Instrumental Album): The Amphions offered very clear definition of space and placement, and great detail of both musicians' fingering, bowing, and picking. It was surprising how much varied sound two acoustic instruments can make, in the right hands. The mix had a "3D" feeling, with clear width, depth, and height. It also sounded balanced and detailed across the room.

St. Louis Symphony, David Robertson (conductor) – *John Adams: City Noir* (Best Orchestral Performance): Typical of modern symphony recordings, the perspective is somewhat distant and crowded (congested) when many instruments are playing together, but the speakers did a good job of voicing individual instruments and maintaining the stereo spread available in the recording. This is dark and moody music, as the title suggests, but the recording is somewhat bass-shy. Some solo parts seem to float above the orchestra, which is a very interesting sonic effect.

Hilary Hahn with Corey Smythe – *In 27 Pieces: The Hilary Hahn Encores* (Best Chamber Music / Small Ensemble Performance): The Amphions excelled in bringing out the details of Hahn's violin and Smythe's piano. The recording has a very close, produced quality, but is not harsh. Rather, those superb violin details are sometimes too much (as when we can hear the horse hair on the bow making high-pitched resonances). The fact that the "too much" is audible is a credit to the speakers, because professional monitors must tell all, the good and the bad.

Chick Corea Trio (Corea, piano; Christian McBride, bass; Brian Blade, drums) – *Trilogy* (Best Jazz Instrumental Album): A crisp and detailed recording like this spotlights the even frequency response and quick dynamics of the Amphions. The instrumental balance is maintained no matter how loud or soft the ensemble plays. Despite the close mic'ing, nothing is overly bright or boomy. The speakers brought out the precision of the playing and the careful choice of notes during improvised solos.

Gordon Goodwin's Big Phat Band – *Life in the Bubble* (Best Large Jazz Ensemble Album): Through the *Two18*s, this recording was very peppy and driving, but never annoying. The music and players offered plenty of texture and dynamics, and there was also very nice stereophony. Although it's totally different music, this album brought out the same good things in the speakers as St. Vincent's album — very fast response to percussive and dynamics shifts, and wide and even frequency response that allowed me to hear all the details in complex music and mixes.

After a couple months with the Amphions, I'm not letting them go. I believe they have brought my monitoring environment to a higher level of precision, and the fact that I can work at lower SPLs will prolong my audio career and music-listening enjoyment. They are one of the few studio monitors of any size that I have encountered that both sound accurate and are a pleasure to hear.

($3000 each; www.amphion.fi)

–Tom Fine <tom.fine@gmail.com>

Boz Digital Labs
+10db Compressor & EQ plug-in bundle

I'm a fan of compressors that bring something to the table other than transparency, so I've had an eye on the legendary ADR F769X-R Vocal Stressor, an early outboard "channel strip" incorporating the Compex compressor alongside an EQ, for quite some time. When Boz Digital Labs announced that they were modeling "a very highly sought-after compressor hardware unit" that looked identical to the ADR, I jumped at the chance to try it out. The Compex can deliver incredibly aggressive compression; I like to think that it's one of the kings of "punchy" and/or "raw" sounding compression, in that it wrangles whatever you throw at it into submission with great ease, and you are well aware that it's happening. For instance, it's perhaps most famously tied to John Bonham's drum sound in Led Zeppelins' "When the Levee Breaks." On the other hand, the ratio knob offers more subtle ratios of 1:1, 2:1, and 3:1 (in addition to 5:1 and 10:1), so it's capable of being inconspicuous if it needs to be. The two other main elements of the Vocal Stressor are the 4-band parametric EQ and the expander; the EQ allows you to boost and cut the same frequencies as a Pultec, while the expander is unique in its ability to operate at the same time as the compressor. Additionally, in side-chain mode, the EQ allows you to compress a drum bus without the kick pumping the gain reduction, or you can use the compressor as a colorful de-esser (formerly pre-emphasis mode on Compex compressors). At the end of the chain is a limiter that is fixed at 100:1, harking back to its intended use in broadcasting.

The *+10db* plug-in bundle faithfully models the Vocal Stressor in all of its glory, and includes ten presets by multi-platinum producer David Bendeth (Underoath, Paramore, Breaking Benjamin) to help you find your footing. Some find the EQ operation to be odd relative to modern EQs, as you have to toggle whether you are cutting or boosting, and then adjust the dB pot, similar to a Manley Massive Passive, but I wouldn't let that bug you. Another thing to keep track of is gain-staging, as you have control over the input gain, the expander threshold, the compressor threshold, the output level, and a wet/dry knob — without the aid of a fancy GUI and lots of meters. But again, it's all part of the charm of this legend — not a big deal.

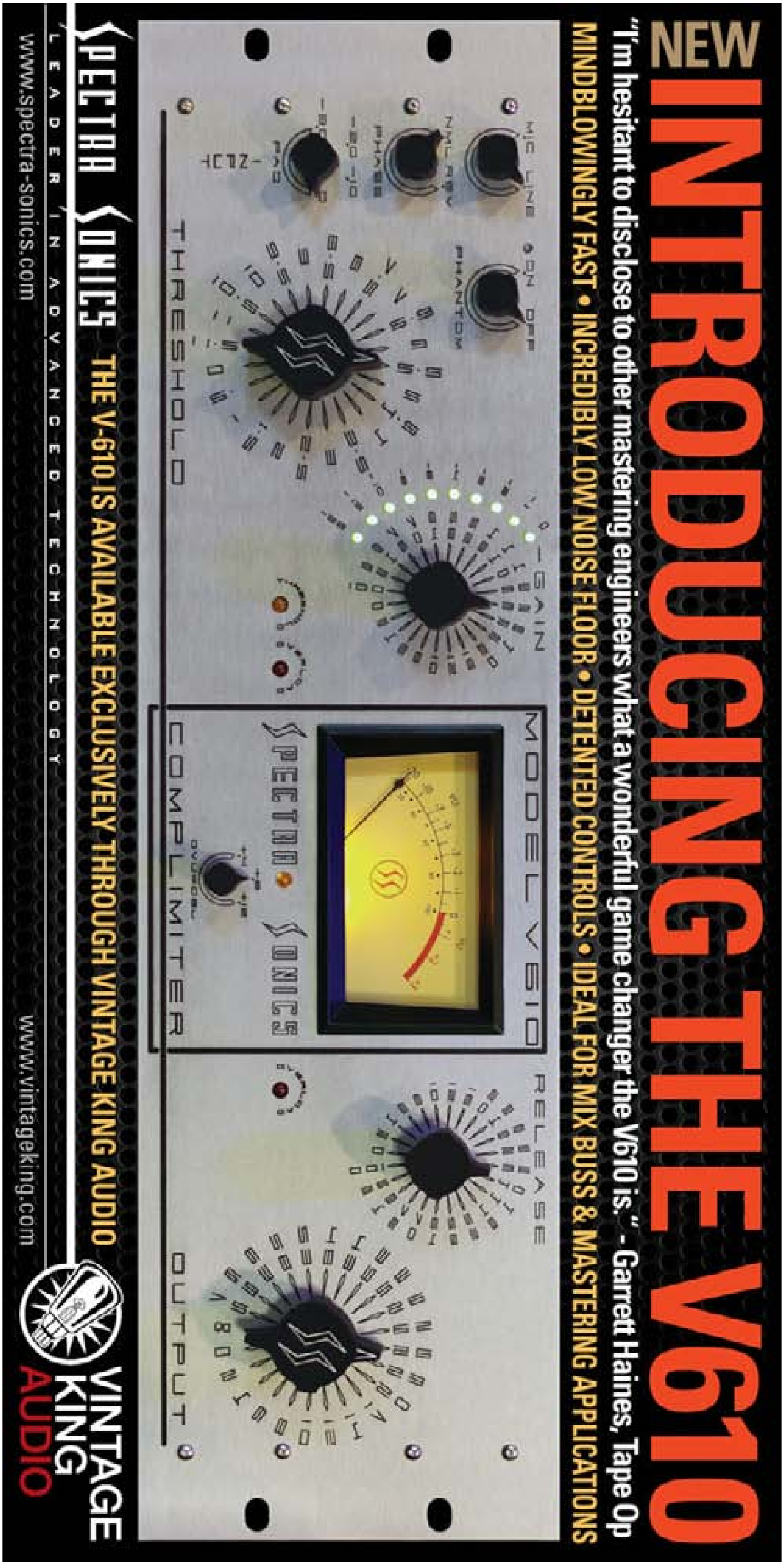
In use, my main objectives were to thoroughly test the things for which the Compex is most recognized — compressing kick drums, room mics, drum buses, and for "stressing" vocals. You can probably guess that I loved it, but keep reading anyway. I'd also been looking for more tension in my mixes — a way to better create a feeling of urgency and struggle — and *+10db* delivered in each of these scenarios. The most straightforward way of describing its style of compression is that whatever you process sounds more intense. Sometimes with other more transparent compressors, you work hard to find a sweet spot that creates a sense of intensity, but it's really easy to do with the *+10db* and its soft-knee curve. And what is perhaps most powerful, even more so than in the original unit, is the ability to take something right to the edge of nastiness and then dial in some dry signal with the wet/dry knob. Another fun and rather obvious use is utilizing the compressor as a killer de-esser, and then instantiating another *+10db* plug-in to do your compression. You can both have your cake and de-ess it too. Grab the free demo, and see if it's right for you. *(Compressor $99, EQ $99, both $199; www.bozdigitallabs.com)*

–Dave Hidek <dave@treelady.com>

Aviom
A320 & A360 Personal Mixer
AN-16/i v.2 Input Module
A-16D A-Net Distributor

It has always been a priority for me to provide the ability for each member of a band to dial in her or his own headphone mix during a tracking or overdub session. It was essential to me when building my first studio nearly 15 years ago, and the long-discontinued Oz Audio Q-Mix HM-6 boxes [*Tape Op* #37] gave me that functionality in a very affordable, robust package. When setting out to help producer, musician — and now studio owner — Shahzad Ismaily put together Figure 8 Recording in Brooklyn this past year, I started doing research about which modern cue system we could employ that would give us the ultimate monitoring flexibility for a two-room facility (that sometimes operates as a single space), at a reasonable price point. Shahzad had had good experiences as a studio musician using Aviom's 16-channel digital mixers, so I started there, and honestly never really looked back. While the upshot is that we are extremely happy with our system, it is not without a couple minor concerns, which I will describe below. But let's start at the beginning, with installation.

When trying to decide between an analog and digital multichannel cue system, there is one factor that really makes it an unfair fight: cabling. For Figure 8, I calculated our cue line runs at around 1600 ft total for both rooms. That amount of high-quality (Gepco or equivalent) 16-pair analog cable would cost roughly $5,000, whereas that amount of shielded Cat 6A (Ethernet) cable was well under $500. (And that's for the good stuff — Aviom systems will work over unshielded



Cat 5e as well, but I wanted to future-proof our runs.) In addition to cost, there's also the question of conduit space. Gepco 22 AWG 16-pair is approaching an inch in diameter. You can get at least four Cat 6A cables in that same amount of conduit space, which was absolutely essential to us in the end, since it allowed us to put a couple spare lines through each in-wall run. A side benefit of running Cat 6A to every panel in our studio is that there are now convertor boxes for many other types of data to run over those lines. We're running USB, MIDI, and HDMI (for getting video to the recording spaces for film scores and the like) over the spare lines with great success, but we can allocate those jacks to the cue system with a simple crosspatch.

One of our main requirements for the cue system was that we could patch to any panel from either control room, so a percussionist downstairs could easily overdub onto a session running upstairs, for instance. This was achieved quite easily with a couple of simple, affordable Cable Matters RJ45 patch panels in the machine room, where all cue lines terminate. Via short Cat 6 jumpers, we can patch out of either of the two Aviom distribution systems into any of the cue jacks in the building. When deciding on a distributor, we chose Aviom's stripped-down half-rack *A-16D* over the more full-featured D800. At four times the price, the D800 offers a bunch of features that we figured we would never use, such as support for up to 64 channels and direct integration with digital mixing consoles. The *A-16D* functions like a simple PoE (Power over Ethernet) switch (as they're called in the networking world), and the only catch is that it doesn't provide its own power for each of the eight Ethernet lines; you need a separate 24 V wall wart for each line you want to drive. With eight extra wall warts, you still come in way under the price of a D800, and luckily, the power supplies can be attached to the back of the *A-16D* in whatever rack enclosure you have it in, so there is still a single Ethernet connection to each personal mixer which provides both power and audio, eliminating extra cabling and mess in the live room.

Speaking of the personal mixers, there are currently two options: the *A320* and *A360*. The A-16II mixer some of you may be familiar with (the clunky blue thing slightly resembling Boba Fett's face mask), has been discontinued, and frankly there are a few features I was surprised to see Aviom do away with. One is the ability to daisy-chain boxes; both the *A320* and *A360* require a "home run" directly to a distribution hub. The other, at least vis-à-vis the entry level *A320*, is knobs. The *A320* has only two knobs: a Master Volume and a rotary encoder. The latter determines either channel volume, channel stereo position, or master tone settings, depending on which button you press before turning it. Normally this kind of multifunction encoder drives me crazy, and I was a little disappointed to see it when the *A320*s first showed up. But now, after using the *A320* daily for a few months, I have come to embrace it, as it cleans up the face of the mixer significantly, and is actually less intimidating for non-techy musicians. I figured there would be a learning curve for musicians using the *A320* for the first time, and there is, but it's very short.

At the opposite end of the spectrum is the *A360*, which has a ton of extra features and functions, like built-in decent-sounding reverb, an internal mic for piping in some of the ambient sound of the room around you, tone controls for each channel in addition to the master controls, both mini and 1/4" headphone jacks, and an analog mono output of your mix for feeding a speaker or an amp. Plus, it's got more knobs, for people who aren't intimidated by them. Additionally, if the *A360* is connected to a D800, it can transmit its digital mix back to the D800, which is useful for driving wireless in-ear monitors from a central location. But I would say most of the features are more suited for large theater or live productions, and seem like overkill in a medium-sized studio setting. (Touring musicians take note: bringing an Aviom system around with you for driving your in-ears or wedges would absolutely slay; if I still toured extensively I would invest in such a thing in a heartbeat.)

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I haven't mentioned the *AN-16/i v.2* yet, but it's quite essential, being the "input module" which does the A/D conversion for the whole system. It has sixteen TRS inputs (and sixteen TRS thru jacks), which allow you to take outputs from your patchbay, recorder, or mixing console to feed the A-Net's 16-channel system. You can select an input level for each pair of inputs (four steps from –10 dBV thru +22 dBu), and you can also very conveniently gang each odd/even pair into a stereo input (with stereo width and balance control available at the personal mixer). Conversion happens at 24-bit, 48 kHz, and honestly sounds extremely good. We have had numerous musicians comment on how good they think the headphone system sounds, and how it really allows them to feel as though they are "inside" the music. Being a digital system, there is a very small amount of latency introduced, but even when coupled with the slight latency when monitoring through a Pro Tools HDX card, not a single person has complained about things sounding late or phasing with what they're hearing in the room, which I have experienced with other digital cue systems.

Of course, how good the system sounds depends largely on which headphones you use, and this brings us to my one real caveat about this system. I would not say that these are the quietest headphone amps on the planet, especially on the *A320* (both the *A360* and the discontinued A-16II mixers get louder, and sound slightly better, to my ears). There is a fair amount of self-noise in the onboard amplifier itself, which is present even with all input channels all the way down. The good news is twofold: one, since the feed is digital, the noise doesn't increase at all as you turn up each input channel (unless the signals themselves are noisy, of course); two, with the proper headphone choice, this noise falls below the threshold of annoyance for almost all types of music (and we work on a lot of very quiet music here at Figure 8). Low impedance headphones are the trick — our 38 Ω ATH-M50 headphones [*Tape Op* #63] work for more types of music than the 250 Ω Beyerdynamic DT 770 we have. The lower impedance gives your headphone amp more headroom, so the Master Volume can live around noon to 2 o'clock, where it sounds best. In any case, I don't plan on using the Aviom system for critical precision headphone monitoring, as that's not what it was designed for.

I have a few tips for getting the most out of this system. The first is to make your own Ethernet cables. Terminating twisted-pair data cable with RJ45 connectors is really easy, and the tools are dirt cheap and widely available. For the interconnect cabling from the wall panels to the mixers, I used Belden 1305A Multi-Conductor UpJacketed CatSnake cable, which is actually Cat 5e, and has a very nice "ruggedized" feel to it, almost like a mic cable. You will never ever want to use an unruly off-the-shelf Ethernet cable again once you've used this stuff. If you use the *A320* mixers, I would suggest terminating one "regular" RJ45 on the mixer end and one Neutrik etherCON connector on the end attaching to your panels, which should have their own female panel-mount etherCON jacks. If you use *A360* mixers, you can put etherCON connectors on both ends — even better! etherCON ain't cheap, but if you hate dealing with RJ45 connectors as much as I do (broken tabs, crappy strain relief, hard to disconnect, etc.), you'll be very happy you spent the dough. Thanks to the fact that networking supplies are incredibly cheap, you're saving so much money on the rest of the system, you should splurge a little on those connectors. Lastly, buy one Aviom stand adapter and cheapo mic stand for each mixer you have, and strap the Ethernet cable to the stand for extra strain relief. The *A320* especially is very light, so I much prefer it mounted on a stand instead of sitting on a tabletop. It's not built like a tank, but the tradeoff is that it's very portable (and affordable).

This is a really fantastic system that allows small-to-medium studios to get sixteen channels out to their performers via individual portable stations, taking the burden off the engineer to please everybody's monitoring desires — a frankly impossible task if you have only one stereo cue send. As for price, each of our rooms' five-box systems came in just above $4000, including cabling, which is a lot compared to my old Oz Audio Q-mix HM-6 system, but it's an excellent deal considering how much easier it makes the most quintessential component of performing well in the studio — hearing properly.

(A320 $399 street, A360 $799, AN-16/i v.2 $1195, A-16D $399; www.aviom.com) –Eli Crews <www.elicrews.com>

Soundizers

StereoMonoizer software

One of my biggest frustrations as a mixer is importing audio files from a client and finding 160 stereo audio tracks have been created in my DAW. Most of the time, I'll figure out that the bulk of the stereo tracks delivered to me do not actually contain any stereo information but are, in fact, dual-mono or simply single mono tracks which are slightly panned in the stereo field. Usually, I can cut the voice usage in half simply by splitting these supposed stereo tracks and keeping the mono data. Thanks to the folks at Soundizers and the release of their *StereoMonoizer* application, the process of analyzing and converting audio files has been streamlined and automated.

The standalone *StereoMonoizer* application (Mac OS and Windows) provides a large dropzone where you may drag and drop single audio files or a folder of audio files. Keep in mind that the files must be either WAV or AIFF format. The program goes about analyzing the files for stereo content, resulting in a file-by-file report (single channel mono; stereo file, mono content; stereo file, stereo content; stereo file, panned hard left; etc.) and a waveform display of how much "stereo-ness" ("stereocity"?) each file contains. After analysis, the user may choose if and how to convert each file. The program automatically chooses the option that would be most useful for each file, but sometimes you may want to keep the panning information for a given file, so you can override the default and choose not to convert that particular file. Global preferences are available to define how and where the new processed files are saved. For instance, *StereoMonoizer* will create a backup folder of the original files and then overwrite the files in your folder, or it can place the new files into a user-specified location. Other preferences include variable pan depth settings, identifying blank files, and even normalizing gain to a preset level. Unfortunately, Broadcast WAV files do not retain their timestamp after the stereo files have been processed, but this feature will be implemented in a future update.

This simple-to-use program presents lots of information about audio files and provides very useful processing options. Why didn't I think of this? Every music and post editor and mixer will find *StereoMonozier* extremely useful and informative. Best of all, *StereoMonoizer* only costs 49 bucks. A fully functional version of the software can be demoed for 14 days before purchasing.

($49 download; www.soundizers.com)
–Adam Kagan <about.me/adamkagan>

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AUDIO

Neumann

U 47 fet condenser mic (reissue)

The *U 47 fet* reissue from Neumann is a faithful recreation of the solid-state version of the U 47 manufactured from 1972 to 1986. It has a fixed cardioid polar pattern and employs a K 47 capsule and the same head grill and nickel finish as the original *U 47 fet*, as well as the classic Neumann badge on the front of the mic.

The original *U 47 fet* was intended to recreate the sound of the tube-based U 47 in a solid-state model, but initially, it was not nearly as popular a mic. Many did find, however, that it worked beautifully in front of a kick drum, an acoustic bass, and later electric bass cabinets. This is how I had come to know and love this mic's ability to capture the full picture of sources heavy in low-frequencies. Great studios around the world likely have at least one vintage *U 47 fet* in their mic locker, and with good reason.

Out of the box, the reissue product says quality. Even the outer cardboard packaging has a slick, faux leather finish. The mic comes in a cherry wood box with dense foam lining for a snug fit. A swivel mount is fixed on one side of the mic, and the XLR connector sits at the bottom of the mic body. On the back of the mic, there are switches for attenuation (–10 dB) and low-cut filter (40 or 140 Hz). On the bottom is an output level switch (–6 dB).

Recording bottom-heavy sources was my first stop. Not surprisingly, the *U 47 fet* delivered a big, fat truckload of goodness to kick drum, and it made life easy when planted in front of an Ampeg B-15 cabinet as well as my acoustic bass — full bodied and extended, with nice clarity on all of the above. It was easy and fuss-free to get to a great sound and tone. I looked around the studio to see what else I could throw at it. Timpani? Wurlitzer through the Kustom? Killing. This reissue delivered the sound this mic is "known" for. Like an Oxford-educated playboy Paul Bunyon in a Jil Sander suit — tough, articulate, well-toned, immaculately presented, and smooth. But what else can it do? $4000 is a lot of bread for mid-level studios and weekend warriors who need their mics to excel in more than one application.

In addition to recorded music, I do a fair amount of audio for film and video. The *U 47 fet* had just arrived, and although it's not the normal choice for a location recording, I thought I'd put it into service in a non-traditional way. We had to record an on-camera interview with a female subject. The mic had to be out of the shot, away from her mouth, and I needed some meat on the bone in terms of tone. It delivered a nicely balanced warm tone without any hype and was extremely quiet. It would make for an excellent broadcast mic (although you'd better be doing one hell of a podcast to justify the cost).

Each year I get roped into doing the audio, sound effects, and music for my daughter's school play. I typically end up recording many of the sound effects myself rather than downloading them from iTunes. The sound of "magic" was required for the stage production, and Hazel handed me her baby chime/rattle that she thought would fit the bill. She was right. This particular item sounds like fairies on helium — perfect. Now, let's get the mic that the world uses for massive meaty kick drums and record some magic fairies. Well, it captured this tone with incredible realism — very smooth, warm, and pleasing to the ear. Although not a "known" go-to for high-frequency and detail recording, I liked what the *U 47 fet* did to smooth the top end. For the sake of hearing it, I took a few whacks at some different cymbals. Again, I liked the warmth and smoothness.

Recording a strummed Gibson Dove flattop steel-string acoustic guitar with the *U 47 fet* pointed at the twelfth fret resulted in a balanced tone with a nice, mellow top end. What I really liked was that it didn't have a hyped sound that many modern mics impart. There was no clicky, plinky-plonky nonsense — just a nice representation of the instrument. It was robust, with nice full-bodied low end, without being boomy. The recorded track took EQ well to sculpt it into a dense mix, but sounded lovely left alone and paired with a vocal.

I expected this mic to sound solid on electric guitar, and it did not disappoint. I put it up in front of my Vox AC30, plugged in a Telecaster, and cranked it up. All the shimmer and shine was there, stewed gloriously with the grit and meat. What a treat — it sounded just like the amp in the room. I would go to this mic for guitars all the time. It captured a realism that is missing from the typical SM57 setup. The SM57 is absolutely a great choice in many situations, but the *U 47 fet* simply had more life. And, for the price tag, it should!

I also thought it was a great choice for female vocals that were a tad harsh or peaky. It smoothed them out and lessened the need for EQ and frequency-dependent compression. On male rock vocals, I found that its slight rise in the 2 kHz and up range helped it cut through nicely, but without any brashness. Because of its high SPL-handling capabilities, it was good in front of a loud rock vocal and didn't collapse like some condensers do when used in this way. As is the case across the board, you can find a combination of mic/preamp/singer that meets your needs. For male vocals, I liked this mic paired with a Daking (Trident A-Range style) preamp (*Tape Op #45 & #71*). The Daking complemented the

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mic's warm bottom end by adding some clarity and shine. The same rang true for any source I used this mic on. With a proper pairing of mic/preamp/source, I was able to get great, solid, useable tones on pretty much everything. (If you only have one type of mic preamp, and this was your only mic choice, you might find yourself reaching for an EQ to create the space you need for each element in your mix.)

The more I used the *U 47 fet*, the more I liked the cumulative effect it had when used on many elements of a mix. It reminded me of some of Ethan Johns' recent work, like his solo album *The Reckoning* and Laura Marling's *Once I Was an Eagle*. These recording have a nice easy-on-the-ears quality that is likely the result of great mics, preamps, tape, and superb performances — in contrast to so many contemporary productions that are so insanely bright that they are fatiguing even over short listening periods. Point being, that when left alone, I liked the character of the upper-midrange and high-frequency response of the *U 47 fet* and its non-hyped tone. Compared to "modern" multipurpose condensers, the *U 47 fet* maintains its "classic" tone by not having a pushed or over-accentuated "*fffttt*" in the upper ranges — without sounding dull.

I would still probably use this mic where it shines the greatest, on kick drum, bass, or other low-end rich sources that need capturing in the most complete and compelling way. Bass players and drummers will be thrilled to hear the power they are projecting into the room, captured with such authority. Guitar players will swoon at the balance and clarity in the recorded tone of their amps. The *U 47 fet* is a beauty of a mic, and anyone owning one will find great joy in putting it to work in a variety of situations. For a mic that has a street rep as a kick drum mic, it sure serves its master well for many a task. *($3,999 street; www.neumann.com)*

–Geoff Stanfield <www.geoffstanfieldrecording.com>

Native Instruments

Komplete Kontrol S49 keyboard
Komplete 10 Ultimate bundle

You guys, my home studio now sounds like the *Blade Runner* soundtrack, and looks like the set of *TRON*. And, what? There are *K*s where all the *C*s should be! Wait — lemme explain.

Native Instruments sent us their latest MIDI controller keyboard to test, the *Kontrol S*-series, along with the latest iteration of their flagship software instrument bundle, *Komplete 10 Ultimate*. As one would expect, the two are tightly integrated, with many features of the software not only directly controllable from the hardware controller, but performance-enhanced in ways that wouldn't be possible via a standard MIDI integration. We received the 49-key *Kontrol S49*, but it also is available in 25-key and 61-key versions. All three are similar, with semi-weighted key action, automatic parameter mapping to the eight touch-sensitive rotary encoders, built-in arpeggiator, and an LED-driven performance and control feedback system NI calls Light Guide (Why not "Light Cycle"?). *Kontrol* requires an external power supply in addition to the USB connection to the host computer, and also offers MIDI I/O, plus expression and sustain pedal inputs.

As software libraries go, *Komplete 10 Ultimate* is ginormous and does away with the multi-DVD installers in favor of a single 2.5" USB 2.0 hard drive to facilitate the installation. If you choose to install all of it, you'll end up with 320–440 GB of instruments, effects, and sound manglers, including stalwarts like Kontakt and Reaktor, plus new innovative instruments like Rounds. I went for it and installed everything, but specified that the library locations (the bulk of the packages) be installed on a fast external drive; this is highly recommended, not only for economy of record-drive real-estate, but for the best performance in general. As it is, the applications alone require 12 GB of free disk space. Although faster than the previous Komplete DVD installers, it still is a bit of a time investment to get everything up and running initially. Luckily, after everything *is* installed, you manage all of your NI software through one simple application, Service Center, which is NI's one-stop-shop for product activation, registration, and updates. Service Center is a comprehensive utility which many others have mimicked since its introduction. Keeping everything up-to-date is a snap, and NI already had quite a few hotfixes and product updates available throughout our test period. One note for *Kontrol S* users — I had to navigate to the NI website to find a firmware updater for the *Kontrol* keyboard. Service Center didn't seem to prompt me for a firmware update that was available. I didn't get that prompt until I launched Controller Editor, a separate application dedicated to the NI hardware controllers.

All of the *Komplete* software, as well as the *Kontrol* drivers and software browser plug-in (*Komplete Kontrol*), are available in 64-bit VST, AU, and AAX format — so DAW compatibility is pretty much universal. I had no issues testing in Pro Tools 11 [*Tape Op* #101 online] or Ableton Live 9 [#95], although many instruments can be fairly processor-intensive, particularly in sample-based applications like *Kontakt* with high instance count or multi-timbral parts loaded. I found that the instance counts and CPU loads aligned fairly well with other similar applications or plug-ins — no surprises there.

Komplete is a serious composition tool, one that could be almost overwhelming in its depth and complexity. This is where the integration with the *Kontrol* keyboards really shines; the keyboard controller has a built-in browser function that calls up a simplified navigation structure for all of the *Komplete* library, filterable by genre, tags, and instrument type. You load the *Komplete Kontrol* plug-in into an open MIDI track, press the Browse button on the keyboard, and go. The browse window itself pops up in your DAW and maximizes the screen real-estate with clear type and graphics. (It's obvious that the browser was meant to be legible from across the control room or stage.) Having the browser feature alone makes *Komplete* feel much more spontaneous and inspired. And if this "browse, discover, and load" hardware integration with *Komplete* feels a tad familiar to NI's now-discontinued Kore controller [#54], well, frankly, it is — and it isn't. *Komplete Kontrol* feels more like it has taken on the duty of presenting the entire *Komplete* library as one practical, browsable system of sounds and presets, and it succeeds at that task, whereas Kore was ambitious to a fault with its broader scope including third-party plug-in mapping. You can always dive deeper within the Controller Editor software mentioned above to create your own templates and mappings, but out of the box, this controller is meant to work seamlessly with NI's instrument library.

The *Kontrol* keyboard has a nice, clean Darth Vader–esque look, and feels like a serious instrument that belongs in a studio. The Fatar keybed isn't spongey, and the encoders, buttons, display, and ribbons are all high quality. Yes, ribbons — the mod and pitch wheels have been supplanted by these cool touch-sensitive strips with customizable physics. If you want to get an endless ping-pong mod parameter going, it's possible. The display strip below the encoders is clean and legible, and the automatic parameter mapping just works. Most instruments have at least two pages of parameters (some many more), so it's nice that the most commonly used parameters, like filter and envelope, are always present on the first page.

Also very compelling for me were the scale and chord

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features on this keyboard controller. I'm a terrible keyboardist, yet often find myself writing melodies and harmonies — on the damn keyboard. Punishing myself, like an ape-man learning to use a Commodore 64, I hammer away until something useful can be carved out of the hours of, um, "improvisation" I've tracked. Well, similar to the way the scale modes work on Ableton Push [*Tape Op* #97], *Kontrol* has an option to remove any non-standard notes from a particular scale, and another feature that allows chord sets to be played with a single key. I think you can see where I'm going with this; hardware tricks like this can be a force of good or a force of evil, my friends — it really depends on the application of said trick. In my case, I've been doing a great deal of short film scoring and also building quick and dirty content for the web, both of which require high production value in a short amount of time. Using the chord modes, grounded in *Komplete*'s cinematic strings, Action Strikes, and the on-board arpeggiator, I was able to build some really stellar soundtrack-worthy material for a short film project with a deadline that would make John Barry blush. And with a recent update to the *Kontrol* software, you can now "write" arpeggio or chord performance information back into your DAW for further editing, or to simply archive your performance. With a little creative MIDI routing, I had the *Kontrol* keyboard sending arpeggio and chord MIDI data, locked to a particular scale, to my Teenage Engineering OP-1 synthesizer — all via USB. I could even go back and manipulate that data after tracking it to my DAW. Pretty cool.

Another nice addition to the controller is its advanced host integration, which allows for transport controls and automatic track focus within the most recent versions of Ableton Live, Cubase, Nuendo, or Logic X. This means if you navigate to another instance of *Komplete Kontrol* in another track (within any of those four DAWs), the controller parameters automatically follow. The hardware automatically switches to MIDI mode if a third-party plug-in is present in the track. This advanced integration is not available in Pro Tools. Hopefully, NI will expand that feature to Pro Tools soon. Note that you can still access, browse, and control your *Komplete* library in Pro Tools, you're just missing out on the transport controls, track selection, and automatic track focus.

The Light Guide feature, which illuminates notes as they are played, or conveniently illustrates parameter/sample mappings across the keyboard, can be disabled, but why would you do that? It looks so cool! But maybe I've watched too many early '80s sci-fi films. Light Guide is also very handy if you wish to display notes within a particular scale as mentioned above, or if you're a live performer who needs to set up specific key mappings.

This is truly a great controller keyboard. Although the price may seem a bit steep, that price includes *Komplete Select* — ten NI instruments from *Komplete*, including one of their stellar sampled pianos (The Gentleman), vintage organs, and the always fun Retro Machines (a deep collection of analog synth instruments sampled from rare original instruments like the Crumar Orchestrator, Korg MiniKorg-700, and Moog Memorymoog). That's a bargain if you don't want to plunk down $500 for the full *Komplete* suite (or $1000 for *Komplete 10 Ultimate*). And if you are someone who has invested in NI's *Komplete* ecosystem, this is a no-brainer pairing. Although it functions just fine as a standard MIDI controller, I couldn't imagine using *Komplete* without this keyboard. And there is just so much within *Komplete* to explore; I could get pleasantly lost inside the new Rounds synth and Polyplex drum machine alone, not to mention the 40-bajillion other potential sounds available in the *Komplete 10 Ultimate* library. Fun stuff. *(Komplete Kontrol S49 $599, Komplete 10 Ultimate $999 ($399 update); www.native-instruments.com)*

–Dana Gumbiner <www.danagumbiner.com>

DPA Microphones

d:dicate ST2011A stereo cardioid mic kit
d:dicate MMC2006 omni mic capsule
d:dicate SBS0400 stereo boom

When I was in school at the University of Michigan, I did a lot of recital recording of ensembles and piano. Through this, I learned that all the orchestra recording was done with DPA mics (marketed under the Brüel & Kjær brand), which were apparently the best of the best. So I was excited to test and review several products from DPA: a *d:dicate ST2011A* stereo kit, which includes two *2011A* cardioid condenser mics, clips, windscreens, and a waterproof Pelican case; two *MMC2006* omni capsules; and the *SBS0400* modular stereo boom with shockmounts. I've been out of the recital business for a while and had not heard of these models, so I avoided looking up prices until I was done testing. I knew that the B&K line was quite expensive, and I didn't want that to cloud my judgement.

The first order of business was to record a freshly tuned 2009 Steinway Model B in a beautiful room with vaulted ceilings. I brought along what I would have tried if I hadn't had the DPAs — my trusty pair of Audio-Technica AT4051 cardioids with AT4049-EL omni capsules (used many times on the aforementioned recitals) along with the recently reviewed Monoprice 600700 [*Tape Op* #98] and 600850 [#105] mics. For one pass, I mounted the *d:dicate MMC2006* omni capsules onto the MMP-A preamp bodies of the *d:dicate 2011A* mics, and placed the mics over the strings, facing the player, but angled down towards the back of the piano, using the beautifully engineered *d:dicate SBS0400* stereo boom. The boom allows for precise spacing and angle setting, for various stereo techniques, of which I chose X-Y. Only the angle adjustment has tick marks, so bring a tape measure if you're doing ORTF. The Lyre shockmounts could not handle the weight of my bulky cables, and no matter how I dressed the cables, I couldn't get the Lyres to sit quite right, but the setup worked nevertheless. Bring thinner cables. I think thin cables would fit in with the overall theme of the accessories — stealth. For instance, all the mic holders' pivots are tightened by a small screw; there's no fitting a coin in there — you need a screwdriver. Clearly, the stuff is designed for doing live recording and sound reinforcement as invisibly as possible, by the kind of pros that bring their toolbox to gigs. The ATs with omni capsules were placed over and behind the player's head. I recorded four tracks at 24-bit, 96 kHz. Then I swapped in the cardioid capsules on both pairs of mics for a second pass. I followed that with two more passes after switching placement of the AT and DPA mics, and then swapping capsules again. Finally, I tried the Monoprice 600700 mics with omni capsules over the strings, while the 600850 mics were in a Blumlein pair in the aforementioned behind-the-head setup.

In all cases, I liked some mix of the string and rear mics, and overall, I can tell you the DPA mics crushed the others, but all the tones we got would be considered good in the right context. The *DPA d:dicate* mics would excel at audiophile piano recording of any kind; they sounded by far most like the piano in the room. I wished I'd had a second pair, because the omni capsules won the rear shootout and the cardioids won the strings shootout for fidelity. The Audio-Technica AT4051/4049 mics would be appropriate for a softer sound. They were very pillowy, and closer to "Hey Jude" than hi-fi, in a nice and dreamy way. The Monoprice mics were bluesy and imparted a honky-tonk tone, which may sound like an insult to a mic, but that's a pretty good trick if you can do it to a freshly maintained Steinway. The Blumlein pair by itself made the piano sound 100 ft wide when panned all the way — a good trick for ambience mixed low.

I compared the aforementioned piano mics, sans the 600850s, by measuring with my usual setup in an untreated room — not accurate for reference, but ok for comparisons. This was interesting, because the traces were not that different from each other, but the sound of the mics is obviously different. I think this is rooted in the time-domain; *i.e.* the transient response of the DPAs seems to outclass that of the other mics (and my speakers too).

I tried the DPAs in my home studio, on acoustic steel-string guitar, drum overheads, and male and female vocals. In all cases, they outperformed whatever I was comparing them to in terms of fidelity. As with the piano recording, any given mic sounded pretty good until you compared them to the DPAs, but other mics might still be chosen for "flavor" in the right context.

The *DPA d:dicate* mics and accessories that I tested are all built so well. The capsules screw on effortlessly, and the finish is low-reflective and scratch resistant. They are durable, proven by dropping an omni capsule from about belly height right after measuring. It hit my shoe and then rolled on the hardwood floor. I measured it again immediately, and the results were exactly the same, and there were no dings. Sorry about that DPA — but nice build quality to be sure. The *d:dicate* capsules use two mini diaphragms, a great idea that optimizes noise performance and transient response. I encourage you to look up more about that. These may be the ultimate "if you can only have one mic" mics. I was praying to the audio gods that these are from a new budget line, but their top-notch quality means they've got to be expensive. How much are they? Oh yeah, this is the good stuff. They are an amazing value for what they are though, in the same way a Porsche 911 is, and I'm not being ironic. If it's any consolation, the mics in the *d:dicate* 20-series are about half the price of those in the 40-series. Special thanks to the esteemed Gary Schultheis for his help.

(ST2011A kit $1,979, 2011A cardioid mic $949 each, MMC2006 omni capsule $499 each, SBS0400 stereo boom $479; www.dpamicrophones.com)

–Joseph Lemmer <jlemmer@siriusmedia.com>

Audient

ASP880 8-channel mic preamp & ADC

I first came across the *ASP880* when I was looking to expand my mic preamp inputs for an upcoming recording project. I was looking for a multichannel preamp unit to augment my collection without breaking the bank. I found many of the usual suspects (*e.g.*, API, Daking, TRUE Systems, etc.) too expensive for the project's budget, so I kept searching for a more cost-effective unit. I settled on a stellar solution in the *Audient ASP880*. The unit contains eight mic preamps in a single rackspace, and when you divide the street price of $1399 by eight, you get a very reasonable cost of about $175 per channel. Now that is certainly a bang-for-buck price tag, but in true infomercial style, "Wait, there's more!" I'm happy to say that you really do get more with this box — a lot more.

Aesthetically, the *ASP880* looks great with its attractive silver faceplate and colorful light-up buttons, but let's talk about all the physical I/O on this thing. To start, the front panel gives you control over the eight mic preamps, which are the same Class A design as found on Audient's flagship console. Two discrete JFET DI's are available on channels 1 and 2 in front. In back are the mic and line–level inputs. Each channel has switching for 48 V phantom power, –10 dB pad, polarity reverse, and input impedance (220, 1200, or 2800 Ω), as well as knobs for gain (0–60 dB) and high-pass filter (25–250 Hz). Small LEDs indicate signal and peak. In addition to all of this, each channel has a button to enable its pre-ADC analog insert point, which can also be used for direct access to the ADC channel, bypassing the preamp circuit. For this price, you wouldn't think that we'd even be talking about A/D conversion (more on this later), but the folks at Audient pulled out all the stops with this box. This is a big step up from their previous ASP008 offering.

The back of the unit has eight Neutrik Combo jacks (XLR and TRS) for the inputs. The aforementioned analog inserts are on two DB25 connectors. Eight channels of digital output are on DB9 (AES3 or S/PDIF) and ADAT optical (two of which are provided for S/MUX at higher sampling rates). A BNC word clock input includes switchable termination.

So, what about the mic preamps? How do they sound? Great. Each channel is made up of eight discrete transistors and an op amp, and the sound is clean and transparent with plenty of headroom on tap. Compared to some of my other preamps, the Audient seemingly has less character and color, but if you drive the Audient preamp a little harder, it can impart some analog goodness to your signal. I believe the Audient design is on par with preamps that cost significantly more and offer fewer onboard tools. Speaking of onboard tools, I found the input impedance to be very useful in getting tonally different sounds out of some of my dynamic and ribbon mics. There were a few tracks in a recent session that benefited from this feature, and I'm sure I'll employ it again. The high-pass filter is handy and useful for getting rid of some of the "flub" when you need to, polarity reverse and phantom power do what they should, and with regards to noise floor, the *ASP880* is pretty dang quiet.

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Now, onto the outputs and the ADC. The flexible routing offered by the *ASP880* is beyond useful. You want to utilize the preamps and then patch into external analog gear? Sure! You want to return your processed signals back into the *ASP880* and convert them from analog to digital? No problem! Maybe you want to use this box as a standalone eight-channel ADC. You're covered there, too. The converters are quality Burr-Brown PCM4220 multibit and offer 24-bit output up to 96 kHz. The converters sound pretty nice, and it's great to have all the digital and analog connectivity on hand.

There are a few nitpicky items, such as limited indicators for signal level, no included AES3 breakout cable, and no power switch; but when you look at what you get for your money, the *ASP880* is a righteous steal. Here's a real-world testimonial to the strengths of this box: I loaned my *ASP880* to a fellow producer for some additional tracking on a project, and shortly thereafter, he went out and purchased his own. If you're in need of a professional multichannel mic preamp, this one delivers. *($1399 street; www.audient.com)*

–Will Severin <www.willseverin.com>

Q2 Audio

ADR Compex F760X-RS dynamics processor

I've been thinking a lot lately about how we evolve as engineers, and in a way, it's my favorite part of the job. You wake up one day and realize you are doing something totally differently, like the day I realized I'd stopped obsessively fighting phase issues between bass DI and mic signal. In this case, a piece of gear helped — the Little Labs IBP [*Tape Op* #33]. As time goes on, I notice I'm not fighting for sounds as often. This is due to the fact that I've been making records for a while now, and doubtless, the gear here has gotten much better. Recently, I've been noticing the same phenomena in regards to drum sounds here at High Bias — more on this later. This is, however, due solely to the addition of the *Q2 Audio ADR Compex Limiter F760X-RS*.

The *Compex* began life as a limiter used for cutting records, and over time, it was modified incrementally into what it is today with compressor and expander/gate sections. It is a feedback type FET limiter/compressor, which means the sidechain input is taken from the output. The *Compex* is responsible for the sound you hear on countless records; most notably, it's the box behind the drum sound on "When the Levee Breaks." Due to this, the *Compex* became a legendary and coveted piece. I've always silently watched eBay auctions and muttered, "Some day...." Enter Tim Mead the genius behind the reissue/improvement of this venerable box. Tim is local to me in Detroit, and he's long been the go-to tech guy here, so when he told me he was working on an *ADR Compex* reissue, my ears perked right up. He dropped one off one day, and with a short tutorial, I was off to the races. Out of the box, this thing is handsome and sturdy as hell, and setup is as easy as can be. Tim hipped me to the gain-staging and how to dial it in — slightly confusing, but I got the hang of it quickly enough.

The first task for this box was, big surprise, drum room mics! The band for this date were Atlanta psychedelic dance wizards Hollow Stars. These guys play songs that are long and totally absorb the listener. Drummer Devin Brown is a machine in the best way. He just knows how to hold it all down and keep the groove happening for the duration of these 11 minute jams. Here at High Bias, we have four Shure SM82 mics hanging in the live room at all times. I permanently installed these when we opened 15 years ago, out of curiosity and a desire to hear clients in the room, both while playing and as a communication device, since we don't really use cans here. The SM82 is a broadcast mic that has a crude facsimile of the Level-Loc inside, set at a super-low threshold, and it puts out line-level. The SM82 mics have contributed to the sound of many recordings here, and I've always loved the smashed and distorted sound they make. Plus, I get to pretend I'm Tchad Blake! So I took the room mic bus on this session, which consisted of a pair of the SM82s hanging along the walls equidistant from the kit, and strapped the *F760X-RS* across it. I felt slightly stupid compressing something so smashed already. If anything, the unprocessed mics were already providing a nice din of drum racket, the kind that makes even the most anemic drummer sound like a pounder. But somehow, the *F760X-RS* reclaimed the attack of the drums, reminding me of how an SPL Transient Designer [*Tape Op* #21] can change the envelope of percussive sounds. After tweaking the compressor and expander controls to taste, the result was easily the best drum sound I had gotten in recent memory. From then on, the *F760X-RS* stayed in this position for a few months with rare exception, and I quickly learned that the *F760X-RS* really performs — from subtle, barely audible pumping or ducking, to extreme reshaping effects. For sure, sounds take a minute to get to, due to the vast control over the circuit, but settings are easily recallable via the detented knobs that the unit sports.

I finally moved the *F760X-RS* when Chris Bathgate came in to record with his collaborative group SKULLLS. Chris is a searcher. His music ranges from rootsy, psychedelic groove folk to astute, new school compositional piano rock. SKULLLS falls into the latter category. Ben Gajino plays drums for this outfit. Ben is one of those drummers that's a blast to record because he can do nearly anything. Intrigued by its fast and aggressive behavior, I moved the *F760X-RS* from the room mics to the kick and snare. I almost always use a dbx 165 or UREI 1178 in this role. Ben's snare dynamics are intensely expressive, and the *F760X-RS* did amazing things to the transient response. I started out just adjusting the compression controls and then got into some serious sculpting. This thing can really change the whole sound and behavior of tracks. After drum tracking, we moved on to piano using a pair of Josephson C42 condenser mics [*Tape Op* #34], Rascal Audio Two-V preamps [#102], and the *F760X-RS*. The same was true here, and a whole range of sounds was achieved — from barely audible, yet super functional compression, to all-out crusher sounds. This day, I started experimenting with the limiter. The *F760X-RS* has a 250 µs, 100:1 limiter, that when employed, takes care of any peaks that may have snuck through. We had the compressor and expander set just right for effect, but the initial piano hits were a little intense. I kicked in the limiter, and problem solved.

The next use of the *F760X-RS* was for a mix with Beekeepers, a band of free rock upstarts from Hamtramck, MI. They make beautiful and strange art rock. Like Talk Talk meets Tall Dwarfs with David Stoughton. Jeff Else is their drummer, and he records their stuff at home using minimal mics and a simple Logic 9 rig. The sound is very Conny Plank. Super fun to mix. We put the *F760X-RS* on the drum bus at mix, and it was simply magic. Before I even starting turning knobs, it brought the drums to life in a beautiful way. That's one of the things I noticed time and again — the *F760X-RS* brings the material to front in a really natural way, and from there, you can continue to trick out the dynamics to really dial it in. On this day, we used the limiter in pre-emphasis mode. This allows the limiter to de-ess slightly, which was handy since Jeff uses maybe two mics on drums, and cymbals can be an issue. The results were perfect.

Gosh Pith are a New Detroit R&B duo. Their music is smoky and hazy with the dankest trap beats this side of the Mississippi. Josh Smith (get it?) sings, and they, as above, record at home, but in Ableton Live. They do a great job of tracking their songs, so it's usually just them paying me to do a hockey-stick mix, busing it through my trusty Purple Audio Sweet Ten [*Tape Op* #100]. This day, I used the *F760X-RS* on the vocal bus. Again, the forwardness was instantly apparent, and I used the pre-emphasis to de-ess the vocals. The vocals sat perfectly — louder than everything else, but not overpowering at all. The *F760X-RS*'s limiter just provides that kind of high-class limiting that you can't hear.

Scotty Masson is a stone genius weirdo pop songsmith from Ferndale, MI. He cut his teeth in tons of Michigan rock bands in the '90s. Then he went to Chicago with his band The Office. Out of the blue one day, he called and came by to mix his album of expertly recorded jams. Scotty is a gear nerd who's up for anything, so we put the *F760X-RS* on the mix bus! Since our mix had a lot of low-end information, the *F760X-RS* seemed a bit grabby on the bassy material, but this was nothing a little side-chaining couldn't solve. The resulting mix was super-polished and slightly more aggressive than we were going for, so we dialed it back, and it was great — upfront and with character, super pro but not too slick.

Like most engineers, I remember the first time I heard Led Zeppelin, and for me, this totally coincides with the first time I was aware that records are *made*. Until then, I thought that 25 ft tall rock gods created what was coming out of my dad's stereo. (Gimme a break; I was 9 at the time.) The fact is though, that even to my young ears, I sensed something different was happening on those recordings. A decade and a half later, in the mid '90s, I got a job at the incredible Ultrasuede Studio. We had a "live" room that was mostly carpet, so room mics sounded comb-filtered and weird. I'd put pegboard all over the floors and walls, but to minimal avail. Anytime I worked in other rooms or houses that were more reflective, I'd put the drums somewhere live and position mics where I thought they should go — and then sat there wondering where the "Bonham" was. (It turns out the drummer matters!) A decade or so passed, and I got a room I liked here in Detroit. Still experimenting, we put mics everywhere here. Over the course of 200 or so records, I didn't really realize I was chasing this sound I heard as a youth. Some of this was due to the fact that not every band needs or wants drum sounds like this, and also to the ubiquitous nature of this sound and its imitations. I took it for granted and simultaneously assumed it was unattainable. The *F760X-RS* has made me realize just how much the fabric of my engineering modus operandi was sewn together with the threads of this box. It's exactly what I wanted for the above applications, and I never knew it until it showed up. Furthermore, I never realized how magic that sound was and how integral it would become to me, even at this stage of the game. *($2,750 street; www.q2audio.com)*

–Chris Koltay <www.highbiasrecordings.com>