Rihanna's Engineer Greg Ogan on World-Class Vocals

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—What You Need To Know

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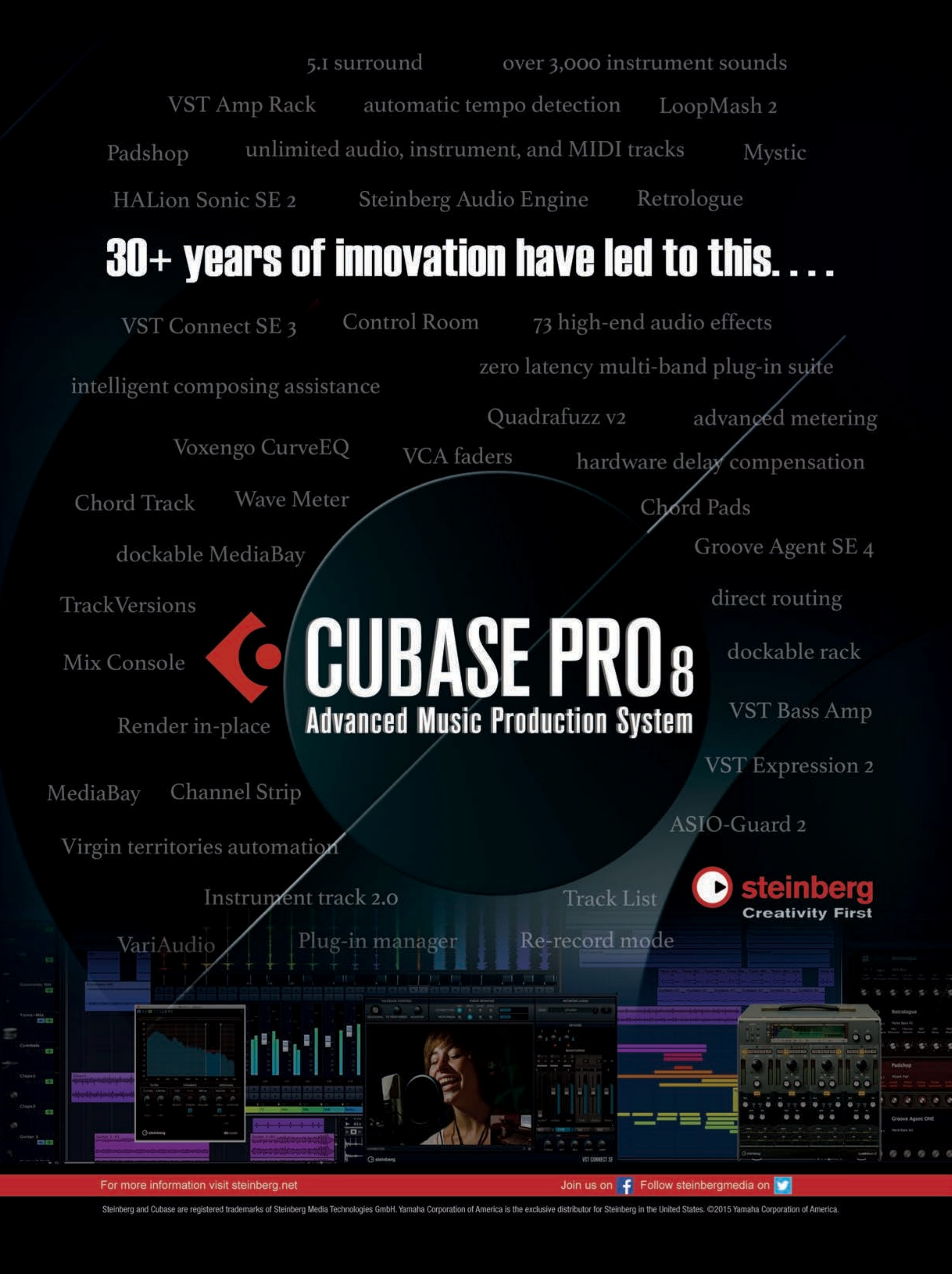
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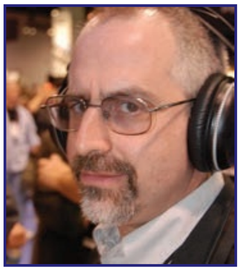


Make new mics but keep the old

Paul Vnuk's been at it again... check out these two photos, each of which tells a fascinating story.

First we have two identical microphones set up side by side at the beautiful Chicago Recording Company (chicagorecording.com). Wait a minute—*identical*?

Yes, believe it or not. On the left is a battered old veteran and on the right is a beautiful brand-new mic... both fine examples of Neumann's famous U 47 fet microphone, a studio standard that's beloved around the world. The U 47 fet hasn't been produced since 1986, and Neumann has just reissued it; check out page 20 to read about whether they got it right.

The other photo is one of Paul's classic "star arrays" for mic comparison. The new U 47 fet is on the left, and the Pearlman Microphones TM-47, which got a stellar review in our September 2014 issue, is at right. On the bottom is the Australian-made Beesneez T-1 Tribute... but what is that insanely beautiful mic up top, the one that appears to be... *glowing*? Meet the Cathedral Pipes Notre Dame, which graced our front cover photo (shot in Paul's own Moss Garden Studio in Milwaukee) and is reviewed on page 60.

These mics—a reverent nod to the past and an amazing design for the future—encapsulate a lot of what's cool about this industry. An intelligent (not slavish) respect for the past, balanced with an eager (not gullible) interest in new ideas, makes for a healthy sort of progress in developing worthwhile music gear.

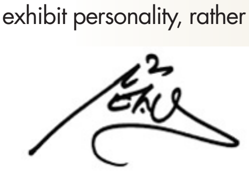
In the "new ideas" department, Audio-Technica's AT5045 mic uses an unusual capsule design that'll be new to most folks but will likely become very popular in the coming years, and Resident Audio's T4 interface takes advantage of the blazing speed of Thunderbolt. In the "reverence" department, API offers 500 Series versions of classic console processors. And in a gorgeous blending of old and new, we introduce you to Universal Audio's plug-in emulation of the Manley Labs Massive Passive EQ... but we also review the real beast and explain why it's a modern classic.

Elsewhere in this issue, we devote our attention to the art of getting great vocals, with fantastic pieces by educator Michael Schulze and top engineer Greg Ogan on everything from choosing a mic and processing a vocal to preparing your DAW for a multitrack vocal overdub session. We also premiere a new column in this issue: "Specs: Removing The Mystery", by stalwart audio engineer, gear designer, blogger, and all-around guru Mike Rivers. In these articles, Mike will explain what all those numbers mean, why you should care... and whether they actually matter at all.

I'll close with a shout out to those who are reading this magazine at the 2015 Winter NAMM Show—welcome to our pages and stop by our booth to say hello!—and a quote from mic expert Klaus Heyne, extracted from an interview that appears as part of our Neumann review (a more complete version will be published online):

"Whoever at Neumann was bold enough to suggest this venture: Thank you for finally recognizing that microphones are the one remaining human link left between musician and digital recording device, and as such, we expect a microphone to exhibit personality, rather than just faithfully reproducing information."

Words to live by. Enjoy the issue!



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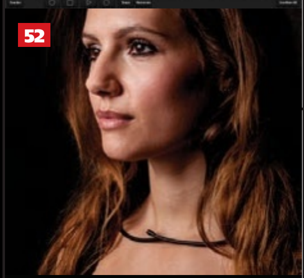

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talkback

Note to faithful writers: When sending your letters by email, please remember to include your name and where you hail from.

Whys and wherefores of +4 and –10

In this issue we include the first installment of "Specs: Removing The Mystery," Mike Rivers' new series on demystifying spec sheets and understanding what all those numbers really mean. This series will run on an irregular basis in future issues, and will hopefully span a wide variety of interesting and often misleading specs. I am looking forward to Mike bringing you precision and clarity on topics that we must sometimes simplify in our attempt to make audio production as accessible as possible to as many readers as possible.

The only trick to working with Mike is knowing that he has so much good information to impart that it's my sad duty to stop him before he fills an entire issue! For example, soon after he sent me Part 1 of his new series, he followed up with this gem of history about where the seemingly random "+4" and "–10" level standards come from. There was no room to fit it into its article, but it was too interesting to leave out of the issue, so here it is. Enjoy!—*MM*

Ohm's Law tells us that 0.775 volts into 600 ohms gives us a power of 1 mW, or 0 dBm (since the dBm measurement is referenced to 1 mW of *power*, rather than a voltage). When we wanted a simple number to represent this voltage, we logically chose to call it 0 dBu. +4 dBu became a standard reference level since this was the voltage necessary to make a standard VU meter read 0. Why didn't we make the VU meter's 0 reading our 0 dBu reference? Because it was impractical to build a meter that was sensitive enough to read 0 with 0.775 volts applied to it, yet also had a high enough input impedance so as not to load down the source driving it.

When "prosumer" audio gear began appearing, designers using the integrated circuits available at the time were limited to a maximum output level of around 3.5 v RMS, about +11 dBV, to keep costs down. Because they wanted to be able to say that this gear had roughly the same amount of headroom as "pro" gear, they simply moved the goalposts. They established a nominal operating level of –10 dBV and, abracadabra, instantly they had 20 dB or more of headroom! This actually worked out pretty well as long as users were only interconnecting pieces of –10 dBV gear to one another. It only became a problem (and developed its "not really professional" bad rap) when it got connected with +4 dBu gear and the level mismatches caused difficulties with gain staging.—*MR*

Tracking the Phase Monster

Dear Paul Stamler: Thanks for your article on the "Phase Monster" (November and December 2014). I have an issue with applying the traditional three-to-one rule as it relates to recording large classical choruses of 30–50 members.

The directors of these groups do not want a close miked recording but prefer a blend of voices and sections. If I apply the rule and place the mics for a stereo recording back from the group, perhaps as much as 15–18 feet from the first row, then I would need to separate the mics as far as the edges of the group. This will result in a hole in the middle.

How do you recommend miking with a stereo pair? My mics are AKG414s, Sony C37As or Neumann TLM103s, into a Millennia Media HV-3.

Thanks for your advice.

Wally Knapp
Custom Recordings, Ellicott City, MD

Paul Stamler replies:

Hi, Wally. You actually raise two (related) questions: the hole in the middle, and managing phase issues in a true-stereo recording. Let's discuss the hole first.

This is a familiar problem with spaced mics; the classic remedy is to use a third mic in the center, mixed to be about 6 dB below the left and right mics. It needs to be the same mic as the other two, and takes another channel of input. This provided excellent results on the Mercury Living Presence and RCA Living Stereo records of the 1950s.

But there's a down side, which you've already guessed: the Phase Monster. It doesn't rear its ugly head in a true stereo recording; one mic is panned hard left, while the other's panned hard right. They don't combine, so there shouldn't be any phase issues. However, if the recording is played back in mono (as it may well be on, say, a clock-radio), then the two spaced-mic signals (or three) will combine, and the Phase Monster will jump up and bite you.

The answer is to do coincident or near-coincident miking. Do you have a pair of small-diaphragm cardioid mics? If so, you can put them nose-over-nose, angled at 110 degrees, forming an XY setup, or splay them outwards, again at 110 degrees, with their capsules separated by 7"—the classic ORTF configuration. Put either assembly a couple of feet behind the conductor's head, 6–8' off the ground. Route the left-pointing mic to the left channel, and the right-pointing mic to the right channel. Tweak the position as needed.

If you only have the mics you listed, though, you should probably use a Blumlein setup. Set the two C414s to figure-8 patterns; mount them so the capsules are stacked on top of one another, and angle them at 90 degrees from one another. Place behind the conductor as above, route left/right, and again tweak as needed. Be aware that this setup will pick up a lot of the hall sound, but with the capsules so close, there won't be phase problems, even if you listen in mono. Peace—*PJS*

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FASTFORWARD

Steinberg Announces Cubase Pro 8 and Cubase Artist 8

Cubase Pro 8 and **Cubase Artist 8** are the newest versions of **Steinberg's** flagship DAW. The new version boasts a wide selection of new and improved features, based upon a complete audio engine rebuild from the ground up. Steinberg's ASIO-Guard buffering technology now applies to multitimbral and disk-streaming instrument tracks for improved performance.

New features include VCA faders that allow grouped fader movements combined with existing automation curves, virgin territories (stripping of redundant automation data), render-in-place to free up CPU resources, Chord Pads and Chord Assistant modes for one-touch chord performance and chord suggestions based on harmonic rules, MIDI tempo detection and alignment to the grid and tempo track, UI enhancements, enhanced channel strip EQ, preamp filter slope selection, new effects plug-ins, the new **Allen Morgan Pop-Rock Toolbox** for **Groove Agent SE 4**, and more.

Cubase Pro 8 adds to the Cubase Artist 8 feature set with MixConsole's new direct routing section for group routing of audio of multiple channels and buses, a Wave Meters feature showing audio events in MixConsole without consulting the Project window, VST Connect 3 with fully integrated audio/MIDI data recording over the Internet, and much more. Look for a review soon.

Prices: Cubase Artist 8, $329.99; Cubase Pro 8, $599.99
More from: Steinberg, www.steinberg.net/cubase

Sennheiser Now Shipping Multipattern Vocal Mic

The **Sennheiser MK 8** was first announced earlier this year and is now shipping. It's a multipattern dual large-diaphragm condenser mic with a selection of five polar patterns: omni, wide cardioid, cardioid, supercardioid, and figure-8. It offers a pair of gold-sputtered diaphragms and switches for 0 / –10 / –20 dB pre-emphasis (pad) and two lowcut filter settings: 60 Hz corner frequency with a steep 18 dB/octave slope, and 100 Hz corner frequency with a shallow 6 dB/octave slope.

Specs include 10 dBA equivalent noise level, 142 dB maximum SPL, 132 dB dynamic range, and a 20 Hz–20 kHz frequency response. We've just obtained a pair for review, so watch for the MK 8 in an upcoming issue.

Price: $749
More from: Sennheiser USA, en-us.sennheiser.com/mk-8

IsoAcoustics Premieres Aluminum Speaker Isolators

We saw prototypes at the AES Convention in Los Angeles, and now they're here with a new name: **Aperta**, the new speaker isolation stands from **IsoAcoustics.** Aperta, meaning "open" in Italian, is the first non-customized IsoAcoustics product to be machined from aluminum rather than combining metal and plastic construction.

The aluminum stand frame, pioneered for IsoAcoustics' build-to-order service, has been reimagined in a lightweight and beautiful yet sturdy form for small studio monitors.

The new speaker isolators measure 6.1" x 7.5" and can support speakers weighing up to 25 lbs. They feature a tilt adjustment mechanism for precisely angling and aiming the speakers, complete with fine gradation markings to assure matching angles between speakers.

Price: $240/pair • More from: IsoAcoustics, www.isoacoustics.com

Which **Scarlett** is for you?

Scarlett 6i6

Duo / Trio

You want to record a small group of musicians, for example a keyboard player and two vocalists. So, you need a durable, portable device that is capable of recording up to 4 simultaneous analog inputs and also has 4 audio outputs for monitors or additional cue mixes. **The Scarlett 6i6 is for you.**

Use the 2 mic pre amps on the front to record the vocalists and use its line inputs on the back to record the output of the keyboard.

- 2 Focusrite Mic Pres
- 2 mic/line/instrument combination inputs
- 2 headphone amps, 4 line outputs
- Stereo line inputs
- MIDI and S/PDIF I/O

Scarlett 18i8

Band

Your band needs an interface that is both portable and expandable. You want to record up to 4 mics for, say, an acoustic set (e.g. guitar, keys, vocalist, and percussion) and independent headphone outputs for different monitor mixes.
The Scarlett 18i8 is for you.

For larger ensembles, take advantage of Scarlett 18i8's ADAT input and track up to 16 analog inputs. (e.g. Focusrite OctoPre MkII)

- 4 Focusrite Mic Pres
- 2 mic/line/instrument combination inputs
- 2 mic/line combination inputs
- 2 headphone amps, 2 line outputs
- 4 line inputs, MIDI and S/PDIF I/O
- ADAT Input

Scarlett 18i20

Band+

You are very proud of your home studio and you love recording bands. You want a high quality, reliable, and rack-mountable audio interface that can record up to 8 mics at the same time. You need multiple outputs for headphone cue mixes and main monitors. **The Scarlett 18i20 is for you.**

For even larger ensembles take advantage of Scarlett 18i20's ADAT input and track up to 16 analog inputs. (e.g. Focusrite OctoPre MkII)

- 8 Focusrite Mic Pres
- 2 mic/line/instrument combination inputs
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FASTFORWARD

Acoustica Rolls Out Version 7 of Mixcraft

Acoustica has released **Mixcraft 7** and **Mixcraft Pro Studio 7**. The latest versions of the popular and affordable Windows DAW include 32/64-bit versions, a new interface, and many upgraded and added features and content.

A new Performance Panel is designed to allow composition and performance live and in the studio, with realtime control over audio loops, MIDI clips, and samples, with extensive support for external MIDI controllers or the **Novation Launchpad.** Mixcraft also supports **Mackie HUI** compatibility and a new iOS/Android **Mixcraft Remote** app for wireless control from a tablet.

Mixcraft 7 also adds the **Copula** time and pitch manipulation system, a new drum sample library, and four new virtual instruments, two samplers and two virtual synths from **G-Sonique** and **AAS.**

Mixcraft Pro Studio 7 adds a large slate of new plug-ins, including effects from **iZotope, Studio Devil, BeatRig, QuikQuak**, and many more.

Prices: Mixcraft 7, $89.95; Mixcraft Pro Studio 7, $164.95
More from: Acoustica, www.acoustica.com

MOTU Adds Multichannel Input And Output Interfaces To Its Line

The **24Ai** and **24Ao** are the two newest rackmount audio interfaces from **MOTU**, featuring 24 ins and outs of up to 24-bit/192 kHz audio, respectively. Aside from the 24 analog channels (available on DB-25 D-Sub connectors), each interface adds three Toslink ports for ADAT optical data, bringing the full complement of ins or outs to 72.

The two interfaces can be networked to one another and to a computer via AVB (Audio-Video Bridging) Ethernet. They can connect to previously announced AVB-capable MOTU devices such as the **16A**, whose Thunderbolt connectivity allows for up to 256 channels of audio I/O. The 24Ai and 24Ao come with USB 2.0 and word clock I/O.

Other features: large backlit meters, digitally controlled analog input trim, 32-bit D/A output trim, 48-channel mixing and DSP effects on board with 32-bit floating precision, flexible routing, the ability to operate in standalone mode, Wi-Fi control without the need for a computer, and much more.

Prices: $995 each • More from: MOTU, www.motu.com

GForce Software Ships Oddity2

GForce Software has released **Oddity2**—a massive upgrade to its well-known **Oddity** virtual synth, an emulation of the **ARP Odyssey** analog monophonic synthesizer from the 1970s. The new version adds extra features not found on the original hardware, greatly widening its sonic possibilities.

New features include three filter modes, an additional oscillator for a total of two syncable oscillators plus a sub oscillator, duophonic legato and fully polyphonic modes, onboard delay, new pan modes, patch morphing, and a new user interface based on the final version of the original Odyssey, discontinued in 1981.

Price: £139.99 (approx. $187)
More from: GForce Software, www.gforcesoftware.com

The Right Mic For The Right Vocal

Know your mics and you're on your way to a great vocal recording

By Michael Schulze

Microphone technology has evolved a great deal over the last century, and throughout that history there have emerged a handful of "go to" mics that have dominated particular periods and genres. Many of your favorite modern bands are creating music that harks back to specific historic periods, using some of the same production techniques that were used by the greatest producers and engineers of all time.

Choosing the right vocal mic for your singer and your project can create a sonic vibe that you can build your entire mix around. In this article I will write about a few of the holy grail vocal mics of popular music history, and how you can emulate the sound of great old and new recordings using currently available and affordable microphones. I'll reference some specific recordings, but you can search for any of these mics on YouTube and find many audio examples and even shootouts with much less expensive modern designs.

Most vocal tracks today are being cut with large-diameter, dual-diaphragm condenser microphones. However, before I explain that mouthful, let's take a look at an often ignored alternative: the ribbon dynamic mic. (We'll also take a look at moving coil dynamic mics in a bit.)

1

Ribbon mics: a rich history

The first commercially available ribbon microphone was the RCA 44-A (shown in Figure 1), introduced in 1931. A few years later the updated 44-BX quickly became *the* mic for vocals, radio announcers, horns, and just about everything else due to its warm, smooth sound. To hear this mic check out Frank Sinatra's *The Voice of Frank Sinatra*. Many of the earliest Elvis Presley recordings used this mic.

Ribbon mics contain a very thin strip of aluminum foil, only 2–3 microns thick (2–3 *millionths* of a meter). The ribbon is corrugated to make it "springy" so it can be stretched to a specific tension, which extends the low end response by establishing a mechanical resonance at a specific frequency. The ribbon is suspended within the field of a large magnet. (I can pick up small screws with my Royer R-121!)

Sound pressure waves arriving at the front or back of the ribbon cause it to vibrate, and vibrating a piece of metal within a magnetic field makes electrical current flow. Wires connect the two ends of the ribbon to a transformer to balance the signal and increase the output impedance to about 300 Ohms, and that's it. See Figure 2 for the inside (and outside) of the Royer R-121. Remember, ribbon mics have a lower output than any other type of mic, so expect to turn up the gain more than usual.

Pressure waves arriving from the sides push or pull the front and back of the ribbon equally, so the ribbon doesn't move and no sound comes out. This is why a traditionally designed ribbon mic has a bidirectional ("figure-8") polar response, picking up sound from the front and back, but not the sides. Later in the 1930s, RCA introduced the pill shaped RCA 77-DX ribbon mic, which combined omnidirectional and figure-8 ribbon elements to produce a cardioid, or unidirectional, polar response. The modern Beyer M160 uses the same principle. A single element ribbon mic is a *pure pressure gradient* mic because it produces sound based on the *difference* in pressure between the front and the back of the ribbon. Pure pressure gradient elements exhibit the most extreme proximity effect of any type of mic: you get more low end response the closer you are to the mic.

While some modern designs like the Shure KSM313 or Sandhill 6011A are fairly rugged, most ribbons, especially vintage designs, are incredibly fragile when hit with gusts of moving air. That 3 micron ribbon can stretch out of shape with the slightest puff of air, even if all you're doing is walking while holding it! So *never* blow into a ribbon mic or put it in front of the hole on a kick drum (guitar amp cabinets should be okay), cover it with the provided "sock" while moving it, and always use one or more pop filters between the singer and the mic... or you'll be paying to have the mic's ribbon replaced. Oh, and store the mic upright; older ribbons can sag under their own weight even if you just set them down on their side for a while.

To get the most out of a ribbon mic you must understand three things. First, if you are using a figure-8 ribbon mic, make sure there is no reflective surface behind the mic. Back up away from studio windows and walls! However, the rejection of sound from the sides provides much more isolation than cardioids. I frequently record 3 or 4 horn players standing shoulder to shoulder in a very small room. I love to use figure-8 ribbons because there is little leakage between the mics.

The Right Mic For The Right Vocal

Some ribbons, like the Royer R-121 and the Shure KSM313, are brighter from the rear, so try them both ways! Be aware that sounds arriving from the back of the mic will output in reverse polarity, or 180 degrees out of phase. This is an issue when you combine the signal with another mic, like a backwards ribbon and a Shure SM57 or Audix i5 on a guitar amp. You will want to reverse the polarity of one of the mics so they don't sound thin when mixed together.

Second, most ribbons have a flat frequency response from 20 Hz up to about 10 kHz. Some have a high end response that extends almost to 20 kHz, but even these will sound darker than moving coil dynamics and condensers, most of which boost high frequencies. If you are tracking vocals for a bright pop mix you may find ribbons too dark, but in other situations you may prefer this. As the Royer Labs website says, ribbons "hear like your ears" and do not color the sound in the high end. Ribbons are wonderful for solo vocals in sparser arrangements, and can tame bright, edgy voices. They are also popular for miking guitar amps for a thick sound with a beefy low end.

Third is that massive proximity effect—use it! Try different distances for the exact low end boost you like. You may find yourself placing it farther away than you expect; skillful placement is your magic bullet! Don't be afraid to use equalization to boost the high end, even aggressively. You will still get a certain ribbon vibe: bright but smooth. Many engineers describe the ribbon sound as "compact" with a warm and detailed midrange, great for tucking your lead vocal neatly between the darker and brighter elements of your mix.

Check out The White Stripes' *Icky Thump* album. According to producer/engineer Joe Chiccarelli, the title track was done with a Neumann U47 condenser but the rest used the RCA 77-DX ribbon or the Shure SM7 moving-coil dynamic. Tracks 2 and 3 in particular feature a nice midrangey lead vocal that sits underneath the brighter instruments, blending in more than riding on top of the mix. I'll bet you $5 that was the ribbon!

2

There are many fantastic ribbon mics being made today. Wes Dooley of Audio Engineering Associates (AEA) has reproductions of the RCA 44 plus his own original designs. You can still buy the venerable Coles 4038, which was Ringo's overhead mic in the early 1960s and also used at Abbey Road for horns and guitars. The Royer R-121 is a bit more clear sounding than some ribbons and is popular for vocals, guitar amps, and piano. The Shure KSM313, with its unique "Roswellite" ribbon, is unusually resistant to wind damage. sE Electronics has a line of affordable ribbons with extended high frequency response, some with a condenser style 10 kHz boost. My current favorite from a price/performance standpoint is the Cascade Fat Head, a great-sounding ribbon for less than $200! It's on the dark end of the spectrum, but that's what EQ is for.

Dynamics: the sound of rock and roll

In moving-coil dynamic mics, a coil is attached directly to a plastic diaphragm. Behind that assembly is a magnet. Sound waves vibrate the coil in the magnetic field, and a signal comes out. This signal is connected to an internal transformer and that's all there is to it.

This type of mic does not respond as quickly to transient signals as a ribbon because of the higher mass of the diaphragm/coil assembly, but it is much more rugged and cannot be damaged by a puff of air. It is also much cheaper than a ribbon.

You can hear Stevie Wonder sing into the Electro-Voice RE20 (Figure 3) on "Superstition", and a Shure SM7 (Figure 4) was used for most of Michael Jackson's vocals on the *Thriller* album. Even the pedestrian Shure SM57 and SM58 have been used to cut hit records! These mics will put you somewhere between condensers and ribbons sonically, and are great when you want something brighter than a ribbon without the sheen of a condenser.

Condensers: the new standard

In Berlin in 1947, Georg Neumann introduced the U47 large diameter, dual-diaphragm condenser microphone (Figure 5). The U47 employed the VF14M vacuum tube, manufactured by Telefunken, which put its own logo on the mic and distributed it. The "U" stands for *umschaltbar*, German for "switchable", because the U47 was the first condenser ever to switch polar patterns, in this case omnidirectional and a roughly supercardioid pattern. Neumann designated the two patterns as "Kugel" (ball) and "Niere" (kidney).

The first U47 mics featured Neumann's M7 condenser capsule. The front of the capsule is a very thin (8–12 micron) circular piece of gold coated PVC (polyvinyl chloride) plastic film. The diameter is 1 inch, which we refer to as "large diameter". This diaphragm is tensioned about 40 microns in front of a brass backplate. DC voltage applied across the diaphragm and backplate charges the capsule. Sound waves arriving at the front of the capsule vibrate the diaphragm, causing a tiny variation of the charging voltage. The internal tube circuitry amplifies this changing voltage and sends it to the output transformer, which isolates and balances the signal.

A precise pattern of small holes drilled through the backplate allows some sound pressure to reach the back of the diaphragm. Amplitude and phase differences between the front and back of the diaphragm result in a polar pattern that is basically supercardioid around 1 kHz, more directional at higher frequencies, and more omnidirectional at lower frequencies. We refer to this kind of capsule as pressure gradient, but not a pure pressure gradient because not all the sound can get at the back of the diaphragm. (If the backplate had no holes in it at all, you would have a *pure pressure omnidirectional* capsule with no proximity effect, but that is a topic for another article!) This kind of capsule exhibits proximity effect, but not as much as a pure pressure gradient like a ribbon.

On the back of the capsule is a second diaphragm sharing the same backplate. In the cardioid setting only the front diaphragm is activated. In the omnidirectional setting both diaphragms are activated and their signals added. The U48, released in 1957, was switchable between cardioid and figure-8. The figure-8 pattern was achieved by *subtracting* the back signal from the front, which also results in signals arriving from the back of the mic being output in reverse polarity.

Around 1960 Neumann started using the new K47 and K49 capsules, which used 6 micron polyester diaphragms instead of PVC. The K49 capsules had more precise matching of the tension between the front and back diaphragms for more uniform omni or figure-8 patterns. Nowadays the original M7 capsules don't hold up very well due to drying and deterioration of the PVC.

By the 1950s Neumann had taken over distribution of the U47 and put its own logo on it, but in the 1970s Telefunken stopped making the mic's VF14M tube. Neumann experimented with different tubes but could not find one he liked, so he discontinued the U47, saying, "Don't try to bend the laws of physics in pursuit of a particular sound. If people want the sound of the U47, it's up to them to try to get hold of one."

However he did introduce the solid-state U47fet. This phantom-powered mic used a Field Effect Transistor instead of a tube. The U47fet was less expensive, and had only a cardioid pattern. Those K47 capsules that did not have the tighter matching between the front and back were used. Even though only the front diaphragm was activated, it is a good thing that the back diaphragm was there, because dual diaphragm mics have better behaved proximity effect up close. This is explained in one of the most important papers you will ever find on microphone technology:

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Unique Directional Properties of Dual-Diaphragm Microphones by Guy Torio and Jeff Segota, available on the Shure website. Read it—it will change how you do everything! The U47fet sounded a bit different than the U47, and didn't catch on for vocals in the same way, but became very popular for kick drum and upright bass. You'll find a review of Neumann's brand-new U47fet reissue on Page 20.

The U47 has an instantly recognizable sound due to the combination of the capsule, tube electronics, and output transformer. The cardioid frequency response exhibits a gradual bass rolloff below 300 Hz and a 4 dB double camel hump rise centered around 4 kHz and 10 kHz (easily seen in Figure 6). This makes the U47 noticeably brighter than the ribbon mics that preceded it, and engineers and listeners have remarked on the enhanced "clarity" of the sound. A U47 has a more pronounced proximity effect than later designs, which balances out the high midrange clarity when you get close. This combination of low end warmth and high-high/mid clarity is the U47 sound!

The mic was expensive, about $3500 in today's dollars, so only well heeled studios could afford it. Capitol Records bought a bunch, and it became Frank Sinatra's mic of choice—he called it his "Telly". Pictures show him a very consistent 1 foot or so away, with no pop filter. At that distance his voice sounds very well balanced, warm but without a noticeable bass boost, but on various songs you can clearly hear him moving in and out. The album *Songs for Swingin' Lovers* is a great example of Sinatra at his best. On *We'll Be Together Again* you hear him sing the first two low quiet lines rather close, and then as he goes into his higher register it is quite obvious that he backs up a bit. The sonic effect is quite remarkable! Low, quiet passages sung very close yield a blooming, intimate warmth, and higher, louder passages at a distance really bark at you in a way that cuts without getting edgy.

Producer George Martin used the U47 and U48 on the Beatles from the very beginning. During the recording of the first few Beatles albums there were strict rules at Abbey Road specifying just how close to a mic you were allowed to get. Some session photos show the Beatles singing about 8 to 10 inches away for solo vocals and up to a foot or two for trio vocals.

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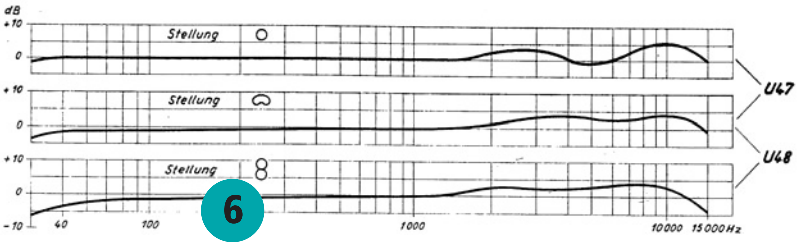

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The Right Mic For The Right Vocal

In those sessions the Beatles set up almost as if they were on stage, with no headphones. Up front a U48 was placed, set to figure-8, and turned sideways to reject the drums and amps. It was common for two Beatles to sing into a single U48, one on either side of the figure-8 pattern. The moderate distance resulted in the clear, aggressive vocal sound of those early records. Beatles producer George Martin has called the U47 his favorite microphone, and used them to record almost everything on *Rubber Soul*. The harmony parts on "Nowhere Man" are a great example of that 4 kHz U47/48 zing!

By the time *Revolver* was recorded, the Beatles had so much clout that they got what they wanted, and what they wanted was to experiment with new techniques. Studio administration goons were banned from the sessions, and Geoff Emerick was brought in as lead engineer. You can clearly hear the Beatles singing closer to the mics. Compare Paul's count-in on "I Saw Her Standing There" from the debut album with George's muttered count-in on "Taxman". The proximity effect is obvious. Vocals on *Revolver* sound thicker and richer than on previous albums.

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Emerick has said that the string players on "Eleanor Rigby" would instinctively back away from his very closely placed U47s between takes, so he had to keep walking out to move them back in! Things kept getting closer and closer. If you watch the "All You Need is Love" satellite broadcast, you will see John Lennon singing as close as an inch away during the quieter verses (using a pop filter (or "spit screen" in the UK)) and moving back about a foot for the louder choruses. He sounds warm and present up close and perfectly blended with Paul and George when he backs away.

The key to rocking a U47 is to work that distance! Use a pop filter so you don't hawk loogie on the diaphragm. Then try singing loudly and softly at different distances. A distance of 1 foot or more works well for background vocals because the proximity effect is not so significant. You will get a clear sound that will not muddy the mix. For lead vocals you can move in and out depending on how loud you are singing. For consistently soft or low parts you can lean in really close, maybe only an inch or two away, then back off for louder passages. But for consistently loud and aggressive stuff you can also just stay about 2 or 3 inches from the mic. You will be feeding lots more level into the tube, which *loves* to be saturated this way. You might need to dump a bit of low end with EQ, but once you do you are left with a sound that has become a mainstay for loud rock singers. Don't forget that the frequency response changes with the polar pattern, so try them all!

If money is no object you can buy an original U47 for maybe $10,000. But for closer to $5000–$6000 there are emulations out there that some engineers prefer to the originals. Telefunken is still around and has re-introduced their version, but also check out mics by Wagner, Soundelux, Manley, and Bock. In the $1500–$3000 range consider offerings from Pearlman Microphones, Lawson, Peluso Microphone Lab, Flea, Audio-Technica (notably the tube cardioid AT4060), and Neumann's own M147, a fixed cardioid, tube, transformerless U47.

Discouraged? Don't be! Consider this quote by the great Alan Parsons: "I've certainly spent many hours with finicky artists trying different vocal mics, all of which sound remarkably similar, and all I have to say is that I felt it was a waste of time." I can tell you that I have done classroom shootouts with U47-style mics priced between $300 and $3000 and found that they do in fact sound remarkably similar. I can personally recommend Lewitt microphones. Between $600 and $1000 they have four mics with a perfect double hump response, including the LCT540 FET fixed cardioid, the multipattern LCT640 FET (see Figure 7), the tube LCT 840, and the remarkable Tube/FET LCT 940 which has a knob to blend the Tube and FET sounds! I feel this is one of the great mic bargains of all time.

For a bit less, consider the RØDE NT2-A, NTK, and K2. The M-Audio Sputnik tube mic costs under $500. At $299 it's hard to beat the Cascade Elroy multipattern tube mic, and for less than $200 you can pick up the sE Electronics sE X1 and the Cascade M20u, solid state mics that are generating positive comments on user forums. (If you are truly adventurous there are mysterious grey market options like the Stellar CM-6, but I didn't tell you that...)

Notable successors to the U47 were the tube U67 and the solid state U87, both using the K67 capsule, with two backplates to facilitate better matching of the front and back tensioning. Both have a flatter frequency response than the U47. Another great classic worth mentioning is the Austrian AKG C12. Early versions had a very U47-like response, and later versions remained pretty flat up until a few decibels of boost around 10 kHz. The C12 became very popular for female vocals because of its flatter response. AKG currently makes the C12VR, and more affordable emulations are available from Peluso and Lawson. An interesting way to cop this type of sound for peanuts is to buy an Apex 460 and modify it using kits available from a handful of suppliers. Kits are also available to modify low cost mics like the MXL990. Google is your friend...

So go forth and experiment! The sidebar has a short list of some well known tracks and the mics that were used to record them. Happy tracking!

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Famous vocals and the mics that captured them...

AKG C12: Michael Jackson–"We Are The World", Jamiroquai–"Supersonic"
Electro-Voice RE20: Paul McCartney–"Uncle Albert/Admiral Halsey", Radiohead–*From The Basement*
Neumann M49B: Norah Jones–"Don't Know Why"
Neumann M147: Gotye–"Somebody That I Used To Know"
Neumann U47: Michael Jackson–"Black Or White", Beyonce–"If I Were A Boy"
Neumann U67: Feist–"1234", The Beatles–"Hey Jude"
Neumann U87: John Lennon–"Imagine", Avril Lavigne–"Complicated", "I'm With You", "Sk8ter Boi", Marvin Gaye–*What's Going On*
Neumann TLM 103: Simon LeBon (Duran Duran)–*Astronaut*
RØDE Classic: Adele–"Rolling in the Deep"
RØDE NT1-A: Amy Winehouse–"Valerie"
Shure Beta 58: Bono (U2)–*No Line On The Horizon*, Trent Reznor (Nine Inch Nails)–*The Downward Spiral*
Shure SM7: John Mayer–"In Repair"
Telefunken ELA M 251: Jason Mraz–"I'm Yours", Deftones–"Diamond Eyes"

REViEW

BY PAUL VNUK JR.

Neumann U 47 fet Collectors Edition

A renowned classic returns, from the place where it was born

One of the most exciting announcements at the AES Convention last October was that Neumann was reissuing one of its classic microphones. The microphone in question was the U 47 fet, and it was going to be a spec for spec re-release of the original, down to the last detail.

With few exceptions, beyond boutique manufacturers "inspired by clones," the legacy companies such as Neumann, AKG and others have never attempted a reissue with this much authenticity. Usually we are treated to remakes or similarly named models with new suffixes added. What that usually means is a recreation of a classic microphone with "improvements" or "enhancements" (quotes intended). But according to Neumann: Not this time!

Once upon a time...

...there was a microphone named the U 47, a tube microphone of some distinction with two selectable polar patterns (cardioid and omni). It was born in Germany in 1947 and used Neumann's M7 capsule and a Telefunken VF14 steel tube. It went on to become the most famous and sought-after microphone of all time.

In 1959 Neumann switched from the original PVC-based M7 capsule to a new K47 polyester/mylar capsule, technically known as the K49 capsule originally. It was a wonderful and worthy successor and is still in use today, now known as the K47/49 capsule.

By the mid 1960s Telefunken ceased production of the VF14 tube and despite varying attempts at replacements such as the NuVistor tube, by 1965 the U 47 was no more. It was soon replaced by Neumann's new tube model, the U 67, which entered into the line in 1960 and was discontinued in 1971.

However, in 1972 Neumann released a new microphone bearing the famous 47 numeration. It used the same K47 capsule and head basket but it featured a new, smaller body. Inside was not a tube-based circuit, but a solid-state FET design, and thus the Neumann U 47 fet was born.

While it sounded very little like a classic U 47, it eventually established itself based on its own sound and it remained in the line until 1986.

One other historical note: before the U 47 fet, Neumann released a solid-state version of the U 67, the well-known U 87. Introduced in 1967, it remains in the line to this day (known as the U 87AI) and is largely considered Neumann's current flagship model. Well, until now...?

A classic returns!

It has been 29 years since the U 47 fet graced the Neumann lineup. Let's open up the wooden display box, set aside the enclosed certificate of authenticity, and take a closer look at this new version to see how well Neumann has succeeded in recreating the past.

The U 47 fet is a cardioid condenser microphone—actually a tad more supercardioid by today's standards. Its K47 capsule is a two-sided, dual-membrane capsule made of 6 micron thick gold-sputtered mylar. It is a center-terminated design, has a 34mm back-plate, and only one side is active.

As on the original U 47 fet, the capsule uses a plastic tensioning ring that contains one of the very few modern-day alterations to the design. This is the addition of a nylon bumper on the top of the capsule. This in no way affects the sound, but serves to protect the capsule from harm, should the microphone fall from a substantial height. (Sound unlikely? Look at the photos later in this article.)

The capsule is housed in a 3-layer mesh grille that is dimensionally the same as the original and physically the same as late-period models.

The mic measures 6.3" with a diameter of 2.48". It weighs just over 1.5 lbs. It is dressed in a brushed nickel finish with a purple Neumann badge. On the front side of the mic is a black etched cardioid symbol. An interesting fact here is that I have seen early vintage models with this symbol upside down (although history may argue that it is upside down now!), and I have also seen vintage models that look like the reissue.

One of the unique features of this mic is an attached "L" bracket/mounting arm with a washer/screw assembly that allows the mic to tilt forward or back. This right-side bracket terminates in a mic stand mount and also contains a black plastic cable lock. It's pretty small, made at a time when mic cables were apparently thinner.

The mic contains two recessed switches on its rear side. One engages a –10 dB input pad, the other is a lowcut filter at 140 Hz. On the bottom of the mic, along with the XLR socket, is an additional –6 dB pad switch, but this time on the output. All three switches require a very small flathead screw driver to adjust.

Guts and glory

Internally the mic has a transformer-coupled output and point-to-point wiring. Compared to today's design standards, the innards are downright messy... but they're completely authentic.

After years of reviewing modern mics with simple dual bracing, vertical circuit boards, and attached capsules, the inside of the U 47 fet is impressively compartmentalized. It reminds me of a multistage rocket assembly with a detachable capsule, octagonal internal posts, a huge custom transformer, and rugged construction throughout.

It has a frequency range of 40 Hz to 16 kHz, a 8mV/Pa sensitivity, an 150 ohm rated impedance, 1 kilohm rated load impedance, an equivalent noise level of 25 dB CCIR and 18 dB A-weighted, a 69 dB CCIR/76 dB A-weighted signal-to-noise ratio, and a maximum SPL of 147 dB (attenuated).

The frequency plot shows the mic to have a low end that slopes down from 500 Hz to 5 dB down at 40 Hz, a mid rise starting at 1 kHz and peaking at 4 dB at 4 kHz, and then dropping sharply down before rising again to 4 dB at 9 kHz and then sloping quickly down –5 dB at 16 kHz.

Taken on its own merit

Before I got into historical comparisons, I worked with the U 47 fet in my studio over a few weeks. I tracked the usual "rock flavored" subjects: drums, bass, guitars, percussion and voice. I also compared it to a broad range of mics, including a Milab DC-96b, Brauner Phantom, Shure KSM44, and Audio-Technica AT4047MP.

The AT4047MP was more imposing with a bigger bottom end. The Phantom was the crispest and most open. The Milab sat nicely in the middle with a neutral low end, a fuller upper midrange, and a hint more high end, and the KSM44 was the most even and rolled-off sounding of the bunch.

Within this group of mics, I would call the U 47 fet sculpted and unimposing. Its low end especially comes across as open and unhyped. Its upper midrange, where vocals and acoustic guitars live, was gently forward and a little hard to describe—natural, but not open. It has a subtle mid-pop to it rather than bite, and this mic could not be sibilant if it tried. It's actually a sound that makes you notice the source itself rather than the mic coupled with the source.

This microphone's claim to fame over the past 30 years has been as the ultimate classic kick drum mic. I think part of that is due to the fact that its subdued bottom end makes room for the natural lows of kick drums and bass guitars to shine through, vs. mics that accentuate the low end to sound larger than life and forceful.

The U 47 fet has a gorgeous proximity effect when a source is one to two inches from the mic, but it is one that rolls off quickly. This makes it wonderful for spoken word and intimate forward lead vocals. It can also yield a very effortless sound on guitar cabinet for similar reasons. On the other hand, it was a tad too "dry" for my tastes on acoustic guitar and drum overhead/front of kit. Not that it was unusable, I simply liked the other mics better in those applications.

Old vs. New

Of course, when a company reissues a classic and claims it to equal the original, the next step is to prove it! To aid me in this part of my review, I traveled south to the Windy City and the Chicago Recording Company, one of the largest professional studios in the Midwest. They happen to own not one but two vintage specimens, and we spent an afternoon comparing the trio side by side. General Manager Chris Shepard set the whole thing up and put me in the hands of the studio's gear junkie in residence, Dennis Tousana, who lined up people and sources to compare the mics.

Since the U 47 fet is best known for its use on kick drum, bass cabinet, trumpet, and voice, that's what we recorded. I have to say that the U 47 fet on trumpet highlighted this mic's upper mid rounding; it instantly made the trumpet sound like a classic R&B or jazz album. I also liked how well the mic worked at a distance on kick and bass cab, maintaining feel and depth without loosing the tone. Classic rock tone on tap! The singer (also the trumpet player) had a very gospel/R&B style voice that showed how well this mic would do on vocals in those genres as well, giving a round non-sibilant sound.

All three mics were tracked through a 4-channel John Hardy M1 mic preamp at 96 kHz into Pro Tools HD while monitoring through an SSL 6064E console on the studio's custom TAD 3-way mains. In the room were Dennis, engineer John Zacks, studio technician Bruce Breckenfeld, and intern Rob Turner. I asked Dennis if he knew the manufacture years of the vintage mics, and all he said was that they were already there when he started in the early 1980s!

One of the studio's vintage models did seem to have a tad more air on top compared to its sibling. This was most noticeable on the vocal pass, possibly due to the singer's placement to that mic vs. the others. The other vintage model alongside the new version were all but indiscernible.

The final verdict among all of us was that any perceived differences between the new and vintage models was no greater than that between the vintage models on their own. Overall each of the mics had the same flavor, weight and feel.

Conclusions

The verdict: Neumann nailed it! This is not a recreation, a replica, or a tribute. It is a real Neumann U 47 fet in every way from its build to its sound. It is one of those mics with a true classic sound and niche all its own, but on all of the sources for which it is legendary... yeah, it is truly legendary! To my ears, this applies to brass even more than kick drum.

At $3999 street, this is a serious investment, but one that stands toe to toe with its vintage counterparts in every possible way, and Neumann needs to be applauded for achieving this level of authenticity. Wow. Just wow....

Price: $3999.99

More from: Neumann, www.neumann.com

Paul Vnuk Jr. (paul@recordingmag.com) is a recording engineer, live sound engineer, producer, and musician, living and working in the Milwaukee area. Paul would like to thank the staff at the Chicago Recording Company (chicagorecording.com) for their assistance: Chris Shepard (General Manager), Dennis Tousana (Engineer), John Zacks (Chipotle and ears), Bruce Breckenfeld (Studio tech), Bernard Chae (bass player), Rob "Rob The Intern" Turner (setup and kick drum), and Leo Q. Allen (trumpet and vocals). Special thanks to Klaus Heyne (germanmasterworks.com).

Mic Maestro Klaus Heyne Weighs In

Klaus Heyne of German Masterworks is one of the foremost microphone technicians in the world, and one of the most knowledgeable people around when it comes to the subject of Neumann microphones, both old and new.

Recently on his online forum, he did a complete teardown of an original U 47 fet as well as the new version, and he was kind enough to answer my questions as to this new model's authenticity from a build standpoint. The following contains excerpts of a much longer interview with additional insights, which can be read at our website: http://is.gd/KlausHeyneRECFeb2015

The first and most obvious question, how did they do? Did they get it right?

Klaus Heyne: In a word: 100% right. [...] From the machining to the plastic molding, from component choices to the careful assembly, it is astounding to me what is going on here: never before has Neumann reissued a vintage or classic model with such perfection and attention to detail.

Anything where they missed the mark or any obvious differences?

With one exception, no detail or mark has been missed or glossed over. Even the messy layout of the high impedance amplifier board was meticulously recreated.

The one exception where the company thankfully did not copy the original: the slider switches. The originals were made of brittle plastic which by now has deteriorated in most of the mics, rendering switching functions inoperable. But even here, Neumann paid attention to detail: while the new switches are not interchangeable with the original version (of hundreds of components, the only part, by the way, that you cannot use on an original U 47 fet!), the externally visible switch actuators look identical.

Any idea why Neumann went with a hard mounted swivel arm vs a shock mount or even the old base swivel designs?

Why attempt to reproduce a model in all its minute details, and then decide to "improve" on the original mount, which was never an obstacle to recording in the first place? The elegant swivel arm is an integral aesthetic part of the original mic design. Besides, the U 47 fet is rather bass-shy to start with, so transmission of subsonic vibrations from the floor is not a major issue here.

What is the purpose of the locking cord holder which looks way too small for a mic cable?

The standard 5mm ø Neumann IC 3 mic cable fits fine. But the point is well taken: I don't believe I have ever seen anyone use the cord clamp on the original, either. So what is the purpose of mounting it in the first place? Adherence to original detail.

I noticed that unlike many modern mics, including most of the other FET47 clones, the capsule sits very high in the basket and is transected by the top bar of the grille. Is this by design and does it affect the sound?

The positioning of the capsule at that level has one intended and one unintended outcome. The elastic capsule mounting column has to have a minimum length, in order to be effective as a damper of acoustic energy and mechanical shock. If you make the column too short, in order to "properly" position the capsule in the basket, you get the nasty effect familiar from the AKG C12: huge low-end bumps with even the smallest of mic movement.

The unintended, but certainly welcome, secondary effect of placing the capsule where it is: unpleasant sound reflections and standing waves between the capsule diaphragm's surface plane and the level (plane) section of the grille in front are minimized if the capsule is pushed up where some of the reflections are diverted and randomized by the dome-shaped upper grille section and the support frame of the basket.

Any final comments on this new reissue?

I applaud Neumann for finally, courageously "revisiting the past" with such passion and meticulous attention to detail. Damn the bottom line! [...] The U 47 fet has a huge personality that one can easily embrace. Let's hope that this is only the first of many more of Neumann's revisitations of the past.

Processing Vocals—Vintage To Modern

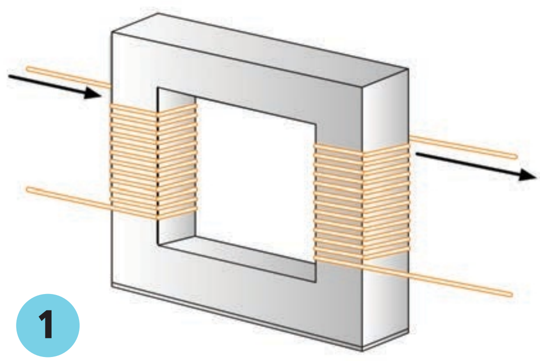
By Michael Schulze

From famous beginnings to affordable plug-ins, here's how to massage your vocal tracks

A vocal leaves the mic and becomes a track in your DAW or on tape. Between here and there, it goes through preamplification, EQ, and compression, and this signal processing does much to define its character and beauty. Understanding what each stage does, with some historical background and some technical insight as to how these devices work, will help you zero in on the vocal sound that's best for a given track.

In the beginning

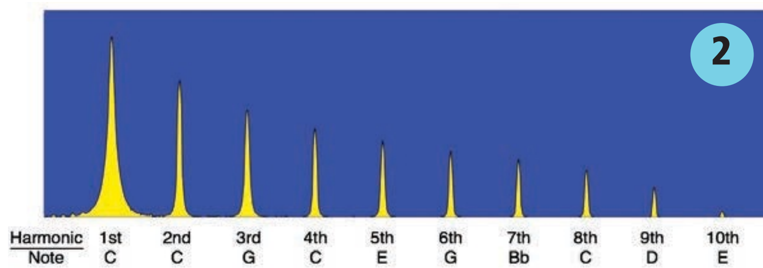
Way back when, there was no such thing as an outboard microphone preamplifier; preamps and some simple EQ were provided by the studio mixing console. The first consoles used vacuum tube circuits—for example, all of the Beatles albums before 1964 were recorded through the REDD.37 console at Abbey Road, which used German Siemens V72 preamplifier modules for all active circuitry.



In the V72 circuit, the microphone input is connected to a *transformer:* a square or round donut-shaped hunk of iron wrapped in two coils of wire (see Figure 1). A signal in the first or *primary* coil generates a magnetic field that envelops the second or *secondary* coil, causing current to flow out of it. The input transformer converts the 3-wire balanced mic signal into a 2-wire unbalanced signal while canceling out electromagnetic interference. In some preamps, the input transformer also provides the first bit of gain. The signal then goes through two tubes, some resistors and capacitors, and then to the output transformer, which re-balances the signal for output to the next module in the console.

Transformers don't faithfully pass extremely low-level signals, and saturate at extremely high levels, especially at low frequencies. This results in a subtle distortion of the sound that is quite pleasing, with a fattening of the lows, a silky smoothness in the highs, and a punchy complexity in the midrange. Audio through great transformers sounds larger than life, as if the Rock And Roll Knob has been turned up. In a vintage console there might be as many as ten transformers in the signal path from input to output, each one adding more mojo!

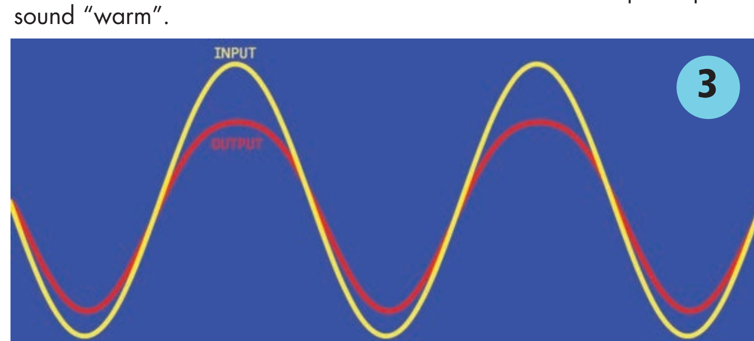
This "mojo" has to do with harmonics. When any musical instrument or voice produces a tone, you are hearing multiple frequencies. When you play A440 on a piano you are hearing 440 Hertz (the *fundamental)* plus integer multiples of 440 Hz: 880 Hz, 1320 Hz, 1760 Hz, 2200 Hz, etc., in varying proportions. Some musical instruments produce spectra where the fundamental and even harmonics predominate, and some produce spectra where the fundamental and odd harmonics predominate. This relationship is what makes a guitar sound like a guitar and an oboe sound like an oboe. Figure 2 is a chart showing the frequency spectrum of a sawtooth wave as you might hear in a trumpet; the harmonics correspond to musical pitches.



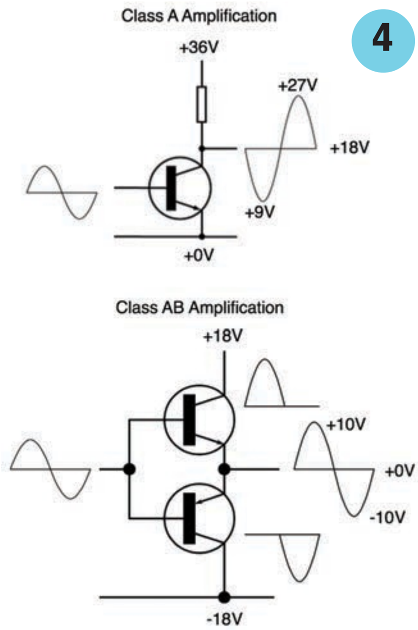
Any analog audio circuit distorts somewhat, subtly changing the balance of harmonics. Transformer mojo is mostly an increase in the third harmonic, an octave plus a fifth above the fundamental. This hints at the delicious tonality of a guitar power chord! This effect can add "richness" or "girth". Vacuum tubes also color your sound. When the signal gets loud enough, the tube progressively rounds off the peaks of the waveform (Figure 3), resulting in a boost of even harmonics. This reinforcement of octaves adds smoothness, or warmth; clarity without brightness. You also get light compression, where the amplitude coming out of the tube does not increase as fast as the amplitude going in. At the same time, though, the harmonics are increasing, making up for the loss in loudness by adding complexity, like turning down your guitar volume while turning *up* the distortion. The signal *sounds* louder. The combination of transformers and tubes adds complex color to the sound!

From tubes to transistors

Tubes are bulky, generate heat, and need periodic replacement. Console manufacturers switched over to transistors as soon as possible. It is a common misconception that tubes sound "warm" and transistors don't. It depends on the overall circuit design! There are tube preamps out there that sound clean and non "tubelike" and transistor preamps that sound "warm".



In the 1960s, British engineer Rupert Neve designed his first solid state mixing console, with transformers at the input and output of every stage and transistor circuitry in between. Like earlier tube equipment, this console operated in *Class A*. A tube or transistor can only handle current in one direction, so the signal voltage must stay between 0 and some other, usually higher, voltage. This is class A. In *Class AB* the signal swings from positive to negative using two tubes or transistors, one handling the positive and the other handling the negative swing. (See Figure 4.) Class A eats more power and generates lots of heat because the circuit is always passing current. Class AB circuits run cooler and eat less power, but suffer from a bit of *crossover distortion* at the point where the signal switches between devices. Class AB does not necessarily sound inferior, but a well-designed Class A circuit can claim lack of crossover distortion as one of its sonic merits.



The best challenge to the "tubes are better" line is to just say, "Neve." Neve circuits exhibit a significant level of delicious harmonic distortion, mostly from the Carnhill transformers used. You can hear it on thousands of hit records from the 1970s up to the present day! Check out Dave Grohl's documentary *Sound City*. Thank me later.

EQ

After the mic preamplifier comes the equalization circuit. The REDD consoles had interchangeable EQ units labelled "Classic" and "Pop" to be used appropriately—a very British attitude, what? Separate boost and cut knobs for lows and highs provided shelving response except for the "Pop" high boost: a peaking response around 5 kHz. Introduced in 1951, the American Pultec EQP-1 program equalizer had a transformer input followed by *passive* equalization circuitry (no power supply in the EQ section), feeding an interstage transformer to a Class AB tube gain makeup circuit and output transformer.

The standalone EQP-1A was intended mainly for broadcast and vinyl mastering. It had separate high and low boost and cut knobs with click-stop selectable corner frequencies. There was a variable "bandwidth" control for the high boost, but the curves produced were broad by today's standards. The EQP-1 sounds warm and punchy, with an uncanny high-end smoothness. Many engineers run *everything* through an EQP-1, because it sounds awesome... even with the EQ flat!

The early Neve 1073 EQ module had high and low shelving click-stop corner frequency selection and a peaking midrange band. The fixed bandwidth and center frequency of each click stop were selected by the golden ears of Mr. Neve himself. The later 1081 EQ module (see Figure 5) added a second mid section, a Hi Q (narrow bandwidth) switch for the mid bands, and high and low pass filters. The Neve is difficult to describe because there is noticeable warmth but also a slightly more aggressive upper mid character with that very detailed midrange. To me Neve sounds present, lively, and snappy—it sounds fantastic before you even touch the knobs!

Processing Vocals—Vintage To Modern

Compression

Early analog consoles did not have built-in compression, so compressors were outboard, intended more for the full mix than for individual signals. Broadcasting was monophonic in those days, so most compressors were too. The Holy Grail units are the Fairchild 660 Limiter (and the 670 stereo version), the Teletronix LA-2, and the Universal Audio 1176.

The Fairchild 660 tube limiter has controls for input level, threshold, and a "time constant" click-stop knob, which provides different combinations of attack and release times, as fast as 0.2 and 0.3 milliseconds respectively. In the extreme settings, release gets as long as 25 seconds! The non-adjustable ratio starts at 2:1 and gets as high as 30:1 depending on how much gain reduction is happening. The compression is done directly by the tubes using the "Variable Mu" technique.

Like a Pultec EQ, the 660 improves the sound even when doing nothing, thanks to its 20 tubes and 14 transformers. Many engineers use this box to barely compress vocals, with the time constant at the fastest settings and the needle barely moving—the 660 can get rather squashy if you let it. Engineer Geoff Emerick would run just about all Beatles vocal tracks through it, even without engaging any compression, just for the enhanced presence. I don't have a 660 photo handy, but Figure 6 shows the Waves plug-in version from the Jack Joseph Puig package—we'll talk about plug-ins later on.

The LA-2 (see Figure 7 for the still-available Universal Audio LA-2A) is a different beast entirely. It is an all-tube design with transformers, but the compression is controlled by an electroluminescent (EL) element that gets brighter as the audio gets louder. This is why the LA-2 is referred to as an "optical" compressor. This light shines on a photosensitive resistor that controls the compression. The resistor will provide a 10 ms attack time, but when the audio gets quiet and the light goes out, the resistor exhibits a two-stage release time, falling in 40 to 80 ms to 50% of its "off" resistance, and then the rest of the way over several seconds. Additionally, the photo resistor has "memory", and takes even longer to reach its off resistance depending on how bright the EL element had been shining and for how long.

Therefore the LA-2 has *program dependent* release, recovering quickly after a few loudly shouted words and slowly after long passages of sustained loudness. This is a very desirable attribute! The compressor will act quickly on short-term program material but also gently ride the level over longer sections. Skillful use of program dependent compressors can reduce the amount of level automation needed in your final mix.

The 1176 (see Figure 8) is a solid state, class A compressor with the compression controlled by a FET (Field Effect Transistor) circuit rather than an optical device or tubes. The attack time is ultrafast, 20 to 800 microseconds! The release time is slower, 50 milliseconds to 1.1 seconds, with some program dependency. (By the way, the 1176 attack and release times get *faster* as you turn the knob to the right.)

The 1176 adds a little edge to the sound, which makes it extremely popular for aggressive rock bass and guitars as well as vocals. The secret mojo of the 1176 happens when you press all 4 ratio buttons down at once and get them all to stick. This was not an intentional part of the design, and is sometimes called "British mode" due to its popularity among some British engineers in the 1960s and 1970s. In British mode the compressor basically freaks out; distortion increases, ratio moves around between 12:1 and 20:1, attack time is magically delayed, and release time alters fluidly. Nobody has ever actually documented exactly what happens because it is very complex and chaotic!

British mode can be used for drastic audio smashing. Listen to Audioslave's self-titled debut: extreme vocal compression with a slowish attack time and fast release. This makes the beginnings and ends of the words pop out, which improves intelligibility. Aside from that, the vocal is held at a very constant amplitude with some added edge, which is magic in a loud mix.

You can achieve this sound easily with an 1176. Press in all the ratio buttons, and the gain reduction meter will slam all the way to the right—*wham!* Set a slower attack time, 1 to 3, and fastest release (7). Increase the input knob until the meter shows 10–20 dB of compression. You'll need to decrease the output by a lot!

Use your ears and tweak the input and attack/release knobs until the beginnings and ends of the words pop and the sustained syllables remain at a very constant level. You can use this same trick on soft vocals in an exposed folk mix. It just involves speeding up the attack time so the beginnings of the words even out, and slowing down the release a bit. You can end up in a sweet spot where the vocals become very close and intimate without actually sounding supercompressed.

Putting it into practice

First of all let's pretend money is no object—we'll get to more affordable alternatives after we describe the method, which involves getting some iron and maybe tubes into the chain. Chandler Limited makes a line of outboard gear based on the REDD circuitry, plus some other Beatles-era processors not mentioned in this article. You can get hold of vintage

Neve preamp/EQ modules pulled from old consoles, or buy something new from AMS Neve, BAE, Aurora Audio, or Mr. Neve's current firm, Rupert Neve Designs. Also consider API, whose circuits were contemporaneous with and similar to Neve's, but with an original, differently colorful sound. Also prepare to be delighted by the kits available at seventhcircleaudio.com!

Dial in your gain and engage some EQ, perhaps a 60–80 Hz highpass to get rid of some of what gets past your pop filter. A little tweak in the high mids might be in order: a slight dip around 3–5 kHz for an overly edgy voice, or maybe a little bump around 500 Hz to warm things up.

You might also like a little high shelf boost if this is the last EQ in your chain. Be gentle here! If you have a nice mic and a good singer, you need only subtly enhance what is already there. Don't be afraid to leave the EQ flat—these units sound great just passing signal!

The next stage can be your compressor. You can buy re-creations and even kits based on the Fairchild, Teletronix, and UA compressors. Universal Audio still sells authentic 1176 and LA-2A hardware at reasonable prices.

Here's a very common approach to working on vocals with these units. First run the signal through an 1176 at 4:1 with attack at about 3 and release at about 7, with the meter barely moving, maybe only 1 to 3 dB of gain reduction. This will almost inaudibly shave off the quickest peaks. If you prefer, at this stage you can do the British mode squash if "that's your bag, baby." Patch the output to an LA-2 tube compressor or a 660 on one of the fastest settings, and again let it compress only a few dB. The combination of the quick 1176 and slower tube compressor can level out your vocal nicely with tons of character. You might get to a sweet spot where your vocal track needs little or no level automation!

After all this, run through your Pultec EQ and dial in a little subtle high shelf boost at whatever frequency suits your taste; 10 or 12 kHz are good choices. This will add that Pultec sheen, more iron/tube mojo, and make up for the warming effects of some of the upstream units.

This chain was quite common in the pre-DAW days. It requires skill to get everything right, as you are printing everything to tape as you go! But if you keep your EQ adjustments to only 2 or 3 dB and prevent your compressors from doing more than 2 or 3 dB of gain reduction, you'll be golden.

Pay attention! If the singer gets too loud or soft, consider retracking some sections a bit closer to or farther from the mic. Or you can patch your preamp directly to your DAW and put everything else in an insert loop, so you can always revert to your original unequalized/uncompressed signal.

Processing Vocals—Vintage To Modern

And if you can't afford $4000 to $30,000 per channel...

"But Mike—I don't have access to all this expensive studio bling!" Well, fear not! Consider a more affordable preamp. Radial Engineering offers the PowerPre and PowerTube 500 Series modules, which combine classic circuitry with transformers at a very easy price.

You can achieve the rest in software. Waves offers plug-ins based on most of the hardware mentioned in this article. Other highly-regarded software comes from Bomb Factory, Universal Audio (running on their UAD-2 external hardware DSP engines), PSP Audioware, IK Multimedia, and many others. Even some of the freeware plug-ins out there sound great.

Apple Logic Pro X's compressor (see Figure 9) has a dropdown menu to select emulations of FET, Optical, and the more modern and linear VCA (Voltage Controlled Amplifier) topologies. This and many other plug-ins have an automatic release feature, which provides program dependent release like the LA-2's.

Set up two compressor plug-ins in order and use the "expensive hardware" settings described above. First is the 1176, select a 4:1 ratio, fastest attack time, and 50 ms or faster release. Adjust threshold for only 2–3 dB of gain reduction and boost output gain by about the same amount. If you want British squash (and the plug-in doesn't have a built-in British mode switch), set the ratio between 12:1 and 20:1, attack and release times around 50 ms, and engage auto release if available. Set the threshold to achieve 10 to 20 dB of gain reduction at all times, and adjust attack and release as necessary.

On the second compressor, set a 2:1 ratio, 10 ms attack time, and release time anywhere from 50 to 150 ms. Again allow only 2 to 3 dB of gain reduction. Follow this with a wide 2 to 3 dB high shelf boost at about 10 kHz, and you have your own sonic time machine!

Another trick is to set up two compressors as described above, but let the first one get farther into gain reduction. Then set the second, slow-release compressor to do the other half of your gain reduction (maybe 6 to 10 dB total between the two—whatever you normally do). Turn on auto release on the second compressor if available. What you have now is an emulation of the complex curve and release time of an LA-2, with faster action followed by a slower release when things get quieter. To fake a 660, set the first compressor's ratio to 2:1 and the second to 30:1, and watch the meters carefully.

In the end, it's amazing what you can do with good ears and average tools. I'm not saying Logic Pro can sound exactly like a $30,000 vintage Fairchild, but I'm also not saying it has to sound *worse*. Even if you have no real iron or glass in your signal chain, you can hang with the big boys and girls when capturing your vocals. Have fun!

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REVIEW

BY PAUL VNUK JR.

Audio-Technica

AT5045 Cardioid Condenser Instrument Microphone

The groundbreaking diaphragm technology of the 50 Series gets small, compact, and flexible

Back in May of 2013, I had the pleasure of introducing readers to a cutting-edge microphone from Audio-Technica known as the AT5040. The AT5040 was built from the ground up with a no-holds-barred, "money is no object" philosophy, with the goal of creating a top-shelf vocal mic that relied on new design principles and technology. The most unique design element of the AT5040 was its large rectangular diaphragm that was actually made up of 4 smaller rectangular diaphragms, combined via onboard electronics into one large "super-diaphragm".

Now the 50 Series, which has superseded the already amazing 40 Series as Audio-Technica's premium line, truly becomes a series as the AT5040 is joined by the new AT5045 Cardioid Condenser Instrument Microphone. Here's what I learned in my time using the AT5045 in a variety of settings.

The quadrilateral, part 2

Like the AT5040, the AT5045 is built around a rectangular diaphragm, but just one this time. The diaphragm is 2 microns thick and measures $1\frac{5}{16}$" x $\frac{9}{16}$". The mic itself is just shy of 7" long by 1" at the base and is only a hair larger than the average small-diaphragm pencil condenser microphone. Its body is finished in the dark pewter gray scheme of the 50 Series, and its fine mesh grille at first put me in mind of "getting a closer shave than a blade"!

The AT5045 is a side-address microphone and, despite its compact size, its rectangular capsule offers more surface area than some traditional large-diaphragm microphones. The long rectangular diaphragm could easily fool someone new to the mic into thinking it was a ribbon.

As with the circuit design of the AT5040, the AT5045 features no switches or pads and has a minimal amount of circuitry. It is transformer-balanced on its output as well.

Why a rectangle?

Rectangular capsules actually date back to the 1950s, pioneered by Sweden's Pearl Microphone Labs and are still in use today by Pearl and its offshoot company Milab. We have looked at models from both companies over the past 2-3 years. Now that Audio-Technica offers its own mics of this design (although internally quite different than the Swedish mics), the rectangular capsule may finally be its on its way to becoming a full-fledged category rather than an exotic oddity.

The rectangular diaphragm offers advantages over traditional circular designs. As I said in a past review, this diaphragm "disperses the extreme midrange resonance peaks inherent in circular capsules, and it allows for an exceptionally neutral off-axis response." As a real-world illustration, anyone who has ever tuned a drum head knows how difficult perfect tension and even tuning can be. Often, a drum head needs to have stray resonances damped down with gels, tape, or weights. A mic capsule can be similar, but usually its stray resonances are damped down electronically. Now picture tensioning a long rectangular sheet of metal or plastic—in essence, it is easier to get a tighter and more even tension from end to end.

The package

The AT5045 comes as part of a kit in a large deluxe molded luggage-style case and can be purchased singly or in matched pairs. Along with the mic, inside the case is a foam wind sleeve which fits over the grille but is open on both ends. Lastly the package comes with an AT8481 isolation clamp.

The AT8481 borrows heavily from the AT8480 that came with the AT5040 mic. It is very similar to the inner clamp found on the AT8480, but this one is smaller and scaled to the new mic. It's simply a secure clamp and not a suspension mount as on the AT5040.

The AT8481 is made up of two pivoting, spring loaded clamps that click snugly around the microphone and lock in place, thanks to a small adjustable lever. Although not technically a shockmount, the clamp holds the mic in place with small rubber washers that do provide some level of damping—and the capsule inside the AT5045 is internally shockmounted already. Overall the clamps are a nice balance of high-tech, low profile, and artistic beauty.

Specs

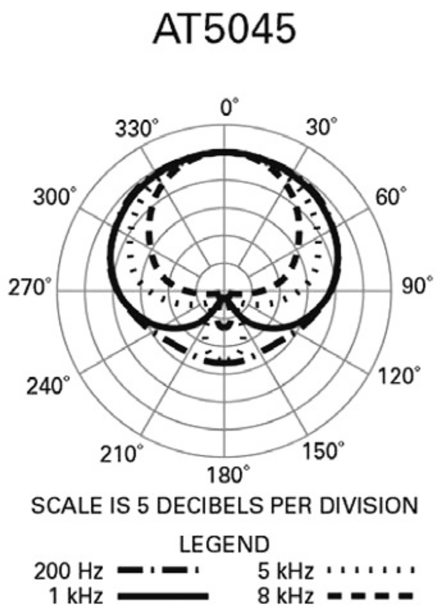
The AT5045 is a permanently polarized cardioid electret condenser microphone that uses a fixed charge backplate. It has a 20 Hz to 20 kHz frequency response, an open circuit sensitivity of –35 dB (17.7 mV) ref: 1V at 1 Pa, a 100 ohm impedance, a maximum input level of 149 dB SPL (1 kHz at 1% THD), 8 dB of self-noise, a dynamic range of 141 dB (1 kHz at Max SPL), and a signal to noise ratio of 86 dB (1 kHz at 1 Pa).

Looking at its spec sheet's frequency graph, we see a mic with a nice low end weight between 20 and 40 Hz followed by a dip between 40 and 200 Hz that cuts down about 5 dB at its most extreme. Its midrange from 200 to 1500 Hz is fairly even. There is a wide rise between 1500 up to 8000 Hz with its most significant 5 dB peak centered around 6500 Hz. There is also a small 2 dB bump around 12 kHz, and then the mic rolls slightly off.

To translate this into what we should hear, the AT5045 has a solid low end with the low-mid mud cleaned out, a natural middle, and an open upper-midrange presence boost. The high end has some minor peaks but is far smoother than many modern mics.

In use

Since I was sent a pair of the AT5045 mics, the first place I put them up was on a drum kit. Consider this a spoiler, but that was and is my favorite use of this microphone. The complex sound of drum overheads does well to highlight the narrow, even, and controlled sound of the AT5045. It fits like a hand in a glove, especially in multi-mic drum setups.

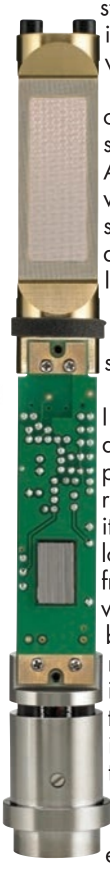


I highlight multi-mic drum setups because the tailored lows and low mids keep the mud of the floor tom and the boom of the kick out of the way, leaving the kick and tom mics to do their job nicely. Also, as an upper mid-forward mic, it has a nice rounded forward focus to the cymbals rather than sizzle or brashness. Finally, their capsule shape and tight off-axis rejection make them easy to position and aim.

For similar sonic reasons, they make great percussion mics as well. It may sound like I'm joking, but this is a very nice mic for tambourines and shakers. Record them about 4' back and you will get a natural forward sound that sits perfectly in the mix but does not bite!

Moving from the kit to bass cabinet, on-axis about an inch off the speaker grille it was more forward and defined rather than big and boomy. I liked it as a secondary mic in tandem with a Shure Beta 52 or Audix D6 for added upper-mid definition. On distorted guitar cabinet I preferred it more at a 3' distance vs. right up in the grille. At that distance it smoothed out some of the high-mid honk and ignored low-end buildup.

Although nowhere in the same league as its vocal-dedicated sibling the AT5040, the AT5045 makes a nice vocal mic as well. In this application, its upper mid boost pushes nicely through a mix. Here is where I noticed its proximity effect; when your lips are right up touching the included wind sleeve, you get a full-on Radio Voice! This rolls off quickly to an even response at distances greater than 2" or so. I did notice that the AT5045 is very prone to air blasts and plosives, as the capsule is quite exposed. I would use a standalone pop filter when tracking any sort of sung or spoken voice.



On acoustic instruments like violin, cello, and acoustic guitar, this mic shines for its focused definition. Again, it controls low-end buildup while accenting the round tone of the strings. Since its 12 kHz range is defined yet controlled, this is not what I would term a "sparkly" or "airy" mic. I preferred it on a plucked acoustic guitar more than an aggressively strummed one.

When I tracked acoustic guitar, I did notice one small thing to be aware of. The AT5045's polar pattern tends more toward omni response at 200 Hz and below; its rear response is very dark, pillowy, and subject to plosives even from 6" to 8" away. In other words, if the mic is pointing down by the 12th fret and anywhere near the player's mouth or nose, it can pick up breath blasts that thump into the mix. To be fair, this is not the first mic that displays this behavior; my beloved Neumann KM84s are notorious for errant air blasts from the extreme rear sides!

Conclusions

There are many things I like about the AT5045. First and foremost, it stands on its own with its own sound and area of best use. It really is a great focused instrument mic. Its depth of field is nicely tight and narrow and it has a tailored edge that rounds and smooths sources just a tad. I like it a lot on solo strings (both bowed and plucked), it's an easy mic to place in a mix, and it's one of the best overhead mics I've ever tried in a multi-miked drum setup. Keep it up, A-T... I can't wait to see what the next 50 Series mic brings us!

Price: $1399; AT5045P (pair), $2499

More from: Audio-Technica, www.audio-technica.com

REViEW

REVIEWS BY PAUL VNUK JR.

API, developer of the now-pervasive 500 Series module standard, has not rested on its laurels. In this pair of reviews, we'll take a look and listen to its newest modules: a simple and flexible DI and a tonally flexible filter bank.

505-DI

The API 505-DI is a 500 Series module dedicated to the direct input of Hi-Z and line-level sources. While that is a simple task that suggests a simple design and layout, the 505-DI adds in just enough extras to make it well suited to said task, while being quite versatile as well.

Its look, layout and design is perfectly in line with the rest of the 500 Series API family, dressed in matte black with standard, proprietary and unique API-style knobs in white and blue. It is originally based on API's 205L 200 Series direct input module, but with a few extra features thrown in—presumably due to the extra real estate available in the 500 Series form factor.

It all starts with a blue input/gain knob marked 0–10. The module offers a total 55 dB of gain; that's quite substantial for a line or instrument input, the sort of boost you'd expect to see in many microphone preamps. In case the incoming signal is too hot, which it was with both my active Fender Jazz bass and an acoustic-electric Takamine guitar, the 505-DI is equipped with a 20 dB pad switch and a 10-stage peak meter (–18 dB to +9 dB) for signal monitoring.

In addition to signal input, the 505-DI offers tone shaping and loading as well. First up, right below the pad switch is a matching load switch with a choice of 100 or 400 kilohms. It's easy to view impedance switching/loading as an "EQ" of sorts, but it does so by altering the load

Next to this is an 8 kHz Bright Boost that adds about 10 dB of high end. It works especially well on acoustic guitars, giving them a much-needed sparkle and string definition when recording direct. I like this definition on DI bass too, for when I want to hear the string plucks. Taken in tandem, the Tone control, impedance switching, and Bright Boost offer a simple but effective way to get your incoming signal sounding full and clear prior to adding EQ and further processing.

When used in the role of a DI, the 505-DI supplements its front-panel 1/4" instrument input with a 1/4" through/output for connecting back to an amp rig. In the "cool little extras" department, the 1/4" input has a blue backlight inside that glows when nothing is plugged into it, even when the unit's switched off via the yellow backlit power button.

Internally the 505-DI is a fully discrete design that makes use of both API's 2520 and 2510 op amps with a large transformer on the output stage. The transformer is in fact so large that it sticks out slightly beyond the module's enclosure, making the 505-DI a very tight fit in both my API lunchbox® and in Radial Engineering's Workhorse enclosure.

In use, the 505-DI was ultra clean with tons of gain, and I do mean tons. On every string-based source, with the exception of my '60s reissue Fender Telecaster, I needed to make use of the pad. This was also true when connecting my Moog Voyager and Dave Smith Prophet-8. My favorite source, and a lovely surprise, was my vintage Fender Rhodes 73; it typically has a very low output, but the 505-DI gave it ample clean gain.

What I like best about the 505-DI is that it sports the full API sound and circuitry through and

API 505-DI and 565 Filter Bank 500 Series Modules

A flexible DI and filters both musical and surgical, from the folks who gave us the 500 Series

placed on the pickup rather than through the use of filters. Unlike a standard filter, the load switch will cause each guitar and pickup to tonally react a bit differently. Overall I found the 400K setting to be a tad more open with a bit of midrange, vs. the 100K setting which was a touch fuller and darker.

The 505-DI also alters tone via a filter circuit labeled Tone, similar to the tone control on a guitar. When fully counterclockwise it is labeled Thin, 12 O'clock is Fat, and all the way to the right is Fatter. This Tone knob is a passive low-frequency shelving filter with a turnover frequency of 1600 Hz and a stop frequency of 600 Hz. When set to Fatter (fully clockwise) the signal is full-range, i.e. flat; moving to thinner settings cuts down the lows and mids by up to 10 dB.

through—op amps, transformer and all. This may not seem like a big deal, but often when instrument/line inputs are added to a microphone preamp, they will often bypass most of the circuitry that makes the preamp special in the first place. Not so with the 505-DI! Its price is right in line with most standalone powered preamp/DIs on the market. With its features, clean sound, and versatility, the 505-DI would be on the top of my short list for these studio tasks.

565 Filter Bank

As its name implies, the 565 Filter Bank contains a set of high- and lowpass filters as well as a 3-stage notching filter. Again the 565 has the API look, this time making use of the company's large pointer-style knobs, all dressed in blue.

The unit starts at the top with a 500 Hz to 20 kHz Low Pass Filter with a switchable slope of 6 or 12 dB/octave. Skipping over the Notch filter for a moment, the High Pass Filter at the bottom of the module has a range of 20 Hz to 600 Hz with a choice of 12 or 18 dB/octave slopes.

Both of these do exactly what they say, allowing you to remove unwanted high and low frequencies such as rumble or hiss from your tracks. Filters like this are also especially useful when tracking for getting rid of thumps, bumps, and both low and high frequency pollution.

Both filters are extremely smooth. The lowpass filter is one of the few I have used that can gently take out high end hiss and bite without sounding too muffled. Similarly the highpass filter is very transparent and cleans up the bottom end nicely. Much of this smoothness is due to the fact that neither filter is a resonant / peaking filter like those found in many comparable units.

The Notch filter is handy for zeroing in on single problematic frequencies, usually high mids that careen through your mix or low mid squonks. The Notch filter is an active twin T-notch design with a broad frequency range of 20 Hz to 20 kHz. Its knob is labeled as 20 Hz to 200 Hz, but it has a 3-position toggle switch that multiplies that frequency range by 1, 10, or 100. It also features a variable Q range of 0.95 to 15.3. This variable Q controls the width, and more importantly the depth, of the notch from about –16 dB down to –54 dB.

Like the 505-DI, internally the 565 Filter Bank uses the API 2520 and 2510 op amps, and it has the same discrete transistor buffers used in API's 550 Series equalizers. It is transformer-balanced on the output stage.

The 565 Filter Bank is a very handy device to have around. Unlike the fixed highpass filters found in most EQ modules, usually set at 80 Hz, a variable-frequency highpass filter can be a lifesaver in many mix circumstances. The same is true of having a variable-frequency lowpass filter for carving out only as much high end as is needed. It's hard to not think of the Notch filter as the real star of the show, however; it excels at carving out single problem frequencies with ease. With a very manageable price, a pair of 565 Filter Banks could make a great addition to any 500 Series setup.

API, developer of the now-pervasive 500 Series module standard, has not rested on its laurels....

Prices: $595 each

More from: API, www.apiaudio.com

REVIEW

Universal Audio

UAD-2 Satellite Thunderbolt and Manley Massive Passive EQ Plug-in

The Manley Massive Passive equalizer ("Massivo" to its friends) is one of those rare modern pieces of gear that is already considered a classic studio staple. Why am I reviewing a 15-year-old established classic, beyond the selfish motivation of "Because I really, *really* want to"?

A few years back, the audio alchemists at Universal Audio spent over six months painstakingly modeling every tube, wire, circuit, solder joint and more, to create the first and only authorized Massive Passive Plug-In for the UAD-2 line of DSP powered plug-ins. I held off reviewing it until I could get a real Massive Passive in hand for comparison purposes. Now, with the help of both Manley Labs and Universal Audio, I have been able to spend significant time with both units side by side in my studio. Here's the skinny.

Just what is a passive EQ?

While the Massive Passive looks at first glance like a standard 4-band active parametric equalizer, it's neither parametric nor active. A passive EQ uses filters made up of resistors, inductor coils and capacitors rather than op amps or ICs. Because of this, a passive EQ needs no electrical power to operate, and its filters can only cut/attenuate the signal. An active EQ can both boost and cut signal bands, and requires power to do so because it's providing gain.

...and a look at the real Manley Massive Passive, too!

The most famous passive EQs are those in the Pulse Technologies Pultec line. If you have seen or used them, in either hardware or software form, you may be wondering: If a passive EQ circuit only cuts a signal, what about the boost frequencies found on said units? The entire frequency range of the input signal is turned down before it reaches that EQ stage, and you're selecting a band to turn back up to its original level!

One drawback of a passive EQ is that signal output can be quite low after all that cutting with no gain stages. For this reason most passive equalizers add an active makeup gain stage before the output. True to Manley's "Tubes Rule" motto, the Massive Passive has a tube amp stage with 6 tubes inside: 2 x 12AT7 EH and 4 x 12BH7EH. This is not a "vintage colored tube tone" box; sonically the Massive Passive delivers a hi-fi tube clarity.

Parallel design

Unlike an active design, where one knob controls both boosts and cuts, individual EQ circuits and controls are required for each boost or cut on a passive EQ. Again, think Pultec, where the low cut lives right next to the low boost, and the two share many selected frequencies. This creates interesting tonal options, as the EQ curves overlap if you choose to boost and cut simultaneously.

In contrast, the Massive Passive bears zero resemblance to a vintage style passive EQ. As I mentioned, it looks like a typical active EQ, with the usual three choices per band of frequency, Q and boost/cut.

The Massive Passive has a parallel design where each band makes use of two completely different circuits, one for boosting and one for attenuating. They're stacked together onto the same controls, selectable by a Cut/Boost switch on every channel. The gain knobs start at zero and are turned up to increase boost or cut, unlike the center-detented gain pots on active designs.

Massive meet and greet

The Massive Passive is a stereo/dual mono equalizer with two matching channels on the left and right, along with additional controls in the center of the unit. Each channel contains 4 bands of EQ with identical control layouts.

From the top down, we start with two backlit toggle switches, one choosing boost or cut and the other selecting a shelf or bell curve—for all bands, not just the highest and lowest. This allows the Massive Passive to function like older vintage passive EQs,

with wide overlapping and interacting EQ bands.

Next is the boost/cut knob, with a throw of 0 to 20 dB. Then we have the bandwidth control; Manley's website explains that this is a more accurate name, since a true Q control requires an active op amp circuit. At 1.0 to 3.0, it is not as wide-ranging as a typical Q, and since it is very interactive with the amount of boost/cut, Manley suggests that it is better thought of as a damping or resonance control.

The last control on each band is an 11-position stepped switch with the following frequency choices:

Low: 22, 33, 47, 68, 100, 150, 220, 330, 470, 680, 1K

Low Mid: 82, 120, 180, 270, 390, 560, 820, 1K2, 1K8, 2K7, 3K9

High Mid: 220, 330, 470, 680, 1K, 1K5, 2K2, 3K3, 4K7, 6K8, 10K

High: 560, 820, 1K2, 1K8, 2K7, 3K9, 5K6, 8K2, 12K, 16K, 27K

As you can see, each band is very broad, allowing for numerous overlapping opportunities.

In the center of the unit are a High Pass filter with choices of Off, 22, 39, 68, 120, and 220 Hz, and a Low Pass filter with settings of Off, 18, 12, 9, 7.5, and 6 kHz. There is an overall continuous gain control for each channel with –6 dB to +4 dB of level adjustment, backlit bypass controls for each channel, and the power switch. On the rear of the unit are connections for both balanced XLR and balanced/unbalanced 1/4" TRS ins and outs.

There is also a mastering version of the Massive Passive (included in the plug-in), where the variable gain knobs are replaced by Grayhill stepped switches in 1/2 dB steps. Here the maximum boost or cut is 11 dB, and the high and low pass filters are tweaked for mastering: High Pass with settings of Off, 12, 16, 23, 30 and 39 Hz, and Low Pass with Off, 52, 40, 27, 20, and 15 kHz.

The hardware itself

A 3-rack space behemoth with a thick faceplate finished in the iconic "Manley Blue," the Massive Passive truly lives up to its name. The pots are buttery smooth yet firm, with great resistance so they stay put where you set them, all of the rotary switches are solid with a hearty click... it's a true showpiece from end to end.

This is how boutique gear should be made; it has a very audiophile build quality about it, not surprising since Manley got its start in hi-fi audiophile gear and maintains a hi-fi line today. All Manley gear is hand-built in a factory in Chino, California, and everything is done in house: circuitboard printing, machining, engraving, assembly, testing and more.

The plug-in's (lack of) software-only features

Usually at this point in my plug-in vs. hardware comparisons, I point out additional software-only features and extras. In this case the list is one feature long: a stereo linking switch that lets you adjust both channels from one set of controls.

An unlearning curve

I have been using the plug-in for much longer than the hardware, and while I love its sound and would gently tweak the occasional preset, functionally I never warmed up to it when starting from scratch. I honestly had a hard time wrapping my head around the controls.

UAD-2 Satellite Thunderbolt

While PCI and PCIe card versions of the original UAD-1 and current UAD-2 have enjoyed stability vs. technological change, non-PCI solutions have had to deal with rapidly changing trends in connectivity. In just seven years, Universal Audio went from the Xpander Xpress and UAD-2 SOLO Laptop card, which required ExpressCard slots (remember those?), to the FireWire-based UAD-2 Satellites. Now FireWire's gone from Macs (although the UAD-2 Satellite FireWire is still sold), and the UAD-2 Satellite Thunderbolt embraces the new Thunderbolt standard.

This new version, like previous Satellites, puts UAD-2 processing power inside a purpose-built enclosure. The new box does away with the flat silver design of the FireWire Satellite in favor of a 2 1/4 lb. black metal and plastic enclosure measuring 7" x 6 3/4" x 1 3/4". It is powered from a "line lump" power supply, and has a pair of parallel Thunderbolt 2 ports around the back. Current requirements are a Mac with available Thunderbolt or Thunderbolt 2 port, OS X 10.8.5 Mountain Lion, or 10.9 Mavericks (no 10.10 Yosemite support yet). I am unaware if work is being done on PC Thunderbolt drivers.

There are currently two Satellite versions, a QUAD ($999 and up depending on bundled plug-ins) and an OCTO ($1499 and up), each pertaining to the number of Analog Devices SHARC DSP chips on board. You can run a Satellite Thunderbolt alongside other UAD-2 devices, such as UAD-2 PCIe cards (including those in third-party expansion chassis), FireWire Satellites, and the Apollo interface/UAD engine lineup. All UAD Powered Plug-ins, of which there are well over 100, are available for VST, Audio Units, RTAS, or AAX 64, and should run in most current DAWs.

Both the QUAD and OCTO Satellite Thunderbolt versions are available in several bundles with different collections of plug-ins and vouchers. At the basic "Core" level, they include the Analog Classics Plus bundle, which gives you legacy versions of many UA favorites like the Fairchild 610, 1176LN, LA-2A, Pultec Pro, the new UA-610 preamp model, and more.

UA is no stranger to the world of Thunderbolt, thanks to the Apollo line (most recently reviewed in August 2014 with the Apollo Twin). Hookup, installation, authorization, and even firmware updates are all quick and painless.

The front panel of the unit glows with a nice white backlight when on; if there is a problem, the Host light will turn red. One advantage over putting a UAD-2 PCIe card in a third-party chassis is that the Satellite Thunderbolt needs no cooling fan. Next to my SSD-equipped Mac, it's dead silent.

There's really not much else to say. I switched to a new 27" iMac recently, which required me to give up my PCIe OCTO card. All of my tests of the new Satellite Thunderbolt OCTO have been flawless and painless; this is a compact, blazing fast, and studio-quiet solution for your DSP needs.

This was because I kept trying to make the Massive Passive behave like a normal 4-band parametric—boost a frequency, do a sweep, find the offending tones, flip the switch, boost or cut accordingly. I also kept treating bandwidth as a normal Q. Working this way, 75% of the time I found the plug-in frustrating, and would move on to easier parametric EQ choices like the Millennia NSEQ-2 (reviewed November 2013) or UA's own Precision EQ. I should note that I must not be alone, as the Massive Passive's manual recommends taking some time to read it cover to cover, so others don't fall into this same trap!

Sadly, it was not until I read the manual and had the hardware under my fingers that I really "got" the unit. Once I got the feel of how the Massive Passive should be used, and what it does and does not do best, then everything fell into place. Once I was comfortable with the paradigm, I used it on everything from vocals and instruments to the drum bus and full mixes.

Massive highs, mojo lows

On high frequencies the Massive Passive is equal parts clean, smooth, and three-dimensional. You can push the Massivo pretty hard at 8 kHz and beyond; sources like cymbals, guitar strings, and even dreaded tambourines get subtly more open but not biting or harsh. Unlike the current trend in active solid-state EQs, this is not a box that adds what most now call the "air" band.

The low end has a nearly intangible fullness that surrounds lows with an almost harmonic resonance rather than pushing or thumping them forward. You feel the low end through this EQ just as much as you hear it; again, the best word is "dimensional."

Bend, pull and overlap

For me the real heart and soul of the Massive Passive lies in the Bandwidth. It is the most important tone shaping parameter on the unit, and it's what gives the Massive Passive a sculptural quality rather than a surgical or utilitarian one.

More than just making a frequency poke out or notching one out like a Q control, here it is all about the width and the overlap between bands. Manley's description of the Bandwidth as damping and resonance will make perfect sense as soon as you turn and twist the controls; you'll hear tones get duller or more excited rather than just wide or narrow.

Having the choice of shelving bands really highlights the elastic taffylike nature of the box. Using these shelves together brings more of the vintage passive EQ flavor forward, and at times can be like boosting and cutting similar EQ bands on a Pultec.

Calling all sources

As to sources, this is one of those boxes that is at home on pretty much anything, though it will *live* on your mix bus if you let it. It's a standout on vocals, where you can get equal parts chesty low mid presence and nice open naturalness on the top that sits nicely in the mix. My most revelatory use of the Massivo, however, was on electric guitars. Here, the midrange shines in both boosting and cutting, especially the 2K7 (2700 Hz) range. In each instance the High and Low Pass filters are more than just add-ons; nicely smooth, they play a large role in the overall sound sculpting of the unit.

Hardware or software?

Now we come to the real crux of this review: How does the plug-in compare to the hardware? This is the part where I usually tell you that they are within 5-10% of each other and the hardware has a bit more vibe and air...

Not this time; I was thrown for a loop. Yes, they are sonically within 5 to 10% of each other, but in this case it's not the high-end air that differentiates the two, it's the low-end vibe. The team at UA nailed the high end and the air amazingly to my ear, but it's the dimensional depth in the low end where the hardware comes out ahead.

This difference in the lows is most noticeable when using the Massive Passive on a full mix or master. When using the two on one element in a mix, you can switch between the two and barely notice a difference at all. The only other thing to point out is that as with many plug-in emulations, I needed to drive the plug-in about a 1/4 dB to 1 dB louder to get the two to match in volume and and overall feel.

How much does the 5–10% plug-in vs. hardware difference matter, especially when the hardware is a $5600 handmade work of art while the plug-in (still a work of art) is only $299... and even less if you watch for one of UA's frequent and excellent sales? Here's a true story about that 5–10%.

I am in the middle of mixing a modern prog album with triple-tracked heavy guitars. For this review, I was testing out the hardware when I found out just how amazing it sounded on a stereo guitar bus with the heavy guitars panned left, right, center. When I played it for the band, they were blown away too!

I let them know that we could use the hardware for the mixes while I was reviewing the unit, but if the mixing took longer than my review period (it did), then I would have to send the hardware back and we would need to continue with the plug-in, or just skip the Massivo and use a different hardware EQ from my personal collection. The client decided the guitar sound was undeniably amazing and chose to continue with the hardware Massive Passive while we could, and then switch to the plug-in after that if remixes or tweaks needed to be made. The great surprise was that while the hardware simply soared on the guitars alone, when we did A/B comparisons in the mix we were hard pressed to tell the difference.

Perfectionism personified

On the plug-in power front, such an accurate emulation does take up a fair amount of power. In mono use it will eat up 37% of a UAD-2 SHARC chip, and 60% in stereo. Essentially stereo use is one to one depending on the size of your card: 1 on a SOLO, 2 on a DUO, 4 on a QUAD, and 8 on an OCTO.

But it's *so* worth the DSP expenditure. For Eveanna Manley to endorse a software plug-in of one of her company's devices is a big deal. In fact, for years at trade shows Manley Labs had a poster of a vacuum tube, stating, "This is a plug-in"! While many companies jumped into the modeling fray early, Eveanna held out until she heard a company get it right, and get it right they did.

And there's more Manley magic to come; just before I started this project, Universal Audio and Manley Labs released a software version of Manley's Vari-Mu Compressor... but we'll save that for a future review!

Price: Manley Massive Passive, $5600 (mastering version, $6300); UAD-2 Massive Passive Plug-in, $299

More from: Universal Audio, www.uaudio.com; Manley Labs, www.manley.com

FIRST STEPS

Part 2: In Your Room

By Paul J. Stamler

In last month's introduction I said that, before digging into the recorded clip that's the spine of this series, I would talk about elements of recording that get little attention and respect. I refer to monitors, and the rooms they sit in.

I'm talking about where you listen, and what you listen on, because everything passes through there. Every clip I post with these articles, you'll listen to in your room, on your monitors. More importantly, all the recordings you track and mix, and all the decisions you make about them, will be based on what you hear on *your* monitoring setup. If that setup has a boomy bass or a sizzly high end, you'll shape your decisions about mic choice and placement, EQ, and mixing on that sound—sometimes unconsciously. And when you listen to your recording somewhere else, or someone else does, it'll sound wrong.

Translation

In audio recording, *translation is the name of the game.* If you could guarantee that everyone who listens to your recordings would listen on headphones or earbuds, you'd be home free. But you can't guarantee that. You can't even come close; though many recordings will be listened to on the ubiquitous earbuds, many will be heard in other ways:

~ On big speakers (hopefully good) in the living room (hopefully quiet).
~ On small, cheesy speakers in a boom box.
~ On even smaller, cheesier speakers attached to a desktop computer.

All the recordings you track and mix, and all the decisions you make about them, will be based on what you hear on your monitors.

Cans (and can'ts)

I'll start by addressing a question I often hear from beginning recordists: "Can't I monitor my tracking and mixes on a good set of headphones? They're a lot cheaper than monitor speakers, and with headphones there won't be problems with room acoustics."

In my opinion, you can't. When I gave that answer to our Editor, he replied with a heartfelt "But I've worked with headphones as my monitors for years and years!" I did, too; I spent years and years doing remote recordings, usually in circumstances where headphones were the only possible monitors.

But the operative phrase, for both of us, is "years and years". It is possible to make good recordings and mixes while monitoring on headphones—but it's a skill that takes years to learn... and I have shelves full of recordings I made while I was learning that sound awful. Well, awful on anything but headphones, that is. [*And so does said Editor, and he uses them to check listening detail on headphones under review.—MM*]

The big problem is that headphones are *too good.* A decent monitor setup lets you hear into the mix, but headphones do this too well; you hear myriad tiny details that are easy to pick out under the headphones' aural magnifying glass. Unfortunately, most of those details won't be audible when you listen to the recording in any other way.

Headphones also create a very different stereo image from loudspeakers; they pump the left channel straight into your left ear, and the right channel straight into your right ear. In a room, the sounds from the two speakers combine, and the stereo image your ear hears from speakers sounds very different from what you'd hear on headphones. Net result: a recording made using headphones probably won't translate to another listening environment.

~ On 1" speakerlets built into a laptop.
~ On a car stereo, in a noisy environment.
~ (If you get radio airplay) On a clock radio—almost certainly in mono.
~ (If it's the right musical genre) On huge speakers in a club.
~ (If you can sell a clip from the song as a ringtone) On the tiny microspeaker in a cell phone.

	A	B	C	D
Fig1		Height	Width	Length
2		(feet)	(feet)	(feet)
3		8	10	12
4	f1	71	56	47
5	f2	141	113	94
6	f3	212	169	141
7	f4	282	226	188
8	f5	-	282	235
9	f6	-	-	282
10	f7	-	-	-
11	f8	-	-	-
12	f9	-	-	-
13	f10	-	-	-
14				

Natural Selection.
Mic Pres
512c • 505 direct input
Equalizers
550A 3 Band • 550b 4 Band
560 10 Band • 565 Filter
Compressors
527 • 525
Racks
5006b-HC 6 Slot lunchbox®
500-8Pack 8 Slot lunchbox
500VPR 10 Slot Rackmount
The Original 500 Series from API
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So you want a recording and mix that can translate to all of these potential listening setups, and probably several I haven't thought of. Each of them represents a potential destination for your music—and a potential market for it.

You might try to simulate many of these listening environments. For example, I check car stereo compatibility by burning a CD of a mix, bringing it out to my car, and taking a spin on I-44. My car stereo is nothing fancy (basic Chevy factory equipment), and with luck the sound I hear is representative of what the average car listener will hear.

But it's impossible to simulate every possibility, so a reasonable compromise is to monitor in a room designed to be as neutral as possible, on monitors that are also as neutral as possible.

The shape of things

Almost everyone knows that the way to make a good control room is to construct it from scratch while the building is being built, using non-parallel walls and a floor plan designed by an acoustical engineer. Nope, I can't afford that either.

In fact, almost nobody can; even the control room at our university's studio was built into a pre-existing space, and the walls are resolutely parallel to one another. Most of us will work in a space that's rectangular, so let's talk about shapes and sizes. And let's start at the bottom, with the worst possible shape—a cube.

What's so bad about a cubical room? To explain that, I need to take a short detour through the annoying territory of standing waves. You can look up standing waves online, and you'll get a lot of explanations (most of them involving jump-ropes).

I'll cut to the chase: in a space with parallel walls, acoustical standing waves develop at particular frequencies (usually low) that depend on the size of the room. At each of those frequencies, there will be places in the room where that frequency is exaggerated, and there will be places where it's barely there at all. So a room with a standing wave at a bass frequency of 100 Hz will, if you stand in the right place, have enough booming bass to rattle the fillings in your teeth. Move a couple of feet, and the bass will be gone; any music you pump into the space will sound like it has no bottom at all.

I've written a spreadsheet that calculates the frequencies of the most important standing waves in a room: the ones that develop between parallel surfaces, like the north and south walls, or the floor and ceiling. (For reference, these are called *axial modes*, but you don't really need to know that.) The spreadsheet is called StandingWaves.xls and can be downloaded at http://is.gd/StandingWavesXLS; it runs on every spreadsheet program I've tried, including Excel, Calc (part of the free OpenOffice suite), Numbers, and Quattro Pro. Be sure to also download the SpreadsheetReadMe text file that goes with it.

Let's look at one possible room, one with dimensions that are numerically related to one another: 8' high, 10' wide and 12' long. Type those dimensions into Line 3 of the spreadsheet; it will automatically calculate the frequencies of the standing waves characteristic of that room, up to 300 Hz. (See Figure 1, a screenshot.)

A good room will have an even distribution of frequencies, and each frequency will appear only once; if a frequency appears twice in the table, that means there will be standing waves at that frequency appearing between two pairs of surfaces in the room. As you see, 141 Hz appears twice, a double whammy that's bad news; parts of the room will have a boom at that frequency, while at other spots it will barely be heard.

There are some room-dimension ratios that engineers consider "golden"; they produce rooms with few standing-wave pileups.

When a frequency appears three times in the table, that's triply bad news; such a "triple pile-up" produces a triple-strong standing wave. In this room, such a triple whammy appears at 282 Hz; that would be a problem frequency if you tried to use this room for monitoring.

The problem is that the dimensions of the room (8' x 10' x 12') are related to one another by simple ratios (4 : 5 : 6), and rooms like that almost always create problems. The worst, as I mentioned at the start of the section, happens when the dimensions are all the same (8' x 8' x 8')—a cube.

Try typing 8, 8 and 8 into line 3 of the spreadsheet. Oops; every frequency in the table appears three times, a triple whammy across the board. A room like that will have very uneven response, and will be very hard to use for monitoring.

There are some room-dimension ratios that audio engineers consider "golden", since they produce rooms with a good spread of standing wave frequencies and no pileups. One example is 1 : 1.14 : 1.39; for a standard ceiling height of 8', that translates to a room 8' x 9.12' x 11.12'; type those numbers into line 3 of the spreadsheet, and you'll see that the standing wave frequencies are nicely distributed with no pile-ups anywhere. Now you can use the spreadsheet to find out whether the dimensions of *your* room are problematic.

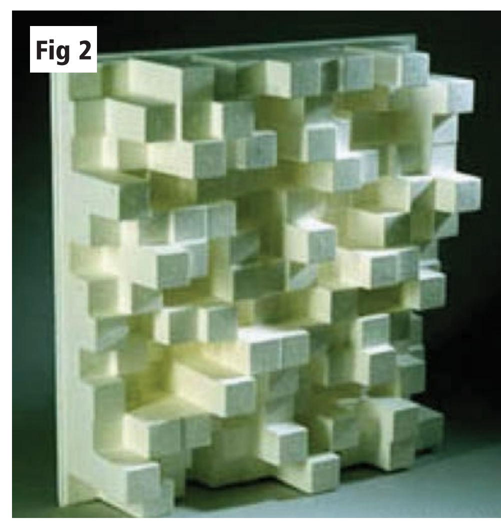


Fig 2

Recycled remedies

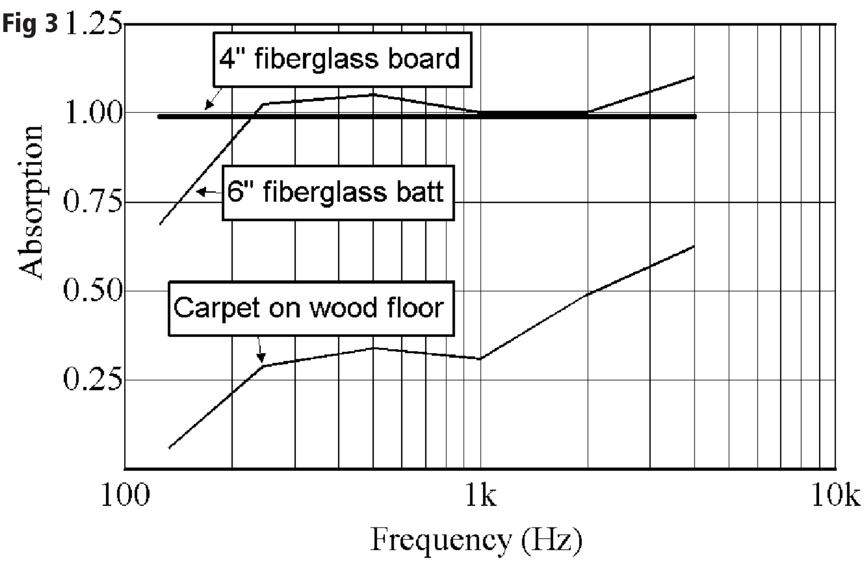
So what do you do if you find that your chosen room will have big standing wave problems? One solution is to move; choose another room in the house for listening/monitoring. For those of

us who live in the real world, sharing houses with spouses, partners or kids, that may not be practical; you may be stuck with a dubious room. In that case, if you own the house you're living in, you might consider changing the shape of the room.

If you read the article "From the Outside In, Part 1" (*Recording*, March 2014) you'll know that I encouraged readers to change the shape of a room by building a wall, shortening one dimension by 4"–8". It's not as intimidating as it sounds; as I said in that article, I did it in my dining room, and though it was a slow process it did get done. I'm about as low-grade a carpenter as there is, but if I built a wall, you probably could too.

Another way to change the effective size of a room is by building a floor-to-ceiling bookshelf on one wall and filling it with books. I talked about that in "From the Outside In" as well; a bookshelf not only changes the apparent dimensions of a room, but the books act as diffusors, which is good for the sound (see below). You might also change the room's height by installing a drop ceiling.

Fig 3



Skyline, photo courtesy of RPG Diffusors; the aforementioned full-wall bookshelf with different-sized books performs a similar function.) But I'll start with absorptive treatments, which get used more often in home studios. Before I talk about treatments that work, I'll talk about a couple that don't.

Urban legends: blankets and egg crates

Audio has its share of urban legends; a big one is that you can make a room sound good by hanging blankets or rugs on the walls, and covering the floor with carpet. Like most urban legends, there's a nugget of truth in this one. Rugs and blankets indeed soak up sound—but mostly at middle and high frequencies; they don't do much at low frequencies. Figure 3 compares the absorption of some common acoustical treatments; you'll notice that carpet absorbs the least, and it gives out at low frequencies. A blanket on the wall will be similar to carpet on the floor; and so, it turns out, will curtains.

So what happens if you treat a room with rugs and blankets? You get a room that's boomy, with lots of resonant sound in the bass frequencies, but midrange and treble that's dead, dead, dead. Granted, this is often an improvement over the boxy sound of many untreated rooms (a cynical old rule says "If you can't make it good, make it dead"), but it's far from the neutral environment you need for making mixes that translate well to other rooms. Soaking up sound at low frequencies turns out to be rather difficult. Everest describes possible solutions; see his discussions of "Mankovsky boxes", for example. But first I want to address the other important tall tale in studio/control room design: the egg-carton legend.

Somehow, back in the 1960s, the story arose that the cardboard trays used to pack eggs for bulk shipment make great acoustic treatments. They don't; this one is pure legend.

One way to change the size of a room without building a wall is to add a floor-to-ceiling bookshelf filled with assorted-sized books.

Or you can apply room treatments to soak up some of the problem frequencies. I've mentioned that F. Alton Everest has written multiple books on small-studio design and construction; let me again urge you to go get his books from your local public library, or to fetch them via inter-library loan. You might even buy them; check out www.abebooks.com, a consortium of used-book sellers. As I've mentioned, don't start with the *Master Handbook of Acoustics;* that's postgraduate stuff. Begin with one of Everest's more beginner-friendly volumes.

Soaking up troublesome standing waves requires narrowly frequency-selective absorbers; you can find DIY plans for building those in Everest's books, or online at www.realtraps.com, a website run by *Recording* contributor Ethan Winer for his firm RealTraps. Bruce Black describes DIY absorbers in our December 2013 and March 2014 issues.

That brings us to room treatments; they're potentially a long and complex topic, but we'll merely skim the surface, if you'll pardon the expression.

Paraphrasing John Windt, one of the engineers who built Motown's studios (*Recording,* October 2014): when a waveform strikes a surface, it can be absorbed or reflected, or some combination of the two. In studios or control rooms, reflective surfaces are often designed to diffuse sound: to bounce it around in multiple directions. (See Figure 2 for an example of a commercially available diffusor, the RPG

Notwithstanding, I've seen too many listening rooms and studios lined with egg-crate separators for comfort. After all, they're cheap (if you buy a lot of eggs), and they look kind of funkily cool.

I suspect that the looks are the basis for the legend; egg separators *look* a lot like real diffusors. Unfortunately, they don't work; you could probably use acoustical foam diffusors for shipping eggs (it'd be expensive), but the reverse isn't true. Forget blankets and rugs, forget egg crates; let's talk about real room treatments!

Soak it up

One traditional material used for making sound absorbers is fiberglass, either formed into solid panels or purchased loose at the hardware store, where it's sold as insulation. See Figure 3 again for a comparison of 4" fiberglass board (the gold standard of acoustic treatment) and 6" R-19 insulation from the hardware store. Acoustical absorption also comes from slabs of plastic foam, but this isn't just any foam; it's designed specifically for acoustic treatment, and the degree of sound absorption at different frequencies is specified by the maker.

That phrase "at different frequencies" is key, both for fiberglass and foam. The unhappy fact is that thinner slabs are less absorptive at low frequencies; in that respect they're like blankets. That means if you're using the kind of loose fiberglass that comes in rolls, you want R-19 (6"), not R-11 (4"). (If you use this or any other form of fiberglass, by the way, you should wear a mask and long rubber gloves. That stuff is nasty in your lungs, or on your skin.)

There are plans for homemade absorbers in Everest's books and on the RealTraps website; you should also check out my "Gadget" design, which is now posted on the *Recording* website (http://is.gd/DIYAcousticsGadget), or Matt Seiler's "Make Your Own Diffusers/Absorbers", again on the *Recording* website (http://is.gd/DIYDiffusers).

Or you can make a rock-bottom absorber, which I call a "Gadgette": Lay a 36" x 74" sheet of cloth on the floor (the cloth used for the underside of upholstered couches works well), put a 24" x 60" piece of R-19 insulation on it (using mask and gloves!), then top off the sandwich with a 24" x 60" piece of 1/8" waferboard. Fold up the edges of the cloth and staple them to the waferboard with a staple gun, then staple a length of picture wire to the back and hang the whole shebang on the wall. It ain't beautiful... but it works.

Soaking up the bottom

I said earlier that it's harder to soak up low frequencies than mids and highs. So what works down in those regions? This is a long discussion, one you can find in Everest's books and on the realtraps.com website. Briefly:

~ *Wallboard.* Bass signals pass through wallboard and out of the room rather well, as you know if you've lived in the wrong sort of apartment house.

~ *Thicker absorbers.* I talked about those already; to soak up lower frequencies 6" thicknesses are minimum, and 8" will be better.

~ *Bass traps,* which can be wideband, for general room treatment, or narrowband, to alleviate standing wave problems. There are some nice DIY plans on the realtraps.com site, and Bruce Black describes a DIY Helmholtz resonator in March 2014. I've had good luck building "Mankovsky boxes" using large plastic tubs (sold for mixing mortar or concrete), with a layer of fiberglass inside and topped with perforated pegboard; see Everest's books for details about designing these to soak up particular frequency ranges.

Winding up

You'll notice that my descriptions of room treatments have been pretty general. That's deliberate, since I don't know what your room looks (or sounds) like. You'll need to decide what treatments to use, and how many; for making those decisions, I once again recommend Everest's books and the RealTraps.com website. The folks who make Auralex also offer a remarkable deal, through the dealer Sweetwater: they'll plan out a treatment for your room for free (http://www.sweetwater.com/shop/studio/acoustic-treatment/room_analysis.php). Of course, they'll specify Auralex products, but that's only fair.

I'll be back next month with some talk about monitor speakers, plus a quick-and-dirty room setup that'll get you started on the path toward a neutral monitoring space. Stay tuned!

Paul J. Stamler (stamler@recordingmag.com) is a recording musician, engineer, radio show host, educator, and collector of vintage recordings, living in St. Louis.

BY MIKE METLAY

Resident Audio T4 Audio Interface

The speed of Thunderbolt in an affordable, compact interface

Announced by Intel and Apple in 2009 as "Light Peak" and first introduced on new Macs in 2011, the Thunderbolt interface protocol offers a whopping 10 Gb/second data transfer in both directions—essentially PCIe performance!—over copper wire (up to 3 meters) or fiber optic cable (up to 100 meters!), in a daisy-chain of up to six devices. Copper-cable Thunderbolt can also power devices requiring up to 10 Watts.

Unfortunately, Thunderbolt has been slow to gain acceptance in the music world. Its cables are expensive and hard to make, it's on all new Macs but few PCs, and Thunderbolt devices cost considerably more than their FireWire and USB 2.0 counterparts. But that price delivers amazing performance, and users have been eagerly watching as the first Thunderbolt audio devices make their debut.

To the small but growing list of such products can be added the offerings of new firm Resident Audio, whose T4 is the first in a line of portable and relatively affordable Thunderbolt audio devices.

Outside and in

The T4 is a 10.6 x 4.4 x 1.8 inch tabletop box, its front panel slightly up-angled for easy access, elegantly encased in aluminum and acrylic. It offers four audio inputs and four audio outputs plus MIDI I/O, all over Thunderbolt, for Windows 8+ and Mac OS 10.9+.

The front panel offers four Neutrik Combo XLR/TRS jacks for the inputs, each with a gain knob ringed by a multicolor green/yellow/red LED to indicate signal strength or clipping. Each pair of inputs (1+2, 3+4) can be switched to Instrument or Line input, and there's globally switchable 48V phantom power. An Input Mix knob with LED level-meter ring accompanies a large Monitor volume control knob that's ringed by a blue power LED.

The rear panel has four 1/4" TRS audio outputs, a 1/4" TRS stereo headphone out, MIDI, and Thunderbolt. There is no provision for daisy-chaining another Thunderbolt device or for an external AC power supply.

The specs are impressive: 92 dB or more of dynamic range, equivalent input noise of –124 dBu (Gain set to +60 dBu @150 ohm termination), and total harmonic distortion + noise (THD+N) under 0.01% at 1 kHz (–1 dBFS), regardless of whether the mic, line, or instrument inputs are selected. The preamps provide 45 dB of gain: +15 to +60 dBu for mic signals, 0 to +45 dBu for line-level or instrument signals.

Making it work

The T4 can be configured in two modes: Stereo Mix or Multichannel. In Stereo Mix mode, the Monitor knob controls the volume of Outputs 1 and 2 and of the Phones output simultaneously. In Multichannel mode, four channels of audio can be routed from your DAW to the T4, but in this mode the Monitor knob

only controls headphone level. Output levels are controlled by the T4 Digital Panel software that comes with the drivers. The clean, simple T4 Digital Panel software has meters for incoming and outgoing signals, level and stereo gang (link) controls for the outputs, and sample rate selection up to 96 kHz.

You access Multichannel mode by plugging a cable into Output 4; for safety, this mutes all outputs until their levels are reset in software. Output 3 doubles as a second headphone output if desired; plugging in here also mutes Outputs 1 and 2.

Note that unlike many such controls, the T4's Input Mix knob only controls the direct monitored signal from the inputs, leaving the signal coming from the computer at the same level at all times. It must be controlled from T4 Digital Panel or your DAW.

The T4 employs Smart Monitoring, in which the Inst/Line switch also controls how input signals are panned in speakers/headphones. Since line-level inputs are usually in stereo, setting the switch to Line pans a pair of inputs hard left and right in your ears. Since guitars and basses are usually mono (sorry, Rickenbacker owners!), the Inst setting places both inputs panned to center. This switch also controls panning if mics are plugged in.

This is all pretty straightforward, aside from getting used to what happens when you plug into Outputs 3 and 4—having your speakers go mute when you plug in a second set of headphones is a bit disconcerting at first!

In use

The T4 has a pristine sound, uncolored rather than "vibey". The preamps are quite clean except at extreme gain settings; weaker dynamic and ribbon mics might benefit from an external preamp feeding a Line input instead. The Instrument inputs have 1 Megohm impedance and don't harm guitar or bass tone. The headphone amp is wonderfully clear and gets good and loud. There's really nothing to complain about in terms of sound quality (impeccable) or latency (blazing fast).

The T4 ships with a half-meter Thunderbolt cable—quite short, but better than no cable at all, considering their expense. My only minor gripes are that I'd prefer dual Monitor knobs for separate control of output vs. headphone levels, and that without a dedicated power supply, the T4 can't use fiber optic Thunderbolt cables for remote setup far away from a noisy computer.

All in all, a very impressive debut. Resident Audio has announced other Thunderbolt boxes in larger and smaller configurations; this is one new company to watch!

Price: $499.99

More from: Resident Audio, www.residentaudio.com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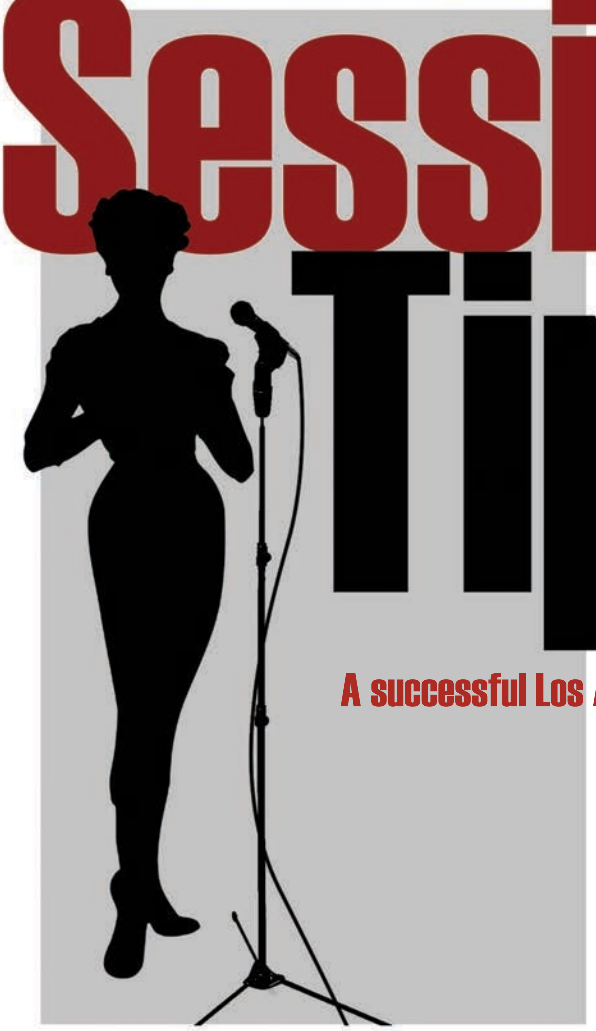
Vocal Session Tips

A successful Los Angeles producer/engineer opens his bag of tricks



By Greg Ogan

Working as a mainstream music producer in LA, I've been fortunate enough to produce and engineer a plethora of vocal sessions with top-of-the-line singers and artists. I'm often asked about my vocal production techniques; I explain that it's different for every situation I encounter. What I'd like to do here is focus on what I would do during a final vocal tracking day with an artist with the intention of releasing the song publicly. Hopefully I may be able to provide some ideas for your own sessions.

Setup and shootout

First off, I like to organize the song and tracks before the singer arrives. I like to turn everything down so that my vocal tracks start out being a little louder than the music. I import my vocal tracking template (as described in depth in my vocal template article on page 54), and I always check the mic and music levels in the headphone mix. I then make sure I have the lyrics on hand. It's important to know where we are in the song and to connect to it. Lyrics help with punching in on words, as well as having a mental picture of the song's meaning.

In an ideal situation, I like to do a microphone shootout when the singer arrives. That's when I line up a few mics going through the same style pre, ideally one that's clear and transparent, and run a section of the song with the singer trying each mic. When I'm comparing the mics to each other, I look for the one that brings out the best qualities of the voice. It may be a richness, it may be a bright high end, or a fullness—I go with whichever one makes me feel the best. I also listen for each mic's bad qualities and stay away from them. For example, I may need to go with the mic with the least amount of sibilance.

I'll pick certain gear depending on the session and genre. On a rock vocal session, I'll want to go with vibey gear that has color, possibly a Neumann U67 or Shure SM7, a Neve preamp, and a Universal Audio 1176 or Empirical Labs Distressor. For a more modern R&B pop sound, I may lean towards a Sony C800 and the Millennia STT-1. That combo is really nice, bright and clear. For rap/urban sessions I've used different variations of clean and colored gear.

Tracking with effects

When recording, I like to EQ and compress to "tape," although I don't process the vocal too hard unless it's a creative move. I know there are many schools of thought on this, but in this digital age of plug-in awesomeness, I like to get some magic out of the real hardware before I put on plug-ins. For EQ I like to simply cut lows and boost highs, cutting around 100 Hz and boosting just a little around 14–16 kHz. As for compression, I like to gain-reduce by around 3–6 dB, depending on the compressor, with moderately fast attack and release.

I really like the singer to hear their voice sounding as beautiful as possible; I'll feed them a monitor vocal with compression, EQ, reverb, and delay—see my template article for specific settings. I try to create an atmosphere and vibe for the singer so they can really get into the music, which supports a great vocal performance. I always ask the singer to holler at me whenever they want a change; with the template it's easy to accommodate their needs.

Vibe

When it comes to vocal production styles, in terms of vocal parts and organization, I have a few thought processes. There's what I call the rock or "organic" vibe, the radio pop vibe, and the sleek and shiny R&B vibe. Every situation, song, and singer may call for different techniques, but this is the best way for me to describe how I may organize tracks and parts.

The "organic" vibe is when I want the listener to really picture a vocal performance from top to bottom. This is where the lead vocal is the star of the show, and the song moves and grows around that vocal. If there are background vocal parts, I always like to picture a section of background vocalists singing behind the lead vocalist.

As for mixing those background vocal parts, one rule I always live by is *symmetry*. If I'm recording background harmony parts, they're almost always in pairs, because I like to pan each note symmetrically "around" the lead vocal. If I want to get doubles of the lead, I'll always get three total. There's the one lead who's alive and up front, and then two backrounds that are blended lower and panned 50% (or so) to the left and the right. For harmony parts, I like to pan them according to note: the higher notes are wider and the lower notes are narrower. I like the separation and soundscape this creates.

I have 4 or more tracks per note of a part then I pan in varying degrees. For the lead note (let's say I have 5 tracks) I'll pan the main track down the middle, and the four other tracks 100% left-100% right, and 50% left-50% right, an even spread of vocals.

For harmonies I'll get 2 tracks (doubles) of each note, panning them left and right by pitch. I'll try not to pan any note to the same degree. Because most pop vocals are really dense, each section is flown throughout the song. That means that if there is any section with repeating lyrics and melody, the same performance is used each time it occurs. To have the song build, I may introduce harmonies or accompanying parts later in the song, or add certain ad libs to create the illusion of an actual performance from top to bottom.

Greg Ogan in session with Sean Kingston and Bruno Mars

For a more mainstream radio vibe, the approach is different. I see it as painting the song in different puzzle pieces that come together to form a whole picture, i.e., each section of the song is treated differently for what's appropriate to that section. For a verse, it may be one vocal that's intimate and up close, the prechorus is starting to be stacked and harmonized, and the chorus becomes lush stacks of vocal textures and background oohs and ahhs to create a big landscape that pops out.

In these cases, the stacks may generally be more dense. The "lead" part may be 1, 3 or 5 parts stacked. The harmonies could be 2 or 4 tracks per note blended in. There could be those oohs and ahhs or background lines that are also stacked heavily. In these cases, I stick to my law of symmetry. If

For R&B vocals, I combine the two philosophies of organic and pop treatments. During most of my R&B productions, there is a lead performance from beginning to end, where each section is an expansion upon the last, adding runs and licks and vibe. However, at the same time there is *heavy* vocal production in terms of stacks, harmonies, little ear candy riffs, runs, counter melodies, etc. R&B vocals are very sleek and shiny, and to achieve this I get tons of layers of vocals. These are the most intense sessions in terms of organization, mixing on the fly, and track management. I've gotten up to 8 tracks of each vocal part! I always stick to my rules of symmetry, by panning in various degrees, and keeping everything even.

Cleanup

Comping, tuning and editing vocals is a very in-depth process with lots of options, worthy of its own article. To sum up my philosophies: I don't like the result of (or enjoy) over-comping, over-editing, and over-tuning. In this day and age of technology and potential perfection, I like to keep vocals feeling human. I don't remove breaths, and I comp in bigger phrases, if possible. I enjoy comping vocal takes with the singer in the room… if I trust them and think they know what they like. If the vocals need a lot of repair, I will do it on my own time without them.

If the vocals either need a ton of repair, or need very little repair to keep an honest performance and organic feel, I will use Celemony Melodyne to tune the vocals. If they need that radio shine and tight tuning, I will use Antares Auto-Tune, tuning the whole performance and changing notes by hand if necessary. To keep things tight, I will use Synchro Arts Vocalign on background parts.

The mix

As far as mixing is concerned—well, at least rough mixing during a tracking date—every vocal part and section is assigned its own bus for independent level control and processing. See my vocal template article for more details. Whenever I have a lead vocal going down the middle, I'll always add a compressor to it before it hits the bus that already has another compressor inserted. I usually serially compress most lead vocals like this. There are other vocals going to that same bus as well, and I really like to glue it all together that way. I'll send to reverbs and delays as appropriate for the song and the part of the song. When a singer exits the booth, they want to hear the song down as quickly as possible, so most of these rough mix moves are happening during the recording process.

This sums up my mindset during vocal tracking days. The one factor I didn't mention is external input from artists, significant others, producers, managers, labels, etc. That's always a wild card! I can't stress enough the fact that every situation is different, but hopefully some of these principles can guide you to some cool creative results.

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REVIEW

BY PAUL VNUK JR.

DPA Microphones d:screet Necklace Microphone

A mic that's great-sounding, simple to use... and yes, discreet

DPA, short for Danish Pro Audio, is a Denmark-based microphone company known for (among other great products) its ultra-clear and ultra-compact microphones. This month, for our Vocal issue, we are looking at the company's new d:screet Necklace microphone, perfect for voice capture in video, public speaking, and theatrical situations.

Typically, a "necklace microphone" refers to a standard lavalier mic literally attached to a piece of string worn around the neck. This was often an option for when a speaker or interviewee was wearing a shirt or dress that could not accommodate a typical clip-on lav mic. By contrast, the new d:screet Necklace mic is a purpose-built high-quality solution that greatly improves on that old workaround.

The microphone

As the name suggests, this mic is built around one of DPA's d:screet series miniature omnidirectional microphone capsules. Typically the d:screet series mics are best described as standard lav-style microphones, i.e. a tiny mic capsule on the end of a long thin cable. DPA offers a plethora of optional kit packages, including solutions for both indoor and outdoor lavalier use—even with heavy-duty windscreen assemblies, instrument mounting systems, and more.

For this new version, the mic is built into a rubber tube-style necklace with a magnetic, locking clasp on the back, along with a pivoting metal swivel where the cable attaches. The 3.6' cable terminates into one of DPA's familiar Microdot plugs, which can then attach to the wireless-pack connector of your choice. There are many different choices that will work with this mic for connecting to almost every brand of wireless belt pack—Shure,

Audio-Technica, beyerdynamic, Electro-Voice, Trantec, and more. You can even get one for a standard XLR mic cable.

This mic is available in black, white and brown in lengths of 18.3" or 20.9". If, like me, you have a huge neck, you'll be much more comfortable with the longer necklace. I was sent the 18.3" model and it was a tad snug.

Specs

The mic itself is an omnidirectional prepolarized pressure gradient condenser element with vertical diaphragm. It has a 20 Hz to 20 kHz ruler-flat frequency response, but comes prefitted with one of the company's high boost grid covers. This adds a 10 dB soft boost at 12 kHz, useful for when a lavalier mic like this one will be chest mounted. The mic is built to handle 144 dBA SPL peaks prior to clipping and further specs are as follows (from the DPA website):

~ Sensitivity, nominal ±3 dB at 1 kHz: 6 mV/Pa; –44 dB re. 1 V/Pa
~ Equivalent noise level, A-weighted: Typically 26 dBA re. 20 μPa (max. 28 dBA)
~ Equivalent noise level, ITU-R BS.468-4: Typically 38 dB (max. 40 dB)
~ S/N ratio (A-weighted): 68 dBA
~ Total Harmonic Distortion (THD): < 1 % THD up to 123 dB SPL peak; < 1 % THD up to 120 dB SPL RMS sine
~ Dynamic range: Typically 97 dB
~ Output impedance: 30–40 Ω

In the field

Mics like this will probably not have too many practical uses for studio-based audio engineers. However, they can be a great option if, like me, your work takes you out of the studio and into the fields of speech for video, producing and recording sermons, dramatic readings, and speeches, as well as general theatrical work. It is a great alternative to traditional lav mics as well as over-the-ear mics like DPA's d:fine headset models (reviewed May 2013).

Trust me, there will come a time when based on factors like a person's attire, or the liveness of the room, that a lav mic is not an option. Also, I know quite a few speakers and actors who feel that over the ear/headset mics are uncomfortable, conspicuous, and even showy.

Anyone can use it

This mic was also designed with the technophobe and novice in mind. Many people get selfconscious and second-guess both lav and headset placement, but almost all of us know how to wear a necklace! This design doesn't offer much room for placement mistakes.

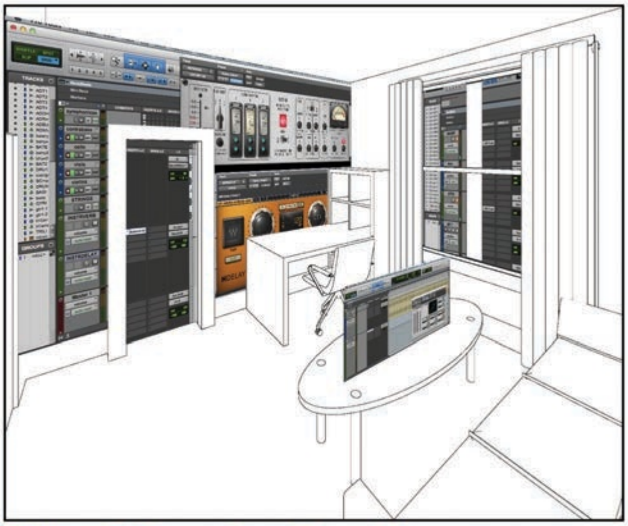
I found this mic sounded wonderfully clean and clear, just like every other DPA mic I have used. Currently I have four various d:fine headsets in my public speaking/theater collection, and this sounds as good as all of them, plus it offers two advantages thanks to its placement. It is much less prone to plosives, and being by the throat, it captures a nice deep resonance.

On the downside, it is an open mic, and just like a lav, it will pick up noise if fabric or long hair crosses its path. I also found that some male speakers I wanted to use it on felt that the black model I was sent looked too much like a gothic choker... but on the other hand, it hides nicely in a shirt collar.

All in all, this is another great solution from the folks at DPA, complete with the sound these mics are famous for!

Prices: $649.95 (not including adapter)

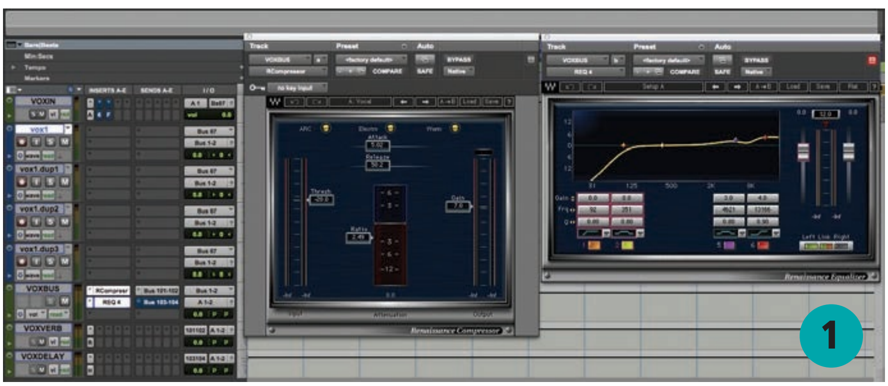
More from: DPA Microphones, www.dpamicrophones.com



A Vocal Recording Template

By Greg Ogan

A little DAW prep will make your vocal sessions run more smoothly



When I'm sitting in the fire during a vocal tracking session, my highest priority is to create a sound and environment that supports the singer's comfort level so that they can perform at their peak. Engineering speed and sonic quality are two key elements to help achieve a smooth flow for capturing vocal takes—and hopefully some magic! My vocal template gives me the means and ability to perform at MY best so that any singer can (hopefully) perform at theirs.

Disclaimer: there are many different philosophies and styles to recording vocals; for example, some engineers track completely dry so they can hear every nuance loud and clear. I prefer to record vocals wet with compression, EQ, reverb, delay etc. This template can accommodate any music genre, philosophy, style, personality or needs of the singer I'm dealing with, even if it means bypassing my normal methodology. This setup is the culmination of thousands of hours of vocal sessions and tweaking. It should also be noted that every vocal tracking session is different, and my experience, instincts, and tastes guide me to make certain choices in different situations. I will explain my setup in Pro Tools terms, but I feel the principles can be applied to all DAWs.

Tracks and buses

First I start with my vocal input track. In Pro Tools, I create an aux track and set the input so that my mic is feeding the aux track. I use this vocal aux input as a starting point in case I want to do any digital processing "to tape." I'm big on commitment, and a lot of times if I get a sound that I like, I print it. This could be anything from running a scratch demo vocal through Auto-Tune and extra compression, to getting a cool telephone/megaphone sound and just printing it.

I set the output of the aux track to a bus that will be feeding the inputs of every vocal audio track in the session. Note that 85% of the time I *do* use this vocal input track, but there are certain sessions where I bypass this track altogether and change all the vocal audio tracks' inputs to my mic feed. See Figure 1: VOXIN is my vocal input track.

Then I set up my system of audio tracks and buses (Figure 1). I create four audio tracks which will feed a vocal aux input. I set the outputs of the audio tracks to correspond with the input of the aux track (eg. vocal aux input is set to bus 1-2, so the vocal audio tracks' outputs are set to bus 1-2). The idea behind this is that a lot of the vocal processing will take place via the vocal aux track, and not the audio tracks themselves. I can EQ, compress, and send to reverbs and delays in bigger batches rather than putting plug-ins on each individual track. It saves time and resources, while adding overall glue to what can become many tracks of vocals. So let's set this up.

First I put on a vocal compressor. There are many vibey compressors that I love and use for mixing, but for tracking I want something that's easy, effective but transparent, and not processor intensive. For this, I use the Waves Renaissance Compressor. There's a great "vocal" preset, and I set the threshold to –20 and the output to +7. Then I put on my EQ; considering my criteria, I generally use the Waves Renaissance EQ. I cut the lows, boost highs, and I also boost the high mids. With these EQ and compressor settings, the vocalists will sound up front, present, breathy, and exciting. When boosting the high mids that vocal will cut through the music so they can hear themselves better. I then lower the output of the vocal audio tracks to around –5 dB. They're feeding a compressor, so I like the room to raise certain tracks to hit it harder.

Next I like to setup my vocal reverb and delay. I create two aux return tracks. For the reverb, I love the sound of the Universal Audio UAD-2 EMT 140 plate. I set my time for a little under 2 seconds with no predelay. This reverb is pretty vibey, and if I want something a little more simple and nice, I'll use the Waves Rverb on the default preset. It's just a nice hall and gets the job done. I also like Reverb One or the Lexicon reverbs. For my delay track, I always start with a simple $^1/_4$-note digital delay with a little feedback and some sort of modulation on that feedback. Line 6's Echo Farm was my go-to for years, but no longer—I now use SoundToys EchoBoy or Waves HDelay.

I create sends on the vocal aux buss that correspond to the inputs of the reverb and delay returns. I raise each level, but then I bypass the send to the delay, making it accessible with one click. Nine times out of ten, when a vocalist enters the booth and checks the mic, they are pleasantly surprised at how their vocal sounds. Most of the time they want to hear ambience and feel vibes. The vocal booth is enough of a sterile environment, so I believe in making their voice sound as fantastic as possible when recording. If someone doesn't like it, I can bypass anything with one click.

Makin' copies

Now I highlight my vocal tracks and my vocal aux input (but not the reverb and delay returns), and I duplicate that whole setup at least four times. I change the output of the subsequent audio tracks to correspond to its own bus, so now we have at least five sets of those vocal tracks. See Figure 2.

I have so many sets of tracks and buses simply for easier mixing. I got this concept from JR Rotem. He didn't like the time it took to automate vocals, and he wanted me to mix quickly, so I soon figured out that every vocal part that I *would* mix differently, should go to a different bus.

Here's an example. In a chorus, I may have a vocal part that's singing the main melody and lyric, which may be doubled, tripled, quadrupled or beyond. In addition, I may have vocal "oohs and ahhs" going on at the same time. I may want to EQ the oohs differently, and obviously they would want to be lower than the main melody, so I record the oohs on a separate bus system. This enables independent groups that I can process quickly, and raise or lower with one mouse click. I go into more specifics in my vocal production article on page 50. If I ever need more tracks or buses, everything is two clicks away with a simple "duplicate" command.

Roll your own

This is the basic setup that I take to every studio when I track vocals. As a song progresses, so must my tools, so I'm constantly swapping out plug-ins, adding different delays, reverbs, compressors etc., but for fresh tracking this is a great starting point.

When a singer enters the booth, we are creating a soundscape together. As the session progresses, so does the mix. If we are hearing a more finalized product, then we can more accurately judge what we need to add or subtract. Technically speaking I'll have all the tools to react to both of our changing needs. It helps me track faster to keep singers in a flow, and it minimizes downtime due to various knob fiddlings and dialing stuff in. The end result is that when a singer exits the booth and wants to hear the song back, it's ready to go, keeping up the vibe of the session.... and that's what's most important.

Korg Gadget

By Devon Brent

Korg calls Gadget a "Music Synthesizer Studio." With up to 17 synthesizers and drum machines plus a nice built-in sequencer, Gadget is quite the powerful studio app.

World tour

Korg decided to name its drum machines and synthesizers after cities around the world. Gadget has three drum machines: London, Amsterdam, and Tokyo. The twelve synthesizers are Marseille, Chicago, Wolfsburg, Berlin, Phoenix, Dublin, Miami, Chiang Mai, Helsinki, Kiev, Brussels, and Kingston. New to version 1.0.3 were two in-app purchases, a loop slicer called Abu Dhabi and a sampling rhythm box called Bilbao. Just as we went to press, Korg released a new virtual instrument app called Module, which includes five "Gadgetized" versions of its keyboards for use within Gadget.

Each drum machine and synthesizer has its own particular strengths. For example, Kingston is a polyphonic chip synthesizer that makes 8-bit like sound effects. Marseille is a PCM sample playback synth that runs the gamut of typical General MIDI synth sounds: pianos, organs, keyboards, strings, choirs, guitars, and more. While most of the drum machines and synthesizers offer their own one-trick synthesis pony features, they do pull off some fabulous tricks.

Sequence, mix, export and share

The sequencer is arguably Gadget's best feature. MIDI data is broken up into Scenes, each of which can have a length of 1 to 16 bars. You can also adjust the time signature for the Scene from 1/1 all the way up to 32/16 and can use any beat count value in between. Each instrument has its own separate track within the Scene. If you have two tracks, with Track One being one bar in length and Track Two is four bars in length, you can set the one bar track to play in loop mode or just do a 1-Shot playback for the whole Scene.

Editing within the Scene is done piano-roll style within a typical quantized grid. Thankfully, the grid is merely a placement tool, and not a hard and fast placement of every note. The grid is adjustable from 1/2 to 1/64 note divisions and can be set to do triplets. You can adjust the grid after entering your piano note roll details, and the notes already placed will stay in position. This lets you, for example, make a triplet in a bar at the beginning of the measure, but easily put in straight 16th notes after the triplets. If the thought of quantizing every note turns your stomach, fear not; you can switch quantize off and move, lengthen or shorten any note however you like with a tap and drag of the finger.

Once you complete your Scene, you can mute, clear, or copy your Scene to a new Scene, or copy a Track into another Track in your project. From there, you chain all of your Scenes in order to make a song.

Along the bottom of the interface is the mixer, which allows you to individually control pan, reverb send levels, volume, and set solo or mute for the whole track, and adjust the master reverb and limiter. Once you're happy with your track within a project, you can Freeze the track to free up CPU power on your iPad if you so choose.

If this still isn't enough control, version 1.0.3 of Gadget allows you to export your tracks to audio into Ableton Live! If you're happy with your song the way it is, you can export the whole mix or individual tracks directly to Dropbox, or via iTunes sharing on the Mac or PC. Gadget also sports the AudioCopy and Audiobus 2 feature sets, so you can use your tracks in your other favorite iPad apps as well.

In use

When I initially got Gadget, I wanted to see how easy it was to use on the iPad itself. Without reading the manual, I managed to create a respectable 8-Scene song, complete with bass line, chords, and a drum track, in about 30 minutes. I found editing and moving MIDI data to be easy and intuitive.

I was also very happy to discover just how darn good Gadget's instruments sounded. From the variety of drums within London, to the synth leads in Berlin, to the breadth of PCM samples within Marseille, it all sounded amazing for an app on the iPad. Each instrument has a good number of very usable presets that show off each synth's strengths quite well.

I also found out track count was heavily dependent on which iPad you have. While the minimum requirements are an iPad 2, expect to be able to pull off a limited 5 tracks without freeze and 8 with freeze. Those of you lucky enough to have the iPad Air can expect 20–25 tracks real time, with 30–35 tracks when frozen. My iPad 3 couldn't even pull off the 9-track demo song, Gadget World Tour, unless I froze a few of the heavier-duty tracks up front. That falls in line with what the iPad 3 is listed to be able to handle: 8 live or 12 frozen tracks.

I tried pulling Gadget into my MIDI studio. Gadget was able to recognize my M-Audio Uno MIDI interface (note for first-time users—you must have the interface up and running before starting Gadget) and I was able to play Gadget's instruments on my MIDI keyboard. However, once connected to MIDI, the Play and Record buttons disappear from Gadget. Sadly, the MIDI information my CME Bitstream 3X sends for Play and Record are not recognized by Gadget. Instructions for external MIDI configuration are completely lacking in the manual as well. I suspect the list of supported MIDI devices on Korg's website, which didn't include my CME, will work as advertised.

So what's really lacking in Gadget? The ability to use or import your own audio as just a separate audio track. Gadget's new devices in version 1.0.3, Abu Dhabi and Bilbao, allow you to import your own samples to play with.

These new instruments are in-app purchases and cost $9.99 apiece. While I suppose these new devices would suffice for the ability to add your own audio tracks, I'd prefer the more traditional approach of a plain-Jane import.

Conclusion

Not only does Korg's Gadget offer a superbly diverse range of instruments, and have a wonderfully intuitive sequencer, it sounds terrific too! With all of this goodness packed in for a mere $39.99, Korg offers on-the-go musicians an amazing bargain with amazing features. Editing and entering MIDI data, even without keyboard integration, was a pleasurable experience. Gadget is simply the best music app I've played with on the iPad, hands down. What are you waiting for? Go get it!

Price: $39.99

More from: Korg, www.korg.com

Devon Brent (devon@recordingmag.com) has moved his studio and music production work to the outskirts of Austin, TX, and couldn't be happier.

SPECS: REMOVING THE MYSTERY

Part 1: Levels

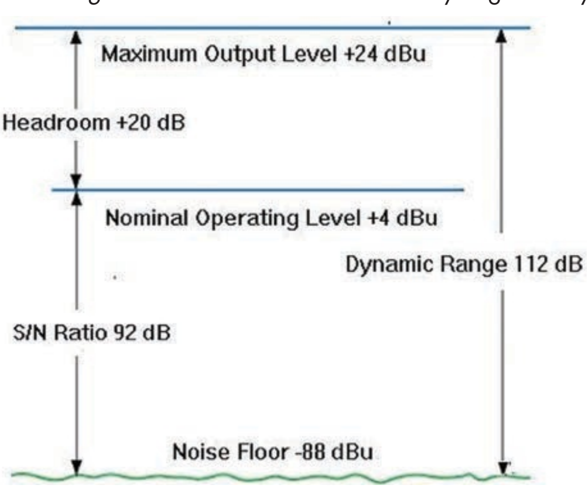
By Mike Rivers

"Lies, damn lies, and specifications." If you've been around these parts for a while, you've probably heard that paraphrase of a quote attributed to Mark Twain. Often the spec sheet is one of the first things we look at when considering a new piece of gear for our setup, but not all spec sheets are created equal. In this series, we'll look inside many of the specifications published for audio gear, explain their meaning (or meaninglessness!), and show how understanding certain specs can help you work better, as well as avoid or diagnose problems.

[*Editor's note: Many of these discussions will involve some math, and some basic understanding of audio terms. Don't let that scare you away! Understanding this stuff will help you work smarter and get better results from your gear. You can check out recordingmag.com's online glossary for help on many terms; when possible, we'll provide ways to make calculations without doing the math by hand (e.g. the online voltage-to-dB calculator mentioned below).—MM*]

Level with me

Since few devices we use in audio production stand alone, understanding level specifications is vital. When chaining two units together, it's important to know that Box 1's output needs to properly match Box 2's input. Juggling output and input levels for optimum performance is known as *gain staging*, and understanding a device's input and output level specifications will help you choose and manage devices in your signal chain.



Level specifications for audio equipment are almost always based on RMS (root-mean-square, a sort of "average") voltages. Notable exceptions are with power amplifiers where power (watts) is specified, and in acoustics where we deal with sound pressure levels (SPL). In transducers like microphones and loudspeakers, both sound pressure and voltage levels are significant; we'll cover those in another installment.

Decibels (dB) and volts

The *decibel* was originally used for power measurement (watts), however today it's been adopted (purists say "corrupted") for specifying voltage. The decibel isn't a physical parameter like voltage—it's the *ratio* of two values, in this case a measured voltage to a known reference voltage. In order to meaningfully express voltages as dB, we need to know the reference voltage used to calculate the ratio.

The reference could be anything, but fortunately we have a few standard reference voltages. The two most common for dB are 1 volt (logical) and 0.775 volts (huh?). 0.775 volts is the reference most commonly used in modern studio gear specs. It's no coincidence that this is the voltage that produces a reading of 0 on a standard VU meter. And yes, there's a specification for the VU meter, too. "Semi-pro" or "prosumer" gear from the early 1980s project studio days usually specifies levels in dB, referenced to 1 volt.

We differentiate between these reference standards by adding an extra letter, dBu (0.775 v reference) or dBV (1 v reference). A level stated only as "dB" is ambiguous. In this era of journalistic shortcuts, it's a good bet that the author really meant dBu, but you can't be sure.

To relate volts to dB, we have to use logarithms (logs), which let us turn very large or very small numbers into more easily managed numbers. For example, the logarithm of 1,000,000 (six zeroes) is 6, and the log of 1/10,000 (four zeroes) is –4. Logarithms are handy because multiplying two numbers is the same as adding their logs, and dividing two numbers means subtracting their logs.

This is why we use dB in level specifications rather than just plain volts. Yes, it's partly tradition, with dBu and dBV being a modernized version of dBm (see below), but it also makes the numbers easier to handle. Since decibels are basically logarithms, we can write a quiet noise floor of 0.00003 volts as –88 dBu, or the 20-volt output of a hot mic preamp as +28 dBu. When all your numbers are in dBu or dBV, calculations become simple addition and subtraction.

Volts and dB are related by the general formula: dB = 20 x log (voltage divided by reference). Commonly, dBu = 20 x log (V/0.775) and dBV = 20 x log (V). You can use the antilog function 10^x on your calculator to convert dBu or dBV to volts. Remember that dB will be a negative number when the voltage ratio is less than 1, for example 0.316 volts = –10 dBV.

If you would prefer to let your computer do the work for you, you can find a handy on-line dBu-dBV-voltage calculator: www.sengpielaudio.com/calculator-db-volt.htm

You may occasionally see a level specified in dBm. The "m" indicates that the reference is 1 milliwatt (power, not voltage). This is a carryover from telephony and broadcast, the source for much of our early studio gear. Back then, everything had an input and output impedance of 600 ohms. Matching output and input impedance maximized the *power* passed from one device to the next. Today's gear is designed with low output impedance and moderately high input impedance. That maximizes the *voltage* passed from one device to another.

dBm is correctly used when, for example, you're specifying the radio-frequency (RF) power output of a wireless mic transmitter, but it's nearly always wrong when used in specs for a mixer or preamp. dBA or dBC are frequency-weighted measurements most often associated with sound level or noise, a topic for another installment.

Level measurements

A product's specifications often originate in the Marketing Department. They want their product to look better than the competition. Even though the final published spec sheet is taken from laboratory testing of the product, tests will sometimes be conducted under conditions that don't represent actual usage, making the numbers look better than you might see in practice.

Let's take a look at *maximum output level* (MOL). A test signal, nearly always a very pure sine wave, is applied to the input of the device being tested. When it's cranked up to maximum, the RMS output is measured with a voltmeter. "Maximum output" is defined as the level where the *total harmonic distortion* (THD) of the output reaches a specific amount. Fair enough, but *what* amount? There's no standard; it could be 1%, 3%, or hard clipping. Sometimes they tell you, sometimes they don't. Furthermore, MOL is typically specified at 1 kHz, but distortion, hence MOL, can vary with frequency. Beware of the unspecified specification!

MOL measurements are also affected by electrical load on the device's output. Modern solid-state gear is happiest when driving an input impedance of 5 kilohms or greater, and that's how it's tested. Feed a modern mic preamp into a vintage compressor with a 600 ohm input impedance, though, and the preamp may distort at a level lower than the spec sheet's MOL.

A proper MOL specification reads something like this: "+24 dBu, balanced, 20 Hz to 20 kHz, 5 kΩ load, THD < 1%." If all it says is "24 dB" you don't have a clue.

Nominal operating level is commonly stated as one of the two de facto standards, +4 dBu or –10 dBV. (These two choices have an interesting history; see page 8.) A mic preamp with a nominal operating level of +4 dBu will need be turned down by about 12 dB when connected to a nominal –10 dBV computer interface to prevent the interface clipping on peaks. Conversely, if you connect the output of a "–10" mixer to a "+4" interface, you may not be able to reach full record level.

Good-to-know levels and their relatives

While level specs are useful on their own, they relate to two important characteristics, *headroom* and *signal-to-noise ratio* (S/N).

Headroom is the ratio in dB between the maximum and nominal output levels. A preamp with +4 dBu nominal operating level and MOL of +24 dBu has 20 dB of headroom. Another "+4" preamp with a maximum output level of +16 dBu offers only 12 dB of headroom. Learning how to use headroom to best advantage is the topic for another article.

Most devices also have a *maximum input level*. The record volume control on your grandfather's tape recorder allowed you to get a proper recording level from nearly any source, but your computer's audio interface may not have such a control. The maximum input level of a computer audio interface is usually the level that produces the maximum (0 dBFS) recording level. If its maximum input level is +16 dBu, it will clip well before your +24 dBu MOL preamp does. Take care!

There are minimum input levels, too, but they're more usefully specified as *sensitivity,* which I'll explain in detail in a future article.

Signal-to-noise ratio (S/N) is the ratio in dB between the nominal operating level and the level of noise in the device's output with no input signal. A device with a +4 dBu nominal operating level and a noise level of –88 dBu has a S/N of 92 dB. There's room for some specsmanship here since noise level can be measured in different ways – frequency-weighted or broadband, and with the input shorted or terminated... but never with an open (unplugged) input since that's always the worst case. *Dynamic Range*, often confused with S/N, is the ratio of MOL to the noise floor. See Figure 1 for a diagram of how these levels relate to one another in a typical setup.

Good to know

Level specs can help you choose compatible gear, avoid weak links in your system, and diagnose "too hot" and "not hot enough" problems. When comparing specs for similar devices, differences may appear relatively insignificant, though a spec sheet that's clearly and completely written is a good indication that the unit is competently designed and the manufacturer has nothing to hide. Finally, since level measurements are the basis for other specifications, it's important to understand their relationships, as we'll see in forthcoming articles.

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REVIEW

BY PAUL VNUK JR.

Cathedral Pipes

Notre Dame Tube Condenser Microphone

Awesome sound to match unbelievable looks

One of my favorite new audio-gear trends in the past few years has been the rise of custom mic builders. Second only to custom car geeks, these folks take the classic mic designs—usually those that start with C, U, or E—and then tinker with their favorite recipe of capacitors, transformer windings, grille covers and more, until they come up with a microphone that offers a classic heritage, but with each designer's own special signature flair.

I have been test-driving a mic that's one of the most pimped-out reimaginings of the venerable Neumann U47 that I have ever seen: the Notre Dame from the folks at Cathedral Pipes.

This ain't your Grandpa's vintage mic!

The Notre Dame, like all Cathedral Pipes mics, is handbuilt in the company's California facility by company founder Charles Dickinson. The line currently contains 4 models: The Saint Jean Baptiste FET condenser, the Seville ribbon, The Regensburg Dom (also U47-inspired), and the king of the castle, The Notre Dame.

These are some of the blingiest mics on the planet, featuring shiny chrome bodies with color fades, coats of arms, and more. The visual icing on the cake is that each microphone's grille and capsule assembly is internally lit with glowing, colored LEDs. There have been a few microphones in the past to have lighted capsules, like Korby's Red and Blue models and at least one Heil mic, but I am not sure if they used similar technology.

Overall I would be curious to hear Chuck's tales of where his chrome covered coat of arms and bright internally-lit capsules came from... there are no other mics on the market that look like these!

The Notre Dame in all its glory

The Notre Dame is inspired by the U47, but due to the demise of the old steel tubes used in the original Neumann, the Notre Dame uses a glass NOS Valvo/Phillips GmbH PF86 tube, giving the mic just the slightest hint of U67 character as well.

Staring with the outside, the mic body is indeed a replica of a U47 down to the dimensions and look of the head basket, although by design the head basket is a tad more open than the original. The mic is finished in a shiny chrome with a white enamel fade and a large Cathedral Pipes coat of arms badge. Its head basket glows a warm white!

Internally this mic illustrates the loving design of a custom builder, with hand-chosen components at every turn. The capsule is Charles' own take on the Neumann M7 capsule, handmade in his Orange County shop. All components are personally chosen by Charles, including the Cinemag CM-2461 NiCo output transformer, Wima MK4 and Solen capacitors, and a monstrous paper and oil coupling capacitor (almost the size of the PF86 glass tube!) made by Tobias Jensen.

Cathedral Pipes offers no specs on the mic at all, but we don't listen to spec sheets... we'll make do with our ears.

And ALL the trimmings

Before we jump into the Notre Dame in use, we must talk about the kit that it comes with. I'll cut to the chase and let you know that I have never seen a microphone package done this well before, and I'm pretty sure you haven't either.

It starts with a deluxe oversized roadie-style briefcase custom made by GOMC (short for Get Off My Case). Inside is a thick, deep die-cut foam lining topped in a harder red foam—everything in the package is the same matching red and white. The kit includes a Rycote shock mount, again custom made for Cathedral Pipes in white and red. There is a matching red pouch which houses a red power cable for the power supply and a custom-made red multipin mic cable with a threaded lock that screws to the bottom of the mic like the cables of old. The cable is made at Cathedral Pipes HQ using special OCC (Ohno Continuous Cast) copper wiring.

And then there's the power supply. One of the first things that drew Editor Mike Metlay and me over to the Cathedral Pipes booth at the 2014 AES show, beyond the bright glowing microphones, was this power supply. You should know that Dr. Metlay is a former nuclear physicist, and this power supply looks like a Geiger counter, so it was a "moth to a flame" thing. [*So sue me.—MM*]

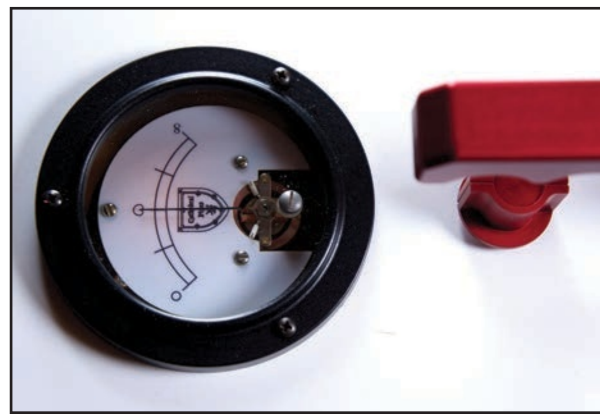
In addition to its large red aluminum handle and power switch, covered by a safety mask like a fighter jet's missile arm switch, the power supply has a unique way of selecting its polar pattern. The Notre Dame has a fully variable polar pattern that is selected by a red Neve-style knob, but rather than boring markings on the top of the unit for showing the chosen pattern, this power supply uses a repurposed voltage meter whose needle moves smoothly from Figure-8 through Cardioid to Omni and all points in between.

Sonically this design is smooth and very finely adjustable, as you can tune the pattern to the precise sound you are after. There is no click or thump as patterns are selected, as there would be with traditional switches, although the pattern takes a few seconds to catch up to your setting when adjusted.

In use

Bright lights and bling aside, the Notre Dame also delivers on sound in a major way. It is a big sound with a full rich tone: the lows are deep and thick, the mids are solid and punchy, and the top end has a nice smooth rounded vintage thing going on. It's not dark at all, but it has zero bite and also a distinct lack of high-end "air".

Over several weeks, I put it through its paces on lead and backing vocals, as a front-of-kit drum mic, room duties, picked and strummed acoustic steel string guitar, bass cabinet, shakers, tamborines, and congas.



For comparison I put it next to a pair of other high-end tube U47-style mics, an Australian Beesneez T-1 Tribute and the Pearlman TM-47 (Reviewed September 2014).

None of these mics is a full-on clone, but they are all greatly U47 inspired as well. Just as no two vintage U47s sound the same, each of these three mics easily had its own thing going on while simultaneously sharing a similar sonic weight and feel.

As I mentioned in my previous Pearlman review, the T-1 and the TM-47 both exhibit a similar low-end weight and midrange punch, with the T-1 being a tad rounder and more solid and the TM-47 having more air, making it more easily "stackable" over multiple overdubs. Throwing the Notre Dame into the mix, I found it to have a touch more low-end girth than the T-1, while its top end was closer in tone to the highs of the TM-47, but a hair smoother and less airy to my ears. There is no clear "winner" here; I found it fascinating how each mic's sound was audibly unique and yet beautiful.

This mic, like any good U47 inspired design, will work on anything, but in a very forward, full way. I liked it more on lead vocals, where it was fully front and center, but a little less so for stacked backing vocals, where I prefer something a tad more open like a C12 or Telefunken ELAM flavor. The Notre Dame is a killer voiceover mic and has a wonderful deep low proximity effect.

On acoustic guitar the Notre Dome captures a huge classic round fullness, but it would not be my first choice if I was after bright and jangly acoustic tones. All in all the same could be said for most sources, which make it a great choice on bass cabinet, front of kit, and non metallic percussion.

Loud and proud

This mic makes a statement in every way possible. First-time clients, walking into a session and seeing the whitish glow from this shiny beast, will be intrigued, impressed, and hopefully inspired. Sonically, if you want big front-and-center beauty, the Notre Dame sounds as impressive as it looks.

Finally, I find the care and quality that went into the power supply, the package, and all of its extras to be second to none. There is no mic kit I have seen in a decade and a half of studio work and reviewing that compares to it... none!

What I really cannot believe is that the whole package comes in at $2400. I just don't see how that's even possible with its impeccable fit and finish, loads of extras, and classy sound—but I'm not going to argue about it, and neither should you. I can't wait to test-drive more Cathedral Pipes mics in the future!

Price: $2400

More from: Cathedral Pipes, www.cathedralpipes.com

Rescuing Audio From Tape Cassettes

Irreplaceable treasures are sometimes found in the unlikeliest of places

By Scott Dorsey

Now and then, you may run across old cassettes that need to be transformed into some format useful in the modern digital world. These days, people forget just how bad cassettes can sound—how they gave a bad reputation to the words "analog recording" for a few years during the beginnings of the digital transition. On the other hand, people may not realize that with a little bit of work it's possible to get workable sound out of cassettes.

In the analog world, media degrade with every generation of copying... sometimes severely, if one of several mistakes I'll mention below happens in the copying process. Therefore, the absolute number one most important thing you can do to improve sound quality is to get back to the original generation, or as close as possible to the original generation. If at all possible, get the original.

But maybe you have a pre-recorded mass-duplicated tape that you bought in Cambodia, of a local band that's so obscure you can't even find any reference to it on the Internet. Or maybe you have a copy of a copy of a recording of your father from when he was 12, and the master was flushed down the toilet by your uncle when he was 6. If that's all you have, that's what you work with.

Step one: proper playback

The first part of the rescue process is to play back the tape, and as Dale Manquen says, the most important thing that a tape recorder does is move tape past the heads. It needs to do it cleanly, with accuracy and stability... and, sad to say, cassettes are not known for this.

One of the amazing things about the cassette is that most of the tape transport is inside the cassette cartridge. The bad part is that if the cassette itself is damaged, you get tapes that squeal or pop or flutter. When this happens, the tape itself needs to be taken out and transplanted into a new shell that does not squeal or rattle. This isn't too difficult a task, but it requires a steady hand; if you are starting with a cassette that is held together with glue instead of screws, it can be an adventure to get it open. (A Dremel Moto-Tool helps.) See Figure 1.

So let's assume that you've managed to get the tape in a good shell and you have it in as good condition as you can. When you put it in your tape machine and play it back, you have to deal with the next problem: azimuth adjustment.

The most severe problem with moving tape past heads properly is keeping the tape completely perpendicular to the heads, both in recording (which you don't have control over) and in playback (which you can control on reel decks but often can't on cassette players). If the original recording was made without the tape completely perpendicular, or it was recorded in one shell but transplanted into another shell with slightly different geometry, then we say it has azimuth error. You'll need to adjust the azimuth angle on the tape head mount to compensate for it.

Figure 1: Cassette shells held together with screws (left) are much easier to work with than shells held together with glue (right).

If you have a Nakamichi Dragon tape machine, it will automatically compensate for azimuth errors. If you have a million cassettes to transcribe, get a Dragon—or contract out the rescue job to somebody who already has one.

If you don't have a Dragon, you're going to have to find the azimuth screw or nut, adjust it for each tape you're rescuing, and then afterward put on a standard tape (one that was originally recorded on that deck before you messed with the azimuth) and adjust it back to its nominal position. With most consumer machines, there are only a limited number of times that you can do this before you wear out the screw threads, but that's fine if you're only doing a tape rescue now and then. There were some machines made with vernier azimuth control for playback of old tapes, but most of them came off the market when the Dragons appeared. They still turn up occasionally, though.

The technique is to listen in mono, not stereo. Sum both channels to mono, and adjust the head azimuth screw for the best high-frequency response. Having an FFT display on a DAW can make this easier, but you can do it by ear.

If you were listening in stereo, you'd hear the stereo image moving from side to side as you turned that screw, advancing or delaying one channel with respect to the other. You'd also hear comb filtering in each channel, but it wouldn't be as severe as the comb filtering you'd hear in mono. Putting playback in mono exaggerates the comb filtering because you're listening to a wider area of the tape, and listening for that (or seeing it on the FFT) is the easiest way to make sure it's correct. See Figure 2.

Once you have set the right azimuth for the tape, play it back and record it into your DAW. If possible, turn the Dolby noise reduction off first; we'll talk about that in a moment.

Now, there is software out there that claims to do "azimuth correction" after the fact, which adjusts the two channels to center the stereo image. That software, though, is only looking at the two channels individually, not each part of each track, so it cannot eliminate the comb filtering that is heard in stereo. Get the transcription correct rather than relying on bandages after the fact to try and compensate for transcription errors.

If you have a second generation tape instead of the master, you can't really compensate for the azimuth errors on the first dub, but you can only do what you can do. Have courage!

Step two: digital cleanup

Get the absolute best sound quality going into the computer that you can get, because once it's on the computer you can't add anything, you can only take things away. And the first thing you will probably need to take away is the Dolby noise reduction.

Dolby B noise reduction is what made cassettes practical as a music medium. Remember, the cassette was originally intended for portable dictation machines. It was never intended for music and it was not designed to have a lot of dynamic range. Dolby B changed all that.

The vast majority of cassettes out there were encoded with Dolby B, and the Dolby encoding is not any great secret sauce. It's a "dynamic equalizing process", which is to say that it's an equalizer that increases the high end on soft signals without increasing it on loud ones. When played back, there is an inverse function applied and the two cancel one another out. In the process, high-frequency noise at low levels is reduced too. The problem is that in the cassette world, because the levels have to be perfect and the response has to be fairly flat for the Dolby system to work, often the cancellation is *not* perfect and the end result is high-frequency pumping.

The first key to avoiding the pumping is to get as good a playback as possible with the azimuth correct, which you've already done at this point. The second key is to make sure the level going into the Dolby decoder is correct. You can't do that with consumer cassette decks, which just have a fixed decoder with no adjustments or calibration.

However, if you have transferred a file into the computer without decoding it (i.e. with Dolby disabled), you can do the decoding after the fact in software, fiddling with the levels of the decoder until you hear as little pumping as possible. This is a bit touchy and requires careful listening, and if you're not willing to do it properly you might just be better off not decoding it at all. But if you want to do it right, there is free software out there that will do it. Hans van Zutphen has a plug-in called Tape Restore Live, which has some silly functions added to it, but which does a very good job of Dolby decoding.

If you aren't sure if a cassette has Dolby encoding on it, try the decoder and listen. If you can't get it to stop pumping, leave it unencoded. On multi-generation tapes, you'll sometimes encounter tapes that were decoded twice without encoding them (or were encoded twice without decoding them) in the process of making copies. On those, you just do what sounds best and hope.

Figure 2: These oscilloscope patterns show the results of azimuth adjustment. The left channel signal controls one axis of the display, the right channel controls the other. When played back in mono, you would ideally see a perfectly straight, thin diagonal line; any bulging indicates that the azimuth is off. On the left is a tape with azimuth out of whack; on the right is a tape with azimuth almost perfect. If you don't have a scope, listen for high frequency response and trust your ears.

Step three: there is no step three

Cassettes are what they are, and they aren't exactly the highest of high fidelity, but there is material on them that needs to be preserved. It's always worth watching someone's face when they hear their father's audio letter from Vietnam restored, or the performance of their first band in high school. There's a lot of material out there—don't let it get away.

Scott Dorsey (dorsey@recordingmag.com) is a recording engineer, electrical engineer, aerospace engineer, and probably several other kinds of engineer as well—we wouldn't be surprised if he could drive a train—living and working in Williamsburg, VA.

READERS' tapes

By Marty Peters

(and Uploads, CDs, MiniDiscs, DATS...)

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Media can be submitted physically (as in an actual recording through the mail) or online at our website. For online submissions, please go to www.recordingmag.com and click on Readers' Tapes, then select "Submit Your Recording" and fill in the requested information. We accept MP3 and AAC files of up to 5 MB size. File bitrate is up to you but we strongly recommend a minimum of 128 kbps; note that the higher the bitrate, the shorter the song that will fit in the 5 MB limit. You're free to submit an excerpt of a longer song if that helps!

Send physical submissions to: Readers' Tapes c/o Recording Magazine, 5408 Idylwild Trail, Boulder, CO 80301. Please be sure to include: a) a CD, CD-R, cassette, DAT, or MiniDisc with only one song preferably no longer than 3:30 in length (or tell us which track you want reviewed); b) a credit list (who did what); c) a list of equipment used. Remember that CD-Rs with unevenly applied paper labels, smudges, or scratches won't play back reliably.

PLEASE state which part of your contact info we can publish (address, phone, and/or email)—if you don't tell us precisely, we won't print anything at all.

John Lindeman

Equipment: 2011 MacBook Pro with Apogee Duet interface running Avid Pro Tools 11, Apple Logic Pro X, and Waves plug-ins. AKG C414 and Sennheiser MD421-II mics, KRK Rokit 5 monitors. Fender mandolin, Taylor 414ce guitar. Bass and drums created virtually in Logic.

Music: "Anymore" is a female vocal "folk" song. Marissa Melton sang the vocals, Taylor Ferrell played the guitar, the fiddler is Stacey Sinclair, and Joe Ornstein rounded things out on banjo. John handled the bass and drum programming and mandolin part, as well as the recording, mixing, and mastering.

Recording: Those of you familiar with the music of the great British group Fairport Convention will recognize the overall vibe of John's submission. The track has an organic feel in keeping with the instrumentation, though we found the mastered mix to have a rather distant quality through our monitors, lending the track a very "live recording" feel. The individual sound sources were recorded in different places rather than together as a unit, but John's ambient/reverb treatment of them would suggest that this was the sound he was after. If there was layering/overdubbing involved, it is being masked to a large degree by the ambience here.

What we are left with, then, is a sound akin to that achieved by "tapers" at live concerts—you know, the tie dyed folks that are always setting up their stereo boom stand rigs around the summer festival sound boards! Not that there's anything wrong with that, mind you—just callin' them as we see them.

Suggestions: In our years behind the wheel here at Readers' Tapes (17 of them, as of this issue!), we can honestly say that no two submissions received have ever been the same. Have there been multiple tracks in certain genres? Absolutely! And yet each one of you has a skill set and mission statement that is as unique as your own DNA.

Readers' Tapes are also online—listen for yourself at www.recordingmag.com!

Part of the fun on our end continues to be trying to put ourselves in your mindset, in an attempt to not only advise, but equally important, to understand just what was going on behind the scenes of your creative process. Was John attempting to construct a "live" recording here, or did the combination of room ambience and overall mix balance push the sound in that direction? While we'll never know for sure, it's pretty cool to think about how the rest of the listening world interprets the recordist's choices, both musically and technically, isn't it?

Summary: Got live if you want it.

Contact: John Lindeman, jlindema@me.com

Jeff Lee / Skyline Hotel

Equipment: Mac with Avid Mbox 3 interface running Avid Pro Tools with Waves Mercury Bundle, Celemony Melodyne, and Antares Auto-Tune. Neumann U87, RØDE NT5, and Shure SM57 mics, Vintech 273 preamp/DI, Yamaha HS50 and KRK VXT8 monitors, Audio-Technica ATH-M50 headphones. Custom Shop Fender Stratocaster through Marshall JCM 800 amp with Celestion G12-65 speaker, Gibson Hummingbird acoustic guitar, American Fender Jazz Bass. Keeley Compressor, Vanamps Reverbamate, MXR Carbon Copy pedals. Roland TD12 MIDI drums playing Native Instruments Abbey Road Drums Sparkle Kit.

Music: "Better Than Alright" is a male vocal rock song. Jeff was the OMB on the project.

Recording: A good strong effort here, one that could be made even more successful with some slight mix rebalancing, and a few touchups to the drums. Before we dive into the particulars, we would like to commend Jeff on the thorough production/equipment notes that he provided us. Detailed step-by-step information like this gives us great insight into James's work methods—always a great learning tool for us!

Cool, then—on to the nuts and bolts. Lots of good stuff going on here, including the sweet vocals garnered from the Neumann U 87/Vintech 273. This is top-quality gear, folks, and it sounds it. The vocal is present yet smooth as silk, with the perfect touch of compression. We also dug the great "quack" sound coming from James' Custom Shop Fender "parts-caster" Strat/Marshall JCM 800. This is a classic combo, and James makes the most of it tonally and with his performance.

Unfortunately, through our monitors, the volume on the guitar seems excessive throughout the track. In many sections it registers louder than the lead vocal, adding to the mix balance issues mentioned earlier. Moving on, we found the rhythm section to be somewhat of a mixed bag. We would have liked to hear a bit more snap to the kick drum and a bit more midrange presence on the bass guitar tone.

Suggestions: James tells us in his production notes that he was pretty pleased with the way his track turned out, and well he should be! He has chosen his gear wisely, and more importantly, made it work for him in a way that sounds effortless.

As far as suggestions go, our desire to hear the kick and bass brightened up may be as much a matter of personal taste as not. We are certainly splitting hairs here. As for the mix balance, a sound as distinctive as Jeff's electric guitar is in no danger of being driven to the shadows. We urge him to set it back into the mix a dB or two as part of the ensemble, raising its volume during the solo as is proper.

Summary: Well done, sir!

Contact: Jeff Lee / Skyline Hotel, skylinehotelmusic@gmail.com

TAXI A&R INSIDER

FEBRUARY 2015

TAXI Road Rally 2014

TAXI's Road Rally Conference 2014: A Photo Diary

Photos by James DiModica and Ryan Taalbi

Pictures are worth a thousand words – we all know that. Here are some great photos that tell you everything you need to know about the TAXI Road Rally held last November in Los Angeles. There's just one thing the photos don't tell you. The Road Rally is free for all members and a guest of their choice. We hope you can join us next year. —Michael Laskow, CEO, TAXI

Legendary Music Supervisor Maureen Crowe receives her well-deserved Lifetime Achievement Award from Michael Laskow at TAXI's Road Rally 2014.

Rock Mafia may be huge, but not too cool to spend some quality time hanging out with TAXI members and taking their music when they finished their excellent interview. Antonina Armato (foreground) and Tim James (background) could not have been more gracious or inspiring!

Briagha McTavish not only showed us how good her songwriting is during this open mic performance, but as you can see in this photo, her passion was off the charts!

Executive Vice President of Creative at Rondor/Universal, Kevin Hall, dishes out some great advice on the A&R/Publishing Panel. How often do you get your music played for somebody at his level?

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Hollywood music supervisors (left to right) Jon Ernst, Michael Laskow (TAXI, CEO), Tracey McKnight, J'ean Demery, and Robert Jordan pose for a group shot right after hearing TAXI members' music on the Music Supervisor Listening Panel.

Mega hit songwriter/producer Antonina Armato of Rock Mafia makes an emphatically impassioned and encouraging point during her interview with husband, co-writer, producer, business partner Tim James (right).

You'll have to take our word for it, but that's the back of mega music supervisor, and V.P. of the Guild of Music Supervisors, Tracy McKnight's head, as she takes music from TAXI members after finishing her panel. Tracy was an amazing panelist!

Making and Pitching Advertising Music panelists Garry Smith (left), Heather Kreamer, and Nick Murray take a moment to pose for a shot with TAXI CEO Michael Laskow (2nd from left) after their incredible panel wrapped up.

Music library owner, and successful composer in his own right, John Fulford, looks like he is saying, "Well, the first thing you could do to increase your chances of getting this track synced would be to…" Wouldn't you like to know what advice he was giving these members during this One-to-One Mentor session?

Apparently, Mega-TV Composer/Songwriter/Producer (and frequent Road Rally panelist) Adam Zelkind knows where the best "After Parties" are being held. We heard there was a great party going on in one of the suites upstairs in the hotel, but we didn't get an invite.

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"Before attending, I thought that TAXI was just another pay-to-play rip-off that was trying to take advantage of writers and our dreams. Wow, I have never been so wrong. Not only am I convinced that the TAXI Road Rally is the best music convention I have ever been to, I am convinced that TAXI is the holy grail for anyone who wants a real career as a songwriter, artist, producer, or composer in this industry."

—Jeff Shane

"If so many musicians only knew the value of your company and had the courage to step outside their comfort zone, they would be able to start walking the path that they have only been dreaming about. I look forward to many more years to come and thank you so much for everything that you and your staff do for us."

—Robbie Hancock

"I was told that Rally would be a life-changing experience and it proved to be absolutely true."

—Adriana Lycette

"To get such an insightful, inside look at how the industry interacts with TAXI, was incredibly eye-opening. I learned a lot from the panels."

—Michael Breze

"I had hoped this Rally would be a transformative experience and it was that. In every respect it surpassed my expectations."

—Gareth Ebbs

From left to right; Bob Mair (Black Toast Music), Jeff Freundlich (Wild Whirled/Fervor), Ben Davis (Defacto Music), and Cindy Badell-Slaughter (Heavy Hitters Music) discuss an instrumental track they just heard on the Music Library Listening Panel.

BY MIKE METLAY

What makes for a "good" subwoofer? When I review a sub, I look for a combination of factors: flexibility in wiring, so it can easily be integrated into a wide variety of studio monitor setups; a sensible and comprehensive set of controls, to allow maximum flexibility when adjusting the sub to match your monitors and room; and bass that sounds realistic and nuanced when successfully integrated in your system. With those criteria in mind, I ran the new PreSonus Temblor T10 subwoofer through its paces...

What's a Temblor?

"Temblor" is a Spanish word meaning "earthquake", commonly used in tremor-prone areas of California. That's an apt description for this sub, which can rattle the fillings right out of your mouth if you tell it to. It's a big box, 12.6" wide and 15.75" high and deep, weighing a hefty 40 pounds. A reasonably nonresonant MDF cabinet holds a 10" front-firing glass composite woofer, just above a generous port that reinforces the speaker's low-end response. With a Class AB power amp that delivers 170W RMS / 250W peak, there's some serious boom awaiting you.

PreSonus Temblor T10 Subwoofer

Lots of boom for your listening room

The rear of the T10 offers the flexibility I look for in a subwoofer primarily designed to support stereo monitors. Stereo inputs are available on balanced XLR, balanced $^1/_4$" TRS, and unbalanced RCA; the unbalanced inputs are blended with the balanced ones, with the TRS superseding the XLR if anything's plugged in there. A rear-panel Input Gain pot sets level from –30 to +6 dB.

The outputs include stereo XLR and TRS plus an XLR Sub Out for feeding a second sub. Controls include an 80 Hz highpass filter (HPF) switch, polarity invert, ground lift, footswitchable subwoofer bypass, and a Low Pass Filter knob with a frequency range of 50–130 Hz. The spec sheet claims a frequency response of 20—200 Hz (no tolerances given), better than 98 dB signal to noise ratio, and 113 dB maximum SPL at 1 meter.

Putting the T10 to work

Hookup is the same as for any subwoofer with built-in bass management: you feed your audio from your mixer or interface to the T10's inputs, and run cables from the outputs to your stereo monitors. You then adjust the controls to get the smoothest possible response in the low end for your listening position in your room. As the manual warns, you should also be prepared to move the Temblor around to find where it speaks most evenly at the listening position, and if possible, be prepared to use a pink noise source and SPL meter to calibrate listening levels at a reasonable volume.

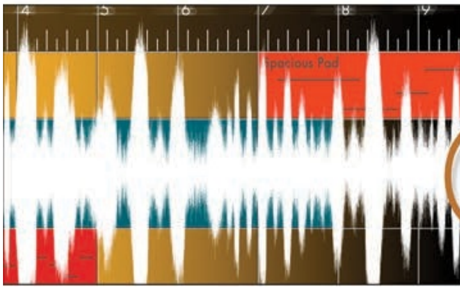
A lot of the controls are in the realm of "if it sounds better one way than the other, leave it there." Does the bass sound stronger and more defined when you flip the polarity switch? Good! Is there ground loop hum that is alleviated by the ground lift? Then let it do that for you. By the way, the lift switch actually puts a 1 kilohm load across the balanced inputs; it doesn't mess with the actual ground for the unit's power cable, which on an amp this big would be asking for Dr. Forrester's Deep Hurting.

Do you want your stereo monitors reproducing audio as low as they can go, or do you want to let the T10 do all the heavy lifting down there? If the latter, set the Low Pass Filter to 80 Hz and flip the High Pass Filter switch, so the sub hands off to the mains at that frequency. Otherwise, you can dial in the bass by adjusting the Low Pass Filter to where the overlap sounds musically smooth with no bump or dip in power. I preferred to leave it set quite low, as my ADAM Audio A5X mains (reviewed September 2014) are 3 dB down at the Low Pass Filter's 50 Hz lower limit.

The included remote bypass footswitch is something every serious sub should have. Defeating the sub so you can hear what it's doing helps you assure yourself that you're hearing real bass rather than "impress the client" thunder. The backlit blue PreSonus logo on the front of the T10 flashes blue and red when it's bypassed; the 80 Hz HPF is also disabled.

How does it sound? I'll admit that just for fun, well after hours in my studio at our offices, I cranked up the T10 and watched the windows and walls vibrate. After that, it was time to get serious. I dialed it in at lower gain with the Low Pass Filter set to just touch up the lows in the ADAMs, and the result was exactly as it should be. All of a sudden, there was extended, present, believable bass that the 5" woofers of the ADAMs couldn't deliver, and my mixing decisions were instantly more accurate and translatable in the lows. A good subwoofer is supposed to do that, and the Temblor T10 delivers the goods.

Price: $399.95
More from: PreSonus, www.presonus.com



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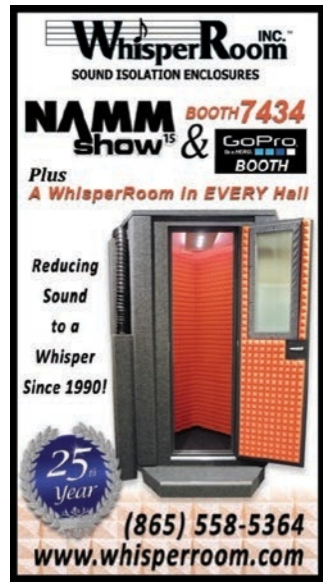

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Vocal Performance

Guest editorial by Beto Hale

In this issue, several of my colleagues have gone to great lengths to share their expertise on how to tweak your vocals to obtain the best possible sound using the gear you have. They've explained, with amazing detail, compression ratios, EQ settings, gain structure, and many other aspects. I would like to offer a perspective on a subject that is often overlooked in gear-oriented magazines like *Recording*: vocal performance.

No Neumann U87 going through an LA-2A or high-end Waves plug-in is going to transmit what a vocal should: emotion.

In 1995, the three surviving Beatles released two singles, "Free As A Bird" and "Real Love". Both of these songs started out as demos that John Lennon had recorded using a cassette-tape recorder in his apartment. In the demos you can hear his vocal and a piano. The remaining lads, produced by Jeff Lynne, added their own parts and arrangements to the songs, and one of these final tracks, "Free As A Bird" went on to win a Grammy for Best Vocal Performance for a Duo or Group the following year.

That's right: John recorded his vocal on a *simple tape recorder*, playing piano at the same time. He went through no fancy gear. This same vocal, with the piano (!), was used on the final arrangement. Of course, the sound of John's voice was processed to make it work with the final version, but he tracked it in the most rudimentary way possible. Yet even as a demo, he transmitted what he needed to: a message, a feeling.

I'd like to encourage all singers, producers and engineers out there to never forget to pay attention to the *performance* of the vocal you are recording, and that no Neumann U87 going through an LA-2A or the fanciest Waves plug-in is going to transmit what a vocal should: *emotion.* Remember: you are tracking the human voice. It all starts with humans.

As a singer, how do you achieve the level of someone like John Lennon, Frank Sinatra, or Sinéad O'Connor? Well, most of us never do. This doesn't mean, however, that you cannot push your vocal talent to its absolute maximum potential, by performing live and practicing and exercising your instrument at home. Melodyne or Auto-Tune may fix intonation problems, but transmitting an emotion is something that, as of yet, no hardware or software product can emulate convincingly.

Where to begin? For starters, in this era of instant access, you can listen to and watch any singer, in any style, perform for you—in concert or in the studio, in the privacy of your own home. All you have to do is type in their name on YouTube's search engine. Put on a pair of good headphones and learn from the masters. If you are into reggae, check out Bob Marley; if you love jazz, listen to Ella Fitzgerald. Let them teach you. Imitate them. What would Paul McCartney be without Little Richard? And Prince without James Brown? Both singers absorbed their respective models like a sponge; they *became* them, and then developed their own styles from there. You should do the same.

At Berklee College of Music, where I was lucky to spend four years, endless hours were spent at the library watching videos of whoever we wanted to become. YouTube did not exist. YouTube is the new college.

I recommend you focus on live performances. These singers will become your teachers, and you want to see and hear them doing what they did best: singing for an audience. Starting in the 1960s, vocal performances released on studio albums were more and more produced, which means you might be hearing take 27 of the vocal. But a live performance simply is what it is. Of course, listening to perfectly produced recording will teach you a lot as well, but at least in their original intent, recordings were supposed to be "records"—as in "documents"—of live performances.

"But I have my own style; I do my own thing," I hear many singers tell me. Well, actually, no one has a pure style to begin with. Even if you were raised by wolves in the middle of a forest, I truly believe you can always learn from those who have come before you.

So what's next? Experience. Sing live as much as you can and always record yourself. There's no tool quite like the raw recording of a live show to learn from your mistakes and your successes. Make sure you get feedback from trusted people: you want to know if you are in tune, if your volume is consistent, if you are blending with the band and other singers. And remember that nuance is crucial. A subtle change in your delivery can go a long way... an elongated phrase, a burst of energy.

Hire a vocal coach. A good one can do miracles. I was lucky to study with some great coaches, and I still apply many of the tips I learned from them: how to warm up the voice, what drinks and foods to avoid, live and studio mic technique, and exercises for improving pitch.

And of course, sing in a studio as often as you can. To engineers and producers, I say: send them back home to study if they can't sing yet. You'll be doing the singer, and the listeners, a favor.

There will be plenty of time to worry about compression ratios later. First, focus on the voice.

Beto Hale (beto@recordingmag.com) is Recording's *Los Angeles-based Editor At Large. He's also a recording engineer, producer, former Editor-en-Jefe of* Recording's *Spanish-language sister publication* Músico Pro, *and a recording artist with several albums out under his own name. Learn more about his work at www.facebook.com/beto.hale and at www.betohale.com.*

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