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ARE LOOPS CHEATING?

Many of us use rhythm loops in our compositions, but at what stage does relying on them become cheating? After all, products such as Apple Loops offer not only rhythms but also complete musical phrases, bass lines, vocal segments and so on that you can stitch together into complete tunes without ever composing an original note of your own. There's no copyright issue when combining Apple Loops or other licensed commercial products to create finished tunes, but does it still feel like your composition?

The answer is, inevitably, somewhat vague and depends on the perspective of the user. The same question used to be asked of sampling in general, but I have to confess that I sometimes find loops very useful. Just a tiny segment taken from a loop can be used in an entirely different context, and taking something like a clock's tick from a sound effects library and then layering it with a TR808 snare doesn't make me feel bad at all. Or maybe taking the attack from a vocal or instrumental phrase and turning it into a repeating rhythmic element? That also feels like fair game.

While I'll almost never use a loop, other than perhaps a rhythm loop, in its original form, you can often get musical inspiration by playing along to one, even if you then decide to drop the loop entirely from the finished composition — we all have to get our inspiration from somewhere. In the case of Apple Loops, which are designed to conform to tempo and can be changed in pitch, there's also the option of converting them to regular audio so that you can slice them up and then transpose or rearrange the different segments to create something new. Once saved as an audio file you can also use time stretch or reverse — or in the case of a monophonic melody you can apply drastic pitch correction, again to create something unique. Further processing via spectral reverbs, rhythmic choppers or stepped filters can take the material so far from its original source that you can really make it your own.

Even so, I still feel a touch guilty when using loops created by somebody else as source material, even when I change them so drastically, but I have to admit it is a level of guilt that I'm learning to live with. **///**

Paul White

Editor In Chief

“Many of us use rhythm loops in our compositions, but at what stage does relying on them become cheating?”

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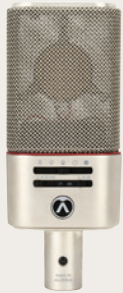


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188 CLASSIC TRACKS

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— WIN —

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Osmose — keyboard technology takes a giant leap

There's always been something slightly unsatisfactory about controller keyboards. Even top-of-the-range models, decked out with ribbons, continuous encoders, trigger pads and aftertouch, often fail to replicate the immediate, intuitive expression available to skilled players of simple acoustic instruments like violins or acoustic guitars. And even the best keyboard controllers, in the hands of the most expressive players, can be let down by the sounds to which they're mapped; after all, the number of hardware control methods is irrelevant if they're not assigned to samples or sounds that respond to those controllers. But now the expressive possibilities available to keyboard players have taken a great leap with the announcement of Osmose, from French company Expressive E.

At its most basic, Osmose is a 49-note keyboard with full-size keys that transmit polyphonic aftertouch, compatible with the MPE (MIDI Polyphonic Expression) and MPE+ standards. But each key can also generate its own polyphonic initial pressure data — so the notes in a chord or melody can sound radically different

depending on whether each note is tapped, hit, stroked or gently depressed. Each key can also be wiggled from side to side to generate vibrato. What all this can do for performance is best heard rather than described: see the video clips at www.expressivee.com/discover-osmose.

But Osmose isn't just a highly capable controller keyboard. It also contains an immensely versatile 24-note polyphonic synth based on the EaganMatrix designed for Haken Audio's premium hand-built Continuum Fingerboard controller a few years ago, and custom-adapted for Osmose. So there are also plenty of sounds on board which use the EaganMatrix's full range of synthesis techniques (including virtual analogue, FM, additive, subtractive, granular and spectral synthesis, plus physical modelling) to take full advantage of Osmose's expressive capabilities. A computer-based editor allows you to delve into the complex modular world of the EaganMatrix if you wish, but connecting Osmose to a computer isn't a must; you can just turn it on and play. Sounds can be layered and split across the keyboard directly from Osmose itself, and six front-panel

encoders (around the screen) allow you to tweak sounds in real time.

SOS saw a very early Osmose prototype at Superbooth 2019, and even then, it represented an impressive leap for keyboard tech. Other controllers have offered intuitive simultaneous control of multiple playback parameters, but not in such an accessible keyboard-based form, and certainly not at this kind of price — Osmose is due to retail for \$1799. That's a lot to pay for a controller keyboard, but when you consider what else Osmose offers — the incredibly deep built-in EaganMatrix synth engine and custom-tailored sounds, and the unique expressive potential offered by the keyboard and MIDI spec — it starts to make sense. As is often the way these days, impressive discounts are available if you pre-order Osmose early, but only until the end of December 2019 (see Expressive E's site for details).

For more comment on Osmose, see our detailed online report at <https://sosm.ag/ExpressiveEOsmose>. The keyboard itself is expected to arrive in summer 2020; we'll bring you a review as soon as possible. www.expressivee.com

New endorsee for Acustica

In addition to debuting their Modula virtual mixer (see our lengthy preview starting on page 36 this month), Italian modelling plug-in developers Acustica Audio have recently announced the results of a three-way collaboration. It involves them, acclaimed mix engineer Dave Pensado (Beyoncé, Michael Jackson) and Californian EDM production gurus Studio DMI (who have worked with Acustica on various plug-ins, including a recent update to Diamond Color EQ).

The logically named Pensado EQ, which is also the first plug-in the hit engineer has explicitly endorsed, is a VST/AU/AAX plug-in processor based on Acustica's Core 14 modelling engine, and was apparently created by undertaking an "almost obsessive" analysis of Pensado's workflow and mixing approach.

The finished plug-in offers four bands of switchable EQ, each apparently chosen and specified to emulate favourite hardware EQs in Pensado's mixing armoury, but here anonymously labelled simply 'Low', 'Low Mid', 'High Mid' and 'High'. There are also straightforward high- and low-pass filters, plus two emulations of different mic preamps, also based on regular Pensado workhorses.



The plug-in ships with presets created by Pensado himself, and Acustica promise more of these to come, as well as a big free update to the facilities, by the time of the NAMM show in mid-January. We'll add more on that to the online version of this story (see <https://sosm.ag/NewsAcusticaPensadoEQ>) when we find out what's involved at NAMM — rumour has it that it could involve machine learning and AI, which was used to assist in the development of various recent Acustica plug-ins such as Cola.

The pre-NAMM version of Pensado EQ is already available, priced in full at €289 from the Acustica website. An introductory price offer, which saw it launched at just €130, may still be running by the time you read this — check the company's online store for details.

www.acustica-audio.com

Behringer: the power of 3(03)

When we learned of the launch of Behringer's all-analogue, ultra-affordable TB-303 clone in the *SOS* office, it was the price that attracted the most comment. At \$149, the TD-3 is amazing value at just under half what the original cost in dollars (\$395, unbelievably) on its launch in 1981 — nearly four decades ago! — and well under a 10th of what a working original Bassline often fetches on eBay these days.

What's more, it appears that in addition to cloning the original voltage-controlled oscillator, filter, amplifier and other circuits, the TD-3, like the RD-8 808-style drum machine clone reviewed on page 28 this month, adds features not included on the original Roland gear (ignoring for a moment the TD-3's SH-101-like silver, blue and red colour variations, which definitely weren't options in the 1980s). The TD-3 has 5-pin DIN and USB MIDI (which was absent in any form on the original 303, of course), and the TD-3's audio input, unlike the Mix In socket on the TB-303, passes through the filter, allowing you to process external audio using the Behringer box. There's also a built-in distortion modelled on the original Boss DS-1 guitar pedal, whereas the original had no built-in processing at all. The on-board sequencer has an enhanced spec with 16 steps and a maximum of 250 possible patterns, but is apparently programmed in the same 'idiosyncratic' way as the TB-303. This was part of the slightly unhinged charm of the original, but if it's just too irritating for you in the 21st century, Behringer have made the sequencer sanely programmable over USB using their free-to-download Synth Tool app, which is also welcome.

As usual with Behringer's affordable clones, a firm shipping date has not been set, but retailers are expecting to receive the TD-3 "in the near future". At this price, it's virtually an impulse purchase; whenever it comes, we look forward to trying it out.

www.behringer.com





Peach Audio: mic pres from down under

Peach Audio, an equipment manufacturer from Sydney, Australia, is a name new to *SOS*, although the company have been in existence down under since the early 1980s. The company's first product to be sold internationally is the seemingly very flexible valve-based M196sx stereo mic preamp, also known as the Savannah.

Offering an internal switched-mode power supply for international operation and over 60dB of gain per channel, the new mic pre can accept mic- or line-level inputs, as well as high-impedance guitar signals via front-panel instrument jacks, and there's transformer-balanced I/O (behind the 'option' plate on the rear panel) for re-amping. 48V phantom power can be supplied on both mic-level inputs if required, and the input impedance on the mic input offers four settings to accommodate both 50 and 200 Ω mics.

The front panel sports switchable twin VU meters, plus dedicated controls for high- and low-pass filters, phase invert and mute buttons for each channel, and Mid-Sides output options (when the Mid-Sides button is pressed, the left line output carries the sum signals, while the right outputs the difference signals). There is even a choice of transformer-balanced and "cleaner-sounding" transformerless outputs, depending on your sonic preference, the different types accessed via a single two-way switch. Output attenuation is also available in switchable 8dB increments, so your amplified mic signals don't run too hot.

Savannah is available in the USA via Vintage King, retailing for \$2999. A review of the M196sx is scheduled to appear in *SOS* in the near future.

www.peachaudio.com

New for UAD v9.11.0

Universal Audio's latest UAD v9.11.0 upgrade for their range of interfaces and DSP accelerators brings with it the opportunity to buy three new plug-ins. There are two emulations of classic analogue studio hardware, and one handy UA model of a great Pro Tools studio plug-in: Sonnox's SuprEsser de-essing processor, which was reviewed in *Sound On Sound* back in September 2008 to praise from Editor In Chief Paul White (see <https://sosm.ag/SonnoxSuprEsser>). As on the original, a spectral analyser helps you pinpoint troublesome sibilance, and there's a dry/wet mix control so you can add some of the unprocessed signal back in if you feel the de-essed audio sounds too unnatural. However, the latency performance of the original plug-in has been much improved, with the UAD-based edition of the plug-in offering "near-zero" latency, according to Universal Audio.

When it comes to the two modelled hardware plug-ins, one emulates German guitar amp manufacturers Diezel's first product, the respected four-channel 100W VH4 valve amp first produced in 1994. The plug-in was developed by hardware modelling experts Brainworx and is fully



endorsed by original designers Peter Diezel and Peter Stapfer. The other models Avalon Design's much-loved VT737 channel strip: including the original's valve-based mic preamp offering up to 58dB of gain, the LED-based optical compressor, and the discrete four-band EQ.

As usual, all three plug-ins come with the UAD upgrade, but you have to pay to keep them active after a 14-day trial period. The Sonnox SuprEsser DS plug-in costs \$249 to buy, the Diezel VH4 Amplifier is \$149, and the Avalon VT737 is \$299.

www.uaudio.com

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[MOTU.COM/M2](https://www.motu.com/m2)

Sylpho: the expressive wind controller

French company Aodyo's innovative Sylpho wind controller was first noted in *SOS* in early 2015, and we covered improvements to the system a year ago (see <https://sosm.ag/Sylpho2018>). Now it's more easily available in the UK via Gear4Music, and in North America via Canadian store Long & McQuade.

To recap, Sylpho differs from other wind controllers, which simply sense blown air pressure. Here, air passes all the way through the instrument, making for a more natural feel for existing wind players without compromising those learning Sylpho from scratch. The French controller also contains inertial sensors, allowing players to alter the sound they're triggering by shaking, tilting or tipping Sylpho while playing, and a further expression slider on the side of the body outputs up to three MIDI controllers.

In North America, a Sylpho on its own now costs 1099 Canadian dollars from Long & McQuade, while a system with the Link wireless receiver is CA\$1614. A complete kit containing Sylpho, the Link unit, a carry case and two spare mouthpieces costs CA\$1754. www.aodyo.com



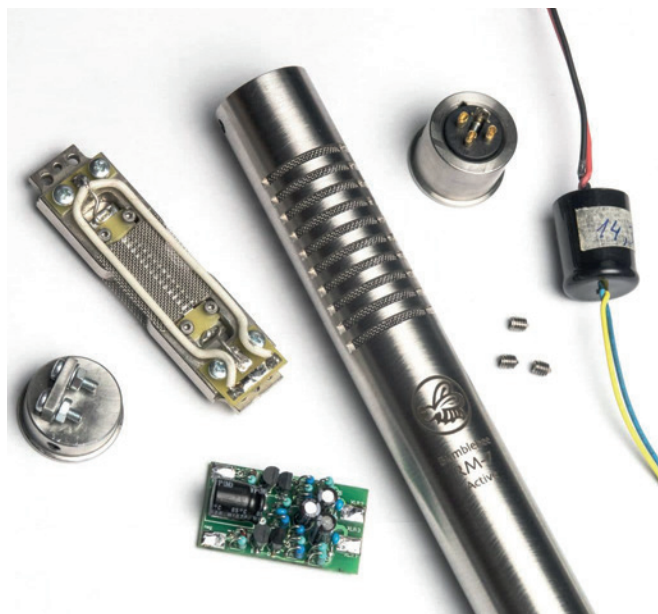
New Bumblebee DIY ribbon mic

Latvian DIY mic manufacturers Bumblebee Pro, known for their ribbon mic kits, DI boxes and preamps, have announced a new low-noise active ribbon mic kit, the RM7. The RM7 (below) incorporates a built-in preamp, is powered happily from any 48V phantom power source (unlike some ribbons), and, according to Bumblebee, produces output levels comparable to condenser mics. As with the company's other kits, apparently only basic soldering skills are needed to put it together.

According to company founder Artur Fisher, the RM7 incorporates several new design features compared to previous Bumblebee kits, including a new motor, the RE423, and thinner magnets, making for a more accurate, neutral high-frequency response compared to those of the company's previous ribbon mic kits, and also those of vintage ribbon mics.

The RM7 kit costs €375 excluding Latvian VAT, which is non-payable on US orders. Delivery charges vary, but are never more than €15 to the USA or Europe.

www.bumblebeepro.com



ProjectSAM Pandora focuses on horror

The fourth addition to ProjectSAM's Symphobia series of orchestral sample libraries aimed at composers of music for film is subtitled Pandora — and opening this particular box reveals, appropriately enough, an NKS-compatible 140GB trove of material suitable for use in action, thriller, sci-fi and horror films.

Many of the mapped samples — including spine-chilling rising and falling string motifs lasting from two to 15 seconds, dramatic crescendos on strings, brass, woodwinds and percussion, eerie violin textures and terrifying brass stabs — can be shaped by engaging a new Sound Design Mode, which allows users to add effects and processing, change the audio's attack or release, edit sample playback start positions, and reverse or timestretch the sampled material. Another handy feature is Adaptive Sync, new for Symphobia 4, which makes it easier to use Pandora's many developing textures with clearly defined endings; you identify the point where you want a crescendo or rising string line to end, and how long you need the cue to be in beats or seconds, and the software selects a performance of the right length from its internal database automatically. If you subsequently change tempo or move the cue forwards or backwards, the software adjusts the cue to compensate, selecting a different sample if needed.

Symphobia 4: Pandora is available now from the ProjectSAM website for \$659.

www.projectsam.com



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PreSonus expand Eris range to include headphones

The Eris name has been associated with PreSonus's affordable studio monitors since 2013, but now they've used the brand name for an affordable over-ear Bluetooth headphone model, the battery-powered HD10BT. With a frequency response covering the entire audible range from 20Hz to 20kHz, and featuring twin 40mm drivers, the new headphones also include built-in Active Noise Cancelling (ANC) technology for listening in less-than-ideal audio environments — and according to PreSonus, ANC can deliver up to 18dB of ambient noise reduction. Hardware controls situated in the centre on the outside of the right earphone allow you to pair the mic with a Bluetooth device, start and stop track playback, navigate to the next or previous track, adjust the headphone volume, and even answer a phone call if you're listening on a Bluetooth phone (the headphone even includes a small built-in mic!).

Offering up to 16 hours of operation from the built-in rechargeable batteries, the Eris HD10BTs are available now, cost \$129.95 a pair, and ship in a hard carry case with an included one-metre USB charging cable, a one-metre minijack-to-minijack audio cable and a minijack-to-quarter-inch jack adaptor.

www.presonus.com



MOTU launch affordable audio interfaces

MOTU have announced two new affordable audio/MIDI interfaces for use with macOS, iOS and Windows. The stereo-in, stereo-out M2 and the four-channel M4 offer:

- Low-latency audio conversion (2.5ms on 'a round trip' in and out of the interface while recording at 96kHz with a 32-sample buffer).
- Colour LCD-based metering for all I/O.
- Independent 48V phantom power and continuously variable gain on the dual mic/line/high-impedance XLR/jack 'combi' inputs on both devices.
- A headphone out with level control.
- Hardware (zero-latency) monitoring options.



On the output side, both interfaces feature balanced quarter-inch jack outputs and unbalanced RCA's (two of each on the M2, four of each on the M4). The M4 has an additional input/monitor mix control that allows users to set a balance between monitoring the live audio inputs to the interface and audio being played back from

the recording DAW.

Both new interfaces are already shipping, bundled with MOTU Performer Lite, Ableton Live Lite 10 and 6GB of free sample library content from providers including Big Fish Audio and Loopmasters. The M2 costs \$169.95, the M4 \$219.95.

www.motu.com

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Voltage Modular grows with P.moon modules

Voltage Modular, the virtual modular synth platform developed by Cherry Audio, is continuing to expand, with over 500 affordable modules now available for the system. One of their newest developers, P.moon, has recently introduced 14 new control modules for the platform. P.moon began as a user of the system and contributor to the Cherry Audio user forums, and developed his

own control modules for the system when he couldn't find what he wanted elsewhere, before deciding to offer them to others commercially.

The new modules include Pulser and multi divider controllers, button controllers and a CV meter utility that reads CVs in the system and displays them either as voltages or their pitched-note equivalents. There are also three kits made up of

assorted modules: one sequencer kit featuring step and tone event generators, and two toolkits featuring button and logic controllers, plus CV collection and limiting modules. Each module or kit is available for an affordable \$10 from Cherry Audio's online store, with the exception of the button controller modules, which are free! <https://store.cherryaudio.com>

DPA go binaural with 4560 mic headset

For years, fans of immersive audio recording techniques have known that if you don't have an expensive multi-capsule or dummy-head microphone, binaural sound can be achieved quickly and affordably by mounting microphones as close to your ears as you can — if possible, *in* your ears — and recording the output as a stereo pair. Because each channel has been captured almost from the same point as the natural audio 'receivers' on our heads, when the resulting recordings are played back on headphones, they can have a three-dimensional realism that is hard to replicate with other mic techniques.

DPA 4060 Core miniature mics have been a popular choice for binaural recording in this way because of their compact size, and now DPA have taken the logical next step and made a custom high-quality solution for binaural in-ear recording. The 4560 Core binaural mic



headset positions carefully matched twin miniature DPA 4060 mics on earhooks attached to an adjustable headband, so that the mics sit just outside the wearer's ear canal, as with iPod earbuds. If the 4560 is plugged into DPA's MMA-A USB audio interface, it can be connected to an iOS device, making

for a super-portable high-quality binaural recording rig that scarcely takes up more room than an iPod and its earbuds.

The 4560 headset will be on display at the DPA stand at January's NAMM show and will be available later in 2020, retailing at \$1099.95.

www.dpamicrophones.com/4560

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RME's original 2010 Babyface interface, reviewed in *SOS* April 2011, got an upgrade five years later to form the Babyface Pro (see <https://sosm.ag/RMEBabyfacePro>)... and now the desktop design has been improved again to make the 24-channel, bus-powerable Babyface Pro FS, the additional letters standing for RME's improved 'Femtosecond' clocking technology. At the same time, there have been various other upgrades, from an improved signal-to-noise ratio throughout to a reduction of six samples in latency to a mere five samples on the A-D side and seven samples on the D-A. For more information, see the link below. The new Babyface Pro FS retails for \$999, the same as the final retail price of the Babyface Pro before the FS version was launched. <https://tinyurl.com/RMEBFProFS>



Fans of London rockers Saint Agnes might like to check out the latest competition from mic manufacturers Lewitt, which gives budding mix engineers the chance to practise on the multitrack of a 'live in the studio' recording made at Metropolis Studios (above) earlier this year. Entrants have to produce the best remix of the band's 'Move Like A Ghost', and winners get to share new Lewitt mics worth over \$2500. But don't forget to enter before December 31st! www.lewitt-audio.com/mix



Italian Max for Ableton Live developers Amazing Noises (AN) have released a 10-voice resynthesizer and polyphonic granular synth with a built-in step sequencer and effects, capable of playing back up to 128,000 slices of a sample per second. AN say Grain Scanner (below) is: "designed for experimental noises, glitchy effects, alien textures". There are over 100 presets, plenty of source sounds and even the option of importing your own samples. It costs \$59. <https://tinyurl.com/ANGrainScanner>



As long-term readers of this magazine will know, since 2010 *SOS* Technical Editor Hugh Robjohns has been making regular use of an industry-standard Audio Precision (AP) audio analyser to perform spec tests and measurements during reviews. Hugh's APx515 is a pretty formidable piece of industrial test kit, with a price tag to match — but anyone who would like a cheaper way to test audio gear themselves and already has an ASIO-capable audio interface now has a more affordable option. APx Flex (above) is a software-only test solution that uses an existing interface for test signal I/O. The only hardware required is a dongle, the APx500 Flex Key, which is supplied with the software and connects via any spare USB 2.0 socket. APx500 Flex systems start at \$3000 for two-channel test software — contact AP for further pricing details. www.ap.com

The open-source code for Spleeter, a source-separation algorithm, has been released by the research team at Deezer, a music streaming service — and early experiments with it show it to be effective at (for example) separating vocals from mixed mono or stereo tracks. The code can be run under Linux, Mac OS or Windows and merits investigation, even though you have to use command line instructions. As *SOS* went to press, web-based front-ends using the code had already appeared at <https://melody.ml> and <https://moises.ai>. Check it out! <https://deezer.io>

Essential stand and support manufacturer K&M have announced a modular base plate and telescopic rod system for mounting live speakers. The heavy steel 26706 base plate (\$310) secures the telescopic mounting rod in one of three possible threaded mounts in the middle, edge or corner of the plate, so it can be used flexibly in small event spaces. The telescopic 24623 distance rod can be extended from 55 to 94 inches in length as required, and costs \$145. www.k-m.de/us

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**SAM INGLIS**

For as long as software mixing has been a thing, there have been things to make it less like software mixing. A pointing device isn't necessarily the best means of adjusting hundreds of parameters, often simultaneously, so all DAW software supports alternative controllers such as fader surfaces, jog/shuttle wheels, rotary encoder banks and so on.

However, the problem for any manufacturer making these devices is that 'support' means different things in different DAWs. Older standards such as MCU and HUI are reasonably universal, but offer only limited control, while others

Softube Console 1 Fader Control Surface

Softube's alternative mixing environment has grown faders — and they do much more than just fade!



The Console 1 Fader complements the original Console 1, adding transport controls as well as 10 faders that can be used for many other purposes besides volume balancing.

such as Eucon are proprietary. Add to the mix the need to control parameters in third-party plug-ins, and the fact that there are significant operational differences between DAWs, and you have a situation where it's almost impossible to create a generic controller that gets the best from each DAW.

1 On 1

Launched back in 2014, Softube's Console 1 presented a radical solution to this problem: a DAW controller that isn't a DAW controller at all. Console 1 is actually a controller for the Console 1 plug-in. In effect, inserting this plug-in on every channel in your mix creates another mixer — an officially licensed emulation of the SSL 4000, no less — that 'piggybacks' on top of the DAW's own virtual console. As the feature set of this mixer is determined by the plug-in rather than the host DAW, it's the same in all supported host programs. The Console 1 controller can thus be designed very tightly around the specific features of the Console 1 plug-in, achieving a depth of integration that isn't really possible with generic DAW controllers.

Console 1 also represented a fairly radical rethink of what we might want a hardware controller to do. Previous DAW controllers typically behaved like scaled-down sections of a large-format console, with the main feature being a bank of faders to adjust multiple channel levels at once. The Console 1 controller is always focused on one

specific channel, and doesn't even have a single fader; rather, it's designed to give you immediate hands-on control over channel-strip processes such as gating, EQ and compression.

Since its launch, Console 1 has attracted many devoted users. Its popularity has been boosted by a drastic price cut that was enabled by moving production from Sweden to China, and by the availability of add-on mixer packs that provide alternative EQ, compression and saturation algorithms modelled on other well-known large-format consoles. Should the SSL 4k sound not be your bag, you can now draw on SSL 9k, American Class A, Summit Audio Grand Channel and more. DAW developers have made setup easier, for instance by implementing automatic transfer of track names and numbers to the Console 1 plug-in, and a tie-up with Universal Audio now makes it possible to load many UAD2 plug-ins within the Console 1 framework, and to use Console 1 as a controller for the UAD Console mixer.

Console 1's popularity is well deserved, but because it makes it easier to add processing than to change the channel volume, you can end up leaning on compression where it would really be better to make some fader moves. If you use Console 1 as your sole controller, it also gets frustrating to have to return to mouse or keyboard to start and stop DAW playback. With so many faderless and transportless Console 1s out there, there's an obvious market for a companion unit that would fill in these gaps. That unit has

been a long time in the works, but finally, Console 1 Fader is here.

Fade To Grey

Console 1 Fader is a pleasing visual match for the original, and its construction gives an equally reassuring sense of solidity. In place of Console 1's numerous rotary encoders, however, the main feature here is a bank of 10 touch-sensitive, motorised 100mm faders, each accompanied by a small eight-LED meter and three buttons. My only, very minor, criticism of the Console 1 Fader's physical design is that the meter LEDs are tiny and tend to

Softube Console 1 Fader \$699

PROS

- Looks and feels good, with nice faders — and 10 rather than eight of them.
- Alternative Fader modes that operate on Console 1 plug-in parameters let you do much more than just balance levels.
- Pairs very nicely with the original Console 1, but can be used without it.

CONS

- Less useful in DAWs such as Pro Tools where it can't control the host program's faders and sends.
- No jog/shuttle wheel.
- Power supply is required and has an annoyingly short cable.

SUMMARY

Console 1 Fader has been a long time coming, but it reflects the careful thought and attention to detail that is Softube's trademark. A very well thought-out companion to the original Console 1, it's much more than just a fader controller.

Closed DAWs

You might be wondering why Console 1 Fader can't adjust mixer channel volume and pan within Pro Tools or Logic, since these applications do support generic protocols such as HUI and MCU. The answer, as I understand it, is that Console 1 Fader needs a way of synchronising actions that affect a mixer channel fader with actions that affect the Console 1 plug-in on that channel, and this isn't part of either the HUI or the MCU spec. So, although there's nothing in principle to stop Fader's faders from fading faders in the Pro Tools mixer, it would be impossible to ensure that its plug-in-specific controls were addressing the same channels.

» become invisible unless you're looking from directly above. The faders have a good, consistent feel and move smoothly and precisely, and providing 10 rather than the usual eight is a masterstroke, ensuring Fader will feel much more natural to anyone not brought up counting in binary. To their right is a compact but not crowded collection of 23 further buttons, along with a single rotary encoder.

The labels on all these controls give a pretty good indication of the basic functionality that's on offer. Under Fader Mode you'll see that the faders can be switched to control not only channel volume, but several other parameters within the plug-in too: Input Gain, low and high-pass filter cutoff, and the level and Character of the Drive algorithm. There are also buttons labelled Send 1, 2 and 3, of which more presently.

The button at the base of each fader acts like the Select Track buttons on Console 1, bringing the selected channel into focus; it's also possible to set it up so that touching a fader automatically selects that channel. One of the key differences compared with Console 1, however, is that there are only a few Console 1 Fader actions that are specific to the focus channel. Of these, the main one is pan position, which is adjusted using the rotary. This also does service for scrolling through menus when needed, and in addition, has a neat secondary function on stereo channels: hold down the Shift button, and clockwise adjustment from the centre position increases the stereo width, while anticlockwise adjustment progressively raises a frequency below which the signal is collapsed to mono. The two upper buttons adjacent to each fader also have shifted functions, Solo doubling as a polarity switch and Mute as Bypass.

The faders can be banked up and down in groups of 10 using dedicated buttons, and there are two other groups of controls worthy of note. The three buttons labelled Assignable have familiar transport symbols beneath them, denoting their default roles, but they can be assigned instead to any key command within the host program. Finally, there are two more buttons dedicated to Fader 1's Layer Mode, which implements a kind of VCA fader setup (see box).

It's probably worth noting at this point that there are also deliberate omissions from the Fader 1's feature set that might bother some people. There is, for example, no jog/shuttle wheel, no master fader, no buttons to arm tracks for recording or automation, and no displays that could show channel names on the unit itself. It is, however, possible to combine multiple Fader units in one setup.

Faders Up

The first thing you need to do on receiving your shiny new Fader is to redeem the licence code in order to download the software. Softube licences can be managed using the familiar iLok License Manager utility, but a dedicated Softube Central program seems to be the preferred option these days. The process was relatively painless on my Mac, except that for some reason the manual for the Fader was not installed and I had to find it on Softube's website.

One key difference between Console 1 and Console 1 Fader is apparent out of the box. The original unit is bus-powered; the Fader also connects by USB, but requires a 'wall-wart' PSU too. It's hardly surprising that the Fader needs more power, given that there are 10 fader motors to drive, but I could wish that the PSU had a longer cable. And since many Fader buyers will be existing Console 1 owners who will want to place the two units together, it would have been great if the Fader could have included a built-in USB hub. As it is, you're faced with the need to find separate USB ports for each unit, plus a third if you want to use an iLok or eLicenser dongle, a fourth if you're running a USB audio interface, a fifth for a MIDI keyboard and so on.

In order to drive the Fader's fader motors, an external power supply needs to be permanently connected alongside the USB cable.


A conventional plug-in window can't easily be made to follow Console 1, so visual feedback is handled by an application called OSD. This runs in the background, and when you touch or move a control, the OSD window appears as an overlay in front of your DAW (and, indeed, any other applications that are running). OSD can be cycled through a number of different display options which trade off completeness against compactness. It automatically senses when one or both of the two controllers is connected, and the display options are tailored appropriately.

Fader Modes

As you'd expect, the default mode of operation is Volume mode. What this actually does depends on the level of integration offered by your DAW. In Studio One, Reaper and Cubase, adjusting a fader in Volume mode moves the corresponding fader within the DAW mixer. In Pro Tools and Logic, what gets adjusted is the output trim in the Console 1 plug-in. The end result is the same, but only if you obey the fairly major restriction of not placing other plug-ins in insert slots after Console 1. More of an issue is pan, especially in Pro Tools. If you use the mono-to-mono version of the Console 1 plug-in, which is the obvious choice on mono tracks, attempting to adjust pan from Fader won't do anything, so you need to use the mono-to-stereo version on every mono channel instead.

One feature only available in DAWs that offer mixer integration is control over send levels. The three uppermost buttons in the Fader Mode section 'flip' the first three DAW sends onto the Fader's faders, so you can for example quickly set up a cue mix or work through your mix applying a global reverb in different amounts to each track. For some reason these Send buttons are momentary, so you have to hold them down in order to retain the focus on sends; I think I'd have at least »





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Layer Mode

Many digital consoles have more channels than faders, and so offer 'layers' that allow the faders to be reassigned to different channels or functions. Often it's possible for the user to create custom fader layers, collecting together all the controls they particularly need on one bank of faders. Another feature often found on digital consoles, and in some DAWs, is VCA faders, which allows control of a user-defined group of channels to be slaved to a single fader. And despite its name, Console 1 Fader's Layer Mode is actually, as far as I can see, an implementation of VCA faders.

When you enter Layer Mode, the Fader faders cease to control individual channels or Console 1 plug-ins within the DAW. Instead, each fader now governs a group of channels known as a Layer. Pressing a fader's Select button while holding down the Assign button creates a new Layer and assigns it to that fader. You then add channels to the Layer by pressing their select buttons, before pressing OK to confirm. Layer assignments and names can be edited after the fact.

The fader controlling the Layer doesn't necessarily have to be on a channel that's part of the Layer, and indeed isn't by default. At the same time, however, the first track that you add to the Layer is designated Master, and gives the Layer its auto-created name. When you switch to Layer Mode, it's the Master channel's volume and other parameters that are reflected on the Layer fader.

Layers are purely a Console 1 Fader thing: the host DAW knows nothing about them, and experiences only their consequences. Move a Fader fader in Layer Mode, and the DAW channels within that Layer will follow, but the Layer faders themselves are invisible to the

DAW. Thus, for example, if you assign a Layer to fader 1, but don't include channel 1 in that Layer, moving fader 1 in Layer Mode won't affect channel 1's fader on screen. Likewise, you can move a Layer fader to automate a fade-out across multiple channels, but to do so, you need to put all of the channels within that Layer into a write mode; it's not possible to automate the Layer fader *qua* Layer fader.

One limitation of Fader Mode is that non-contiguous Layer faders cannot be condensed to adjacent Console 1 Fader faders. Suppose, for example, your drums occupy channels 1-12 and your guitars take up channels 13-16. If you assign a drum Layer to fader 1 and a guitar Layer to fader 13, you'll have no way of accessing both Layer faders simultaneously, because they'll be in different banks.

A better approach is to assign your drum Layer to fader 1, your guitar Layer to fader 2, and so on, and learn to think of those faders as doing double duty. The only problem with this is that if Console 1 Fader is addressing, say, channels 91-100 in your mix, you'll still need to bank all the way back to the first page to find your Layer faders. It would be great to have the option for Layer Mode to automatically bank to faders 1-10 when engaged. It would be even better to have Layer Mode simply ignore any faders that don't have a Layer assigned, so that it always displays the first 10 Layers on faders 1-10, with the original fader number only determining the order in which they appear.



I don't typically use VCA faders much in a standard mixdown context, and compared with some implementations, Layer Mode is quite basic, with no advanced features such as VCA spill. The fact that Layers exist outside the host DAW also means that things can get confusing when automation is involved. However, they do have one very useful application in those DAW programs that don't allow you to freely order different track types in the mixer. In a large mix, you'd typically have a number of subgroup auxiliary channels for drums, guitars, vocals and so on. Some DAWs insist on placing these last in the track list, so that they appear on the right-hand side of the mixer. If this isn't what you want, it can be valuable to create a Layer for each subgroup fader and assign these Layers to the first few Fader 1 faders. Layer Mode could also be handy for cue mixing, where you need to quickly rebalance a mix in response to musicians' requests for more cowbell.

» liked the option to make them latching, especially as it's possible to use the Sends modes to write automation data. I'd have liked access to more than three sends per channel, too, perhaps as a shifted function. In Studio One, it's also quite easy to end up in a situation where Send 1 on the Fader is not the topmost send on a mixer channel, and this has the potential to confuse.

What no generic controller can match are the other Fader modes, which adjust parameters within the Console 1 plug-in. (Consequently, you'll need to enable automation within the Console 1 plug-in as well as in the DAW mixer channel if you want to automate any of these moves.) A core feature of the plug-in is its built-in Drive, a soft-saturation algorithm that aims to emulate the effect you'd get by pushing your chosen console model hard. Also available as a stand-alone plug-in called Harmonics, this sounds great, and to really get a sense of why Console 1 is

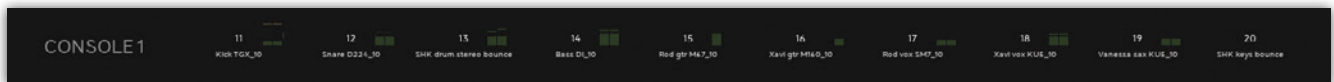
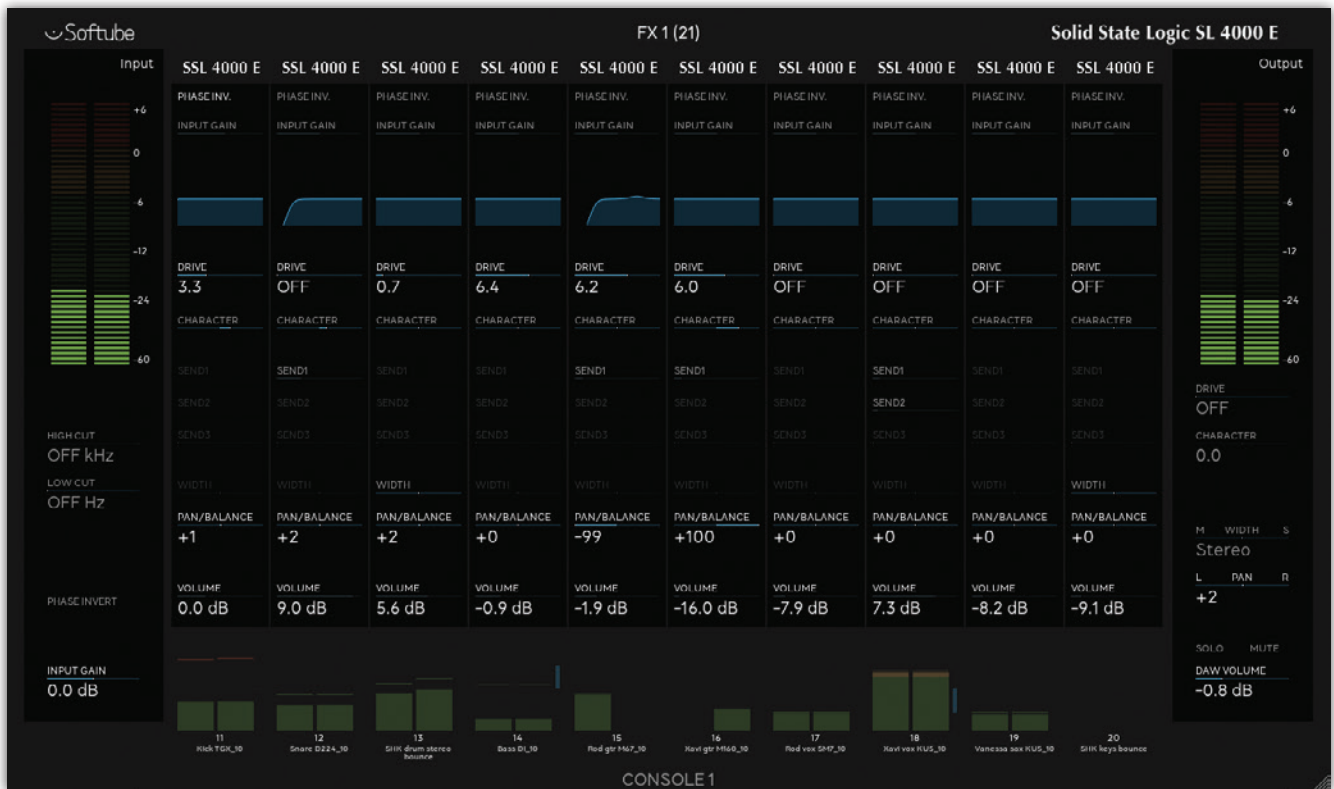
more than just a controller, it's worth diving in and applying it liberally to anything and everything. Drive and Character modes in Console 1 Fader make it ludicrously easy to do exactly that. Both are latching rather than momentary, and repurpose the faders to represent the corresponding parameter within the plug-in. Drive adjusts the amount of saturation, while Character alters the spectral balance of the added harmonics. With Character in the middle position, Drive thickens the sound without noticeably changing its tonality; positive Character values make it progressively brighter, and negative values less so.

The last three Fader modes are all momentary. Input Gain adjusts the level entering the Console 1 plug-in, while in Low Cut and High Cut modes, the faders represent the cutoff points of the channel filters. This might sound boring or utilitarian, but in practice, it's amazingly useful — think about the time it

normally takes you to set a high-pass filter appropriately on every track in a DAW mix, and divide it by about five. In fact, just as Console 1 makes it possible to get close to a finished-sounding mix using only EQ and compression, so too does Console 1 Fader allow you to achieve a surprising amount with only volume, Low Cut, Drive and Character, without invoking conventional EQ at all. Use them together and the world is your oyster!

Fading Fast

Products that deliver a genuine surprise don't come along very often, and the original Console 1 was one. Before Console 1, I'd never been unhappy mixing from a mouse and keyboard, and had found hardware controllers to be more trouble than they were worth. Console 1 made instant sense. It felt complete, mature and intuitive right from the outset. It was rewarding to use, often made mixing faster,



Visual interaction is handled not using the Console 1 plug-in window or any displays on the Fader itself, but through the OSD screen that appears as an overlay in front of your DAW. With just Console 1 Fader attached, there are three display options; the largest of these shows every channel parameter that can be adjusted from Fader, whilst the other two behave more like a simple meterbridge.

and, most surprisingly of all, often led me to make different decisions than I would have done without it.

Would Console 1 Fader have had the same effect, had it arrived first? It's hard to be sure. It too has many unique features that aren't available on any other fader surface, most notably the alternative Fader modes; but at heart it's still a fader surface, and thus not quite the bolt from the blue that Console 1 was. And whereas every aspect of Console 1 proved useful in practice, I'll admit there are one or two Fader features that haven't yet chimed with me personally. However, that's not to say they don't work well, and I can imagine that others might love Layer Mode, or the facility to artificially increase stereo width on every channel.

The Console 1 ecosystem is something you can't really dabble in. You have to throw yourself into it completely, and while you're making EQ and compression tweaks from the Console 1 surface, the DAW itself becomes little more than a tape machine. Making volume changes from Console 1 Fader brings the DAW's mixer back into the equation, and its value is somewhat dependent on the DAW manufacturer providing appropriate 'hooks' into their program. At present, Console 1 Fader is thus more appealing to those who use Cubase, Reaper or Studio One than it is to Logic or Pro Tools users.

Conceivably, it might even tempt some people to switch, because in the right host, Console 1 Fader does seem like the missing piece of the puzzle. It supplies almost all

the functionality that I miss on Console 1 — the main exception being the ability to use the rotary controller to move the DAW playhead — and, as we've come to expect from Softube, adds a huge amount of extra functionality I hadn't realised I was missing! Once you get used to adjusting high-pass filter cutoff or Drive from faders, other ways of doing it just feel clumsy.

If you only ever used Console 1 Fader, I don't think you'd get the full picture of what makes the system so good, so I'd still recommend the original Console 1 as the better place to start. But once you've experienced the Console 1 system, the appeal of Fader will be obvious, and I don't think it will disappoint. **///**

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Behringer RD-8

Analogue Drum Machine

Behringer's RD-8 is a cut-price recreation of Roland's classic drum machine. We put it to the test...



Behringer RD-8 **\$349**

PROS

- Big, convincing 808 sound.
- Individual outputs.
- Built-in effects.
- Performance effects.
- Channel mutes and solos.

CONS

- Many operations confusing.
- No simple backup or companion software.
- Some good 808 sequencer features are gone.

SUMMARY

Sonically, the RD-8 is as good as owning an 808, and even has some sound mods and many extra functions. The modern parts of the interface have room for improvement, but at least they're not as confusing as the original!



SIMON SHERBOURNE

I have to be honest, although I knew there was a buzz around Behringer's 808 reboot I hadn't paid it much attention before one turned up at my front door. Don't get me wrong, the 808 is everything: hearing Cybotron's 'Clear' on the electro mix tapes that circulated at school was like stumbling into a portal to a new world.

A generation or more on, the dominant musical styles of our time still build on a foundation of 808 kicks, snares and hats. I just assumed that today's producers would be indifferent to where those sounds were coming from, or would be happy with the convenience of Roland's digital recreations. I was wrong; how many products generate a 150+ page GearsLutz thread *before* they ship?

The RD-8 is a true analogue recreation of Roland's classic drum machine. As such

it's not really stepping on the Japanese giant's toes. Roland have their miniature digital reissues, and modern drum machines like the TR-8S, which model those iconic sounds. What's more, several other analogue 808 clones are extant.

R-Type

I can't pretend I didn't feel a bit of a thrill of setting a drum machine with the actual physical presence of an 808 on the desk. It's big, but for an instrument that you can actually perform with (and connect a nest of quarter-inch jacks to) it feels about right. The knobs, step buttons and font give it that unmistakable Roland look, though Behringer have playfully inverted the colours.

The build is reasonably solid, and the step buttons are satisfyingly clicky. The only things that feel a bit plasticky are the knobs. Given that this is an 808 tribute costing a third of comparable alternatives »



Get Yourself Connected

High up the list of reasons to get an RD-8 is its complete set of multiple outputs. Every drum channel has a discrete quarter-inch out, which automatically breaks routing to the main mono out when connected. In case you were wondering, there's no audio-over-USB like the TR-08. Again, remember £300.

A cool bonus is a return channel. This lets you connect out individual channels to external

effects and bring them back into the mix at the final output stage.

As well as audio I/O, there are comprehensive sync and external trigger options. Sync mode is set via a dedicated front panel section. The RD-8 can run as both master and slave over regular MIDI, USB MIDI, old-school Roland sync or modular clock.

The back panel also sports the full 808

complement of three trigger outputs (two more than the TR-08). The trigger outputs are derived from the Clap, Cowbell and Accent tracks, the same as on the 808. However, they're not labelled as they are on the original, and this information doesn't appear in the manual. The point, though, is that they work, and they add a lot of potential and fun factor if you have instruments that can respond to them.

» (and a tenth of an original) I'm not complaining. I will grumble, though, that Behringer have made the knobs quite stiff to turn. Perhaps this is an attempt to make them feel a bit more luxurious, but when performing I want to be able to crank the knobs quickly, not fight them.

The real surprise was how much of the panel is different from the 808. Alongside the step switches and the knobs is a host of backlit rubber buttons of the kind you'll find on many modern music tech gadgets. The original's sound selector dial is replaced by dedicated channel select pads. The sequencer mode dial is also gone. This was the point at which the penny dropped for me. Behringer haven't set out to make a straight clone, they've made a hybrid of old and new: the original analogue drum synths with a reimagined performance and sequencing system.

Having got my bearings I decided to jump in and have a listen. Hitting the Play button brought no result: there were no preset patterns in memory. No problem, let's call that refreshing and move on. Manually triggering sounds works just like on the original. Whichever sound is selected can be played via the single Trigger pad.

It's a shame that Behringer didn't make it possible to finger drum on the channel selector pads. However, the RD-8 can be played live from external MIDI sources. Notes hit hard (over velocity 110) are treated as Accented triggers, and you can record directly into the sequencer,

something you can't do on the Boutique TR-08. Anyway, before we get into all that, how about those sounds?

Selection Sixteen

The RD-8 has the same set of 11 channels as its inspiration, with a total of 16 drum synth models. The sounds are close recreations of the originals, and some have extra controls.

First up is that all important bass drum channel, which has the classic Tone and Decay controls but steals some panel space from the neighbouring Accent section to add a Tuning knob. Tuning covers about an octave of range, with the original pitch at around the 10 o'clock position. (Eagle-eyed readers may have spotted orange dots by all the new controls. These indicate the settings that dial in the original sounds.) The decay can go longer than the original, so this is more like an 808 with the celebrated decay mod.

The kick really does the trick, and is especially magnificent pitched down. It's smooth and punches the air with that elusive rumble even on smaller speakers and headphones. A couple of notes: first, it can go super loud compared to the other channels — maybe that's by design. Second, if you play triggers in quick succession, occasionally one will get kind of clipped, losing its decay — something the original was also prone to. I think it would have been awesome if Behringer had provided MIDI note control over the Tuning, giving you a real analogue 808 bass.

To my ears the snare is indistinguishable from that of a pristine 808. The snap is as bright and scratchy as you could need, and the body portion of the sound has that authentic tubbiness. I just wish they'd added a decay control: short 808 snares are in vogue, and this is something the TR-08 can serve up.

The three toms have that lovely dusty low end; warm and true to the original. The congas however are more of a straight pure tone than on the 808, with a less pronounced pitch envelope. I much prefer the RD-8 version! While scrutinising the Tom/Conga channels I noticed they emit a low level resonant hum at all times. This only tends to be noticeable while adjusting the tuning.

On the Rim/Clave channel, the rim shot wins the authenticity battle with its brilliant clear crack. The claves are quiet by comparison, and are slightly shorter and lower in pitch than on a typical 808. They are closer to a straight click/stick sound than the original's characteristic 'pwee'.

Behringer have modded the clap patch with an Offset control. This adjusts the sound balance between the bright, flammed noise portion and the softer reverberant tail. It's a useful addition which makes the clap more versatile, and is fun to tweak in builds and fills.

The RD-8's take on the world's most famous cowbell is unmistakably an 808 cowbell, but is different if you A/B old and new. The RD-8's is noticeably lower in pitch, and its constituent tones are

»

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» more detuned. It's the same story with the cymbal: lower in pitch, with more dissonant harmonics. They might not satisfy purists but I rather enjoyed the wonkiness of them.

The hats show that analogue warmth does not mean you have to lose high end. They have plenty of sharpness if you want it (you can filter with the additional tone control) and have a certain impact and presence. They are not perfect copies of the original, but they wipe the floor with hats on the Roland Cloud 808 plug-in.

First Steps

The 808's 16-step trigger sequencing concept is a design classic, but most aspects of its operation are notoriously tricky to understand. As Behringer have overhauled the sequencing side of the 808, presumably aiming for something more intuitive, I was curious how far I could get without resorting to the manual. Not very far at all, it turns out.

When I hit Play there was no sign of life, and pressing step buttons didn't seem to do anything. OK, got it, I was in Pattern mode where the buttons recall patterns. I needed to switch to Edit mode, the equivalent of dialling from Manual Play to Pattern Write on the 808. Now I got running lights, and could enter trigger steps. But... they disappeared after one bar! Time to take the walk of shame to the manual. Ah, Record needs to be armed to enter notes, otherwise they are just one-off trigs.

After this bumpy start things went more smoothly. Triggers can be entered on the step buttons one sound at a time as expected. The Accent track adds steps of higher level and intensity. The Pattern can be extended up to four bars (64 steps), and you can bank your view between the bars via dedicated buttons, or enable Auto Scroll. One thing that kept throwing me off when entering patterns is that the graphical section outlines above the steps are divided differently to the original. The 808 has a break that serves as a visual cue for where step 9 is. On the RD-8 a similar break is one step later.

Patterns can be tapped into the RD-8 in real time via the Trigger pad, albeit one sound at a time. This is my preferred method on both this and the original. Everything is quantised and, unlike on the TR-08, there's no facility to push notes off the grid into substeps. You do get Swing, though, which was absent on the 808.

Figuring out how to save my work required another trip to the manual. You need to hit Save, then choose a slot to write to, then hit Save again. Despite following the manual I couldn't get it save to any other slot than the current one. To drop your pattern into a different slot you can Copy, which again is a rather obtuse process.

Next Gen Sequencing?

The memory structure is far from clear but I kind of figured it out. Patterns are stored in groups of 16 in 'Songs', which are also, in a sense, banks. Patterns can be triggered manually within the current Song. (You can't access Patterns stored in a different Song without stopping playback). There's a (confusing) process for making an actual song within the Song (still following?). You essentially create chains/playlists of Patterns, which will play back in sequence in Song mode if Auto-Scroll is on in both Pattern and Song modes! I think.

A couple of other places I got stuck: you can create patterns in several slots within a Song, but they are not stored until you deliberately save them. Forget this and change Song (or power down) and all is gone. Also, Song, Pattern and Step modes all have their own Record modes. If Record is active in Step and you change to Pattern, Record remains armed but you can't see it. The Trigger button (or incoming MIDI) will continue to write into the active pattern. This is cool if you know about it but it confused the hell out of me at first.

Despite the hiccups, the Pattern creation process is of course simpler to pick up than on the 808. While you can make a 64-step pattern on an 808 it requires learning (usually re-learning) how to write a two-part pattern with A and B variations. The thing is, though, while Behringer have made a modern sequencer, it's also quite confusing to use, and throws the baby out with the bath water. Yes, the 808 was a nightmare to learn, but features like the A/B variations were useful. The biggest loss is real-time song arrangement. The 808/TR-08 have a song write mode where you simply perform a series of pattern changes and Fills and the sequence will be captured and stored. I would much rather do this than assemble a chain offline.

Fills are handled in an 808-ish way: they're created as separate patterns in slots 13-16 in each Song. They can be any

Fast Dump

The RD-8 can store 16 Songs with 16 Patterns, but sooner or later you're going to need to get your creations in or out of the box. There's no software utility for this, instead the RD-8 relies on good old-fashioned SysEx dumps. There's not really any useful information for the typical user about how to do this, other than a mention of the Dump button on the front panel. Instead the manual devotes pages to explaining the data model and MIDI messaging used to encode the unit's settings, songs and patterns. I can only think that Behringer have dropped this information here hoping that an independent developer will use it to create a backup utility.

length but work best if you keep them short, say one or two beats. Hitting the fill button slots the fill in at the end of the next bar, and automatically fits it in to end at the right point. The 808's Auto Fill mode, which automatically drops fills in at the end of a set amount of bars, is unfortunately not present.

Positive Reinforcement

Let's turn to some of what's new and fun. What I love are the new performance features. The 808's format already made it good for live tweaking, but adding things even as simple as Mute and Solo make a big difference to live improvisation on the RD-8.

Front and centre on the panel you get the Note/Step Repeat section. These functions retrigger/loop an individual sound or pattern step respectively, with a choice of four time divisions. I particularly like the Step Repeat. It's a fairly straightforward stutter/roll effect compared with, say, the Scatter effects on Roland's modern drum machines, but it's exactly what's needed to keep things alive with quick fills or more dramatic builds.

As well as the live Note Repeat function, you can tie one, two, four or eight(!) retriggers to individual steps as part of the pattern. What you can't do, here or on the live Repeat, is add dotted notes for those classic trap-style dragging hats.

RD-8 sequences can store a lot more than just pattern data. Nearly all settings, including tempo, swing, filter mode, and even mutes and solos can be stored per Pattern, per Song, or Globally. You can choose this preference for each parameter type separately, so you might want to store the tempo for a whole song, »

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SOUND ON SOUND

~ Full 3 Page Review (August 2019 Edition)

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Photo: Don Was (Producer) & Krish Sharma (Rolling Stones' Recording Engineer)

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The RD-8 enjoys the spacious layout of the 808 and measures 498 x 265 x 77mm. You might need a bigger desk.

» but have mutes or filter mode stored per-pattern.

Taking a leaf from the Elektron book of sequencing, the RD-8 features Probability: the option to introduce the element of chance into whether active steps trigger or not. The main Data encoder has a master mode for Probability, which adjusts the percentage chance of all steps enabled for probability. The same system works for Flams.

Now, the way that both Probability and Flam have been implemented is the very definition of the word 'assbackwards'. By default, Probability (and Flam) is on for every step on all tracks. So as you dial in Probability or Flam, these effects will start to apply to every note across all channels. To be more selective about which steps are affected you can go into

Alternatives

Other analogue recreations of the 808 are available, such as the Eurorack-sized **System 80 880**, the **Acidlab Miami** (reviewed SOS Nov 2014), and the **E-licktronic Yocto**. However, all are around three times as expensive, although you can get a DIY kit version of the Yocto for RD-8 money.

Arturia's DrumBrute Impact is an original take on a classic analogue drum machine, with a better modern sequencer than the RD-8.

Roland's own TR-08 is of course digital, but still sound great. It's compact, battery powered and has multitrack USB audio. It also keeps the 808's real-time song write mode. In most respects the RD-8's sequencer is more accessible, and has cool new tricks like Step Repeat and per-channel pattern length. Its size and all the outputs make it much more like an instrument than a gadget.

a Settings mode and deselect the steps that you want to mask out. So, if you want to introduce a little randomness to a few snare hits in your pattern, you will need to painstakingly press every step button, on every bar, one channel at a time until only those few snares remain lit! Given that we only have a few short years to enjoy our families before the North Sea takes Kettering, this seems a tad inefficient.

Effects

Still on fun new stuff, two effects have been added to the RD-8 master bus. Both of which are analogue, thus avoiding any potential griping about sully the pure analogueness of the signal chain. Behringer's Manchester crew sure seem to know their market. The effects sit in series on an optional route to the master bus: they don't affect the individual outs or (somewhat frustratingly) the headphone output, and you can send signals via the effects on a channel-by-channel basis.

Signals on the effects loop first go through the Wave Designer, which is somewhere between a transient designer and compressor. It can add punch to transients, make the mix pump and generally add overall oomph. Then you have a resonant filter, which can switch between low- and high-pass modes. A surprising bonus is that you can automate the filter cutoff within each pattern. This can be recorded in real time, or adjusted per step in Settings mode.

It's a nice touch that you can send channels individually to the effects, but the

routing scheme means that sounds have to go through the Wave Designer to get to the filter. I'd actually prefer it if the Wave Designer was on the optional bus and the filter was just on the mix all the time, ready for quick transition effects.

The Verdict

If you've ever lusted after an analogue 808 then the RD-8 will definitely put a smile on your face. Surely only serious collectors would now opt to pay more than 10 times as much for the original? Sonically, it's spot on where it really matters, and where it differs I generally preferred it. The chunky form factor and comprehensive I/O make it ideal as a performance instrument, not just a studio gadget.

It was of course the right move to add a modern, more accessible sequencer, but I think Behringer have fallen short of the mark here. The structure and operation are confusing, some things like the Probability system are nuts, and I wish they'd stayed true to the 808's real-time song write system. As this sequencer will presumably form the basis for other products like the RD-9, it might be worth them taking another look. Even so, these frustrations wouldn't stop me using the RD-8 for its new performance features like mute/solo, Step Repeat, and the automatable filter. In summary, it sounds great and at this price is bound to prove irresistible. **///**

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Quantica Modula

Modular Acustica Plug-in Controller

Rumoured for months, the 'powered by Acustica' virtual mixer is now close to release — and we were treated to an exclusive test drive!

MATT HOUGHTON

We don't often 'preview' a product rather than review it upon its release, but if anything warrants an early look I feel Modula does. Soon to become available under the 'Quantica: Powered by Acustica Audio' name, there have been rumours of its arrival for over a year but, with little to go on other than tantalising glimpses

on Acustica's Instagram feed, it's not always been obvious exactly what it will do! My interest piqued, I persuaded the developers to grant *SOS* exclusive access to a late prototype, a month or two before the product's scheduled release. And having now used Modula intensively for a couple of weeks, I feel it's already worth shining the *SOS* spotlight on it.

First, the branding warrants a quick explanation. Quantica is a company

formed to house this joint venture between mixing and mastering house Studio DMI (<https://studiodmi.com>) and plug-in developers Acustica Audio. The pair have worked together previously to develop Acustica's impressive Diamond Color EQ (first version reviewed in *SOS* July 2017: <https://sosm.ag/AcusticaDCEQ>), Diamond Lift EQ and El Rey compressor. All Acustica's plug-ins are high-quality emulations of real analogue gear, created using a very



When first launched, Modula appears as an empty mixer. To populate the channels, you must load Modula 'Skin' plug-ins in your DAW.

Quantica Modula \$999

PROS

- Touch-friendly mixer for the most convincing analogue emulations around!
- Channel-strip approach makes lots of controls accessible simultaneously.
- A couple of great channel strips are included in the price.
- Drag-and-drop reordering of channels.
- A huge list of new features in the pipeline.

CONS

- Some may prefer to wait for new features.

SUMMARY

A touchscreen-capable host for Acustica's excellent plug-ins with some truly wonderful processors included, Modula could well bring some of the old analogue joy back into mixing and mastering in the box.

different technique from that used by most developers. We've described their approach many times so I won't go into detail here, but as a basic understanding is required to 'get' why Modula might be a big deal, here's a quick overview. Essentially, instead of measuring analogue gear and modelling its behaviour in code, Acustica capture thousands of impulse responses and use a proprietary engine based on their 'Vectorial Volterra Kernels Technology' to determine which impulse(s) will be used to process a signal at any given time. You can learn more about it at www.acustica-audio.com/pages/engine if you want but the key 'take-away' is that while it's a relatively CPU-intensive approach, it's capable of remarkably accurate emulations.

What Is Modula?

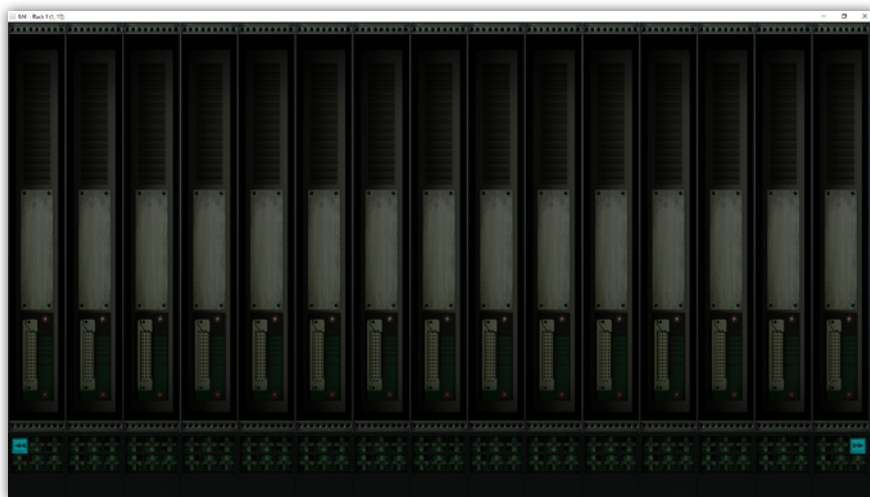
Essentially, Modula is a touchscreen-friendly virtual mixer whose channel strips are Acustica plug-ins with a custom GUI. But the bigger picture is more nuanced. Modula

the effort to develop a suitable workflow, Modula already delivers the intended benefits. How else would you fit 15 separate channel-strip plug-ins on-screen at a size that makes them all usable? Even if you could accommodate that many instances, you'd be very hard-pressed to arrange the GUIs in a way that allowed space for your fingers to operate a touchscreen.

In some respects, then, Modula is competing with other channel-strip products like Harrison Mixbus, say, or Softube's Console 1 controller. And Modula's touchscreen friendliness might put Slate's Raven or Devil Technologies' DTouch in mind. But really, it's very different from any of those — and, of course, none of them have been designed to work with Acustica's unique plug-ins.

OK Computer

Modula's 'mixer' can scroll in banks of 15 channel strips up to a number that would exceed the capability of pretty much any



was conceived as a companion to your DAW rather than a replacement for it, so it doesn't do any actual mixing — the channel routing and summing is left to your DAW — and it doesn't (yet?) offer any DAW control facilities. Rather, it acts as a convenient control surface for Modula-compatible plug-ins that are hosted in your DAW.

The idea is that by laying out many channel strips of Acustica processing, with every strip offering a choice of preamp, EQ and dynamics emulations, you can enjoy both a convincingly 'analogue' sound and a speedy, intuitive analogue-style workflow. I don't think the developers would mind me saying that there's still plenty of potential for further improvement — they already have a long list of features they plan to add over time — but I believe that if you make

current computer — if there is an upper limit, I never reached it. This brings us to the question of computing horsepower, because although the Acustica engine is frequently updated, and each refresh brings efficiency gains, these plug-ins still consume more CPU cycles than equivalent modelling plug-ins.

Until recently, my studio machine was a nine-year-old Windows 10 computer based on what was once a high-spec processor — a first-generation Intel Core i7 965, with four 3.2GHz cores (eight threads). Like my 2018 MacBook Pro (six-core Core i7, 2.6GHz), this could run enough Acustica plug-ins to be useful in mixing or mastering but nowhere near enough to support, say, a 48-channel Modula console. Indeed, for mixing with Acqua plug-ins on these machines, I've used >>



Some of the Modula plug-ins as they appear in your DAW, with all three panels visible simultaneously. The two on the left, Magenta 5 and Diamond Color EQ, are included with Modula. The others, Water and Taupe, are prototypes supplied for evaluation — they'll be paid plug-ins, but won't be publicly available until after Modula is released.

» my DAW's freeze/bounce facilities often, reserving 'live' Acqua processing for group busses, the stereo bus, and major elements such as the lead vocal. The quality of the Acqua plug-ins justified that approach.

But computing power has moved on. Recently, I invested in a PC with a Core i9 9900K eight-core (16-thread) processor, built and overclocked to 4.9GHz by UK-based audio-PC experts Scan. It's a great machine but not expensive as professional machines go — you can buy more powerful models. This PC doesn't break a sweat running Modula with many channel strips: I gave up counting when I got to around 45 instances of Magenta 5 (see 'Pricing & Bundled Plug-ins' box) with preamp, EQ and compression all engaged. Running with Cockos Reaper, Windows 10 Home, and an Audient iD4 interface with a 256-sample buffer, Reaper's performance meters told me this was using about 48 percent of the CPU. So running something like a 24-channel mixer should be well within the capabilities of a more modest machine, particularly if you were to increase the buffer. But if computing power *is* a concern (perhaps you're running other thirsty plug-ins too), it's worth noting that Quantica

have plans to implement a server version in the near future, rather like Acustica did with Nebula 3, and I'm told that will allow you to run Modula processing on a networked computer, taking the strain off your DAW.

Strip It Good

Modula is a standalone application, which you must boot before firing up your DAW project. When I ignored the instructions and booted in the incorrect sequence (I had no manual to hand, but I'm assured one is being written) I observed some glitchy behaviour of controls and meters. When I got the sequence right, these problems were absent. To close Modula, you must first close your DAW session: Modula will unload its channel strips, and you can then close it from the system tray. Modula's settings, including the channel order, are remembered when you next load the same DAW project.

The application initially presents as an empty virtual mixer frame, in the gorgeous-looking skeuomorphic style I've come to expect from Acustica; you even see the 'sockets' where the virtual strips will be 'plugged in'. At the bottom, each slot has the same set of controls. There are mute and solo buttons, a mute/solo group function (solo a channel and only others in its group will be muted), and a channel label that's empty by default. Once the mixer is populated with channel strips, three buttons allow navigation of each strip's three sections (the leftmost

and rightmost channels in the current bank have a two-button scroll/tab facility, with the third button becoming a bank-scroll facility). And you can grab and move a channel to reorder the mixer.

With Modula running, you load your DAW project and instantiate your chosen Modula 'Skin' (their term for a Modula plug-in) in the usual way. 64-bit AAX and VST2/3 formats are available for both Windows and Mac, as well as AU for Mac only. You can easily spot these Skins in your DAW's plug-in list by typing 'M4' in the display filter — all Skins' names include that phrase. Each Skin has two GUIs: one in your DAW, where it functions like any other plug-in, and one in the next available slot in Modula. These are sync'ed, so that tweaks made using one are reflected in the other; they control the same underlying processing.

The version in the DAW is divided into three vertical sections — Pre (preamp), EQ, and Dyn (dynamics) — and you see all three simultaneously. So it's basically an Acqua plug-in with a different GUI. The GUI in Modula has the same layout, but displays only one of these sections at once; Pre (which includes not only the preamp emulation but also the pan and fader controls) is the default view. The aforementioned three buttons at the bottom of each channel allow you to choose which section of its Skin is displayed. This neat arrangement allows a narrower GUI, and thus those 15 channels to be

RHYTHM PRISM

3-VOICE RHYTHMIC PITCH SHIFTER





Here's part of a typical large mixing setup inside Modula — note that we're up to 60 channels of Acustica processing running on one Windows PC, without any trouble! The empty channel and the three M4Chan utility plug-ins used on the project's main effects returns help to break the mixer up visually, which I found a useful navigational aid.

Another important function your DAW must take care of is transport control, as there are currently no transport or marker facilities in Modula — if Modula is in focus, your DAW isn't, so your

» accommodated on screen. It works well. My only criticism is that I could find no way in this prototype version to switch the view for all instances at once; I hope that feature will be considered, perhaps as part of a more generally beefed-up grouping facility, as it could be a real time-saver.

Touch & Go

One of the most enticing things about Modula is that the GUI has been designed for touchscreen use. You aren't *required* to use a touchscreen, but if you do it will make a whole lot more sense. A 27-inch 1920x1080 touchscreen like my Acer T272HL is perfect, making all the buttons, faders and knobs the optimum size, and putting sufficient space around each control that you aren't forever hitting the wrong thing. It also means that the faders are about the same size as typical long-throw faders on an analogue console. I wouldn't recommend using a smaller touchscreen, and while it will work on a larger one there's little to be gained. I'm told that in Acustica's studio, they have a system with five screens, three of which are touch-capable... the mind boggles!

It's intended that Modula will eventually support full *multi-touch* operation, which would allow you to adjust multiple controls or faders simultaneously, but at the time of writing this was not yet the case — I could only move one control at once, and I don't believe this will have changed by the time Modula is released. That might not please those who dream of grabbing fistfuls of faders, but I don't think most of us work like that in the real world; although I often hover my fingers over a few controls I very rarely move two simultaneously (it's one reason

I gave up on multi-fader control surfaces; they occupied more space on my desk than I felt they justified).

DAW Integration

That Modula works alongside rather than within your DAW raises questions of workflow — what will you do in your DAW, and what in Modula? At the present stage of development, with various features yet to be implemented, those questions are particularly important. For example, there are limitations to how Modula plug-ins work with your DAW's automation; they'll respond to automation but cannot write it. The write facility is planned but won't be implemented for the initial release. Why? Well, the channel strips are essentially repackaged Acqua plug-ins, and this has long been a limitation for them because Acustica's impulse-based approach to their emulations, with mostly switched rather than continuous controls, doesn't really lend itself to automation. And, in any case, most controls aren't the sort of things you're likely to want to automate; it's not like we're dealing with instruments, or out-there delay effects and filter sweeps.

That said, people will undoubtedly hope to use Modula's faders for level automation — it would allow you to do more mixing jobs, such as refining level automation in multiple passes, without leaving Modula. So even if write-automation were implemented only for the channel strip faders in the first instance, I think it would be a significant improvement. That will come in time — but meanwhile Modula doesn't prevent you doing this side of things in your DAW, and it's one of the few things that I find touchscreens are good for in most DAWs.

DAW's shortcut keys can't be used. It was never conceived as a DAW controller (each Modula channel corresponds not to a DAW channel but to an individual plug-in) but some global DAW-control facilities, particularly basic tape-style transport controls, would mean a lot less switching between Modula and your DAW. In the meantime, I'd consider using a second screen for your DAW, or using a small control surface (eg. an Avid Artist Transport or a PreSonus Faderport), or perhaps a tablet/smartphone-based MIDI controller, as a sidekick for Modula's touchscreen GUI. Alternatively, Cmd/Alt+tabbing between Modula and your DAW could become almost second-nature!

Modula Workflow

So, that's what Modula does and doesn't do, and how it relates to your DAW. But what of the experience of trying to mix with it? Despite this software being at such an early stage of its evolution, it's honestly a real pleasure. I enjoyed getting hands-on with the touchscreen in a way that I find I don't with DAWs (even those that claim to have been designed with touch in mind), and I found that the layout encouraged me to stay focused, avoid distractions and thus make decisions quickly. Apart from the obvious lack of tactile feedback, the experience is closer to that of using an analogue console than I've experienced with any other software or control surface. I can see so clearly what it is I'm tweaking, can hop from channel to channel without opening and closing windows, and I don't see all those tiny details in the DAW that can lure me down mixing 'rabbit holes'. As I mentioned earlier, I missed the ability to

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» write fader automation and often wished for a touchscreen transport control, but I still managed to establish an efficient workflow.

I had access not only to the Magenta 5 and Diamond Color EQ3 Modula Skins that will be included in the price, but also to prototypes of the wonderful Water and Taupe Skins, so I could experiment with different configurations in the mixer. For example, I could construct a mixer mostly out of Magenta (emulated Manley) channel strips, and have the project's master-bus signal flow through Diamond Color EQ3 and then into Taupe. Imagine what *that* mixer would cost in the real world, even if you could build it!

It took a little while to grow accustomed to having two separate 'strips' for the single master bus in this setup, and I found it helpful to visually separate the two master bus channels from others, either by placing an empty channel strip between the master section and other channels or by inserting the bundled M4Chan utility channel on a project's effects returns, and placing those between the master bus and other channels. Ideally I'd like the ability to 'dock' master channels on the right or left of the mixer so it can be seen irrespective of the bank that's selected — there are plans to develop some sort of master module in the future, but don't yet know if it will make this possible.

I should note a couple of GUI quirks, which will hopefully be addressed soon. Much of the gear being sampled — and certainly the bundled strips — is stereo mastering-oriented gear, and there are separate polarity invert switches for the left and right channels. That makes good sense for mastering, of course, but if working on multitrack drums it's a little fiddly having to press two buttons to check for the best phase relationships between channels; a single-button polarity option would be welcome. I also noticed that some common functions, such as the dynamics on/off switch, were in different places for different Skins, which can slow you down a touch. (I'm not comparing the final Skin versions, though, and this sort of thing is easily fixed.)

My biggest challenge was to figure out precisely when I should turn to Modula, and when to use the DAW. Typically, I'll break mix projects into a few stages. First, I'll attend to housekeeping and 'problems' (eg. edits and comps, mutes, de-essing, breaths and pops, and 'cleaning' EQ). Then I'll try to build a 'faders up' mix without any major processing, other than a touch of compression where it's obviously needed to tame things, and basic pan settings. It's an

Pricing & Bundled Plug-ins

The full price of Quantica Modula will be \$999, but there will be discounts when it's launched: the first 100 purchases will be at the special 'Pioneer' price of \$399; and further purchases in the first month after launch will cost \$699. You must own Modula to run the Modula 'Skin' plug-ins, though the following are included: Magenta 5 Red and Magenta 5 Black, which are both based on Manley gear, and Diamond Color EQ3, based on Studio DMI engineer Luca Pretolesi's mix-bus chain.

(To buy the Acqua version of both those plug-ins would cost around \$420.) There's also an M4Chan utility plug-in included, which offers basic channel controls (eg. stereo width and level fader) but no processing. Other cost-option Modula plug-ins will become available over time, and will be less expensive than the equivalent Acqua versions. Acustica couldn't confirm which will be released when, but said that owners of the recent Gold 3 Acqua will get the Modula version free.

attempt to establish a sense of which tracks need to sit where in the mix, and to provide a meaningful context for later mix decisions. From there, I'll move on to what I think of as 'real mixing', working more on the tonal shaping side of things. Typically that involves broad EQ moves, some use of compression and saturation, and sending various sources to effects. For this, I'll often work top-down from the stereo bus, through the group busses, and then to any individual sources which require further attention; I find I end up mixing quicker and with less processing this way. Finally, I'll move on to detailed level automation, often using a control surface to refine that automation over several passes.

Modula already slots neatly into the 'real mixing' stage. I use my DAW essentially to prep a mix, tackling problems and using clip and channel gains to establish my faders-up mix, and rig up some delays and reverbs as send effects. Then I can spend plenty of time in Modula, massaging the sound and static levels, for which having access to all those EQs and compressors on a touchscreen GUI is a dream come true for me; I can work through things with refreshing speed. I'll have an M4Chan utility plug-in for each effects return and for some critical channels on which I don't want to use any Acustica plug-ins. Then, I'll return to the DAW to micro-manage the details of things like parallel distortion and level automation, before returning to Modula for final tweaks to the master bus processing. (I realise mixing isn't usually quite as linear a process as I've just described, but I do generally break it down into broad phases like that; hopefully you get the idea.)

Verdict?

It's the job of a reviewer to pass judgement, but you must bear in mind that I'm evaluating a prototype here — that there's plenty yet to come, in terms both of refinements and of new features. I look forward particularly to the prospect of DAW write-automation support, and

hopefully some form of transport-control too; I suspect those two features alone could turn an already attractive product into a no-brainer for many people.

But the bottom line is that while Quantica Modula isn't yet perfect, it already works brilliantly in so many respects. It really can put the sound and workflow of a high-end analogue mixer at your fingertips. I have a feeling I'll be using it on every mix for the foreseeable future, and may very well consider adding a second touchscreen to put more channels at my fingertips...

I'd happily use Modula with the bundled Magenta and Diamond EQ strips alone. (OK, maybe Taupe too!) But Skins are planned for all Acustica's Acqua range, raising the prospect of creating 'sampled' vintage Neve, API, SSL or Harrison consoles, or crazier setups you'd never find in the real world — an 80-channel GML or Sontec console anyone? The looming possibility of using additional computers as a DSP farm for your Acustica processors is also tantalising.

In terms of value for money, it's tricky to assess at this stage. But if you want accurate software emulations of analogue gear, I reckon Acustica are out there in front at the moment — certainly they're amongst the very best, as our reviews of their plug-ins testify. They're also prolific in terms of new releases and updates, and have a track record of providing free updates as the technology evolves. After the initial discounts, the asking price might be a stretch for the average home studio, but it's a professional tool with a 'wow factor', and if you're working commercially Modula could justify the investment, particularly given that Modula Skins will be more affordable than the Acqua equivalent. Well worth checking out! **■■■**

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Barefaced Audio Reformer 112

Guitar Speaker Cabinet



Surely there is nothing new to be done with the design of the humble guitar speaker cabinet? Well, actually, it turns out there is, and I don't think I'll ever want to use an 'ordinary' cabinet again!

DAVE LOCKWOOD

Electric guitar players are a notoriously conservative bunch, with so many of the 'classic tones' seemingly thought to reside exclusively in the guitar and amp designs of the 1950s and '60s. But whilst some of that outlook may be slowly changing with the advent of ever-improving digital technologies, the design of the humble guitar speaker cabinet — still often the ultimate generator

of the actual sound — has remained largely unchanged for half a century. The performance of bass-guitar speaker cabs has been vastly improved in that time, and PA speaker cabs have been reinvented beyond recognition, but the design of guitar cabs, with a couple of honourable exceptions, have remained a choice simply between open- or closed-back.

Experienced guitarists have an innate understanding of what each configuration offers, and often a preference for one or

the other, but neither could be said to be really optimised for the task at hand, both having arisen out of pure pragmatism. The open-back configuration of all classic combo designs was adopted primarily to offer ventilation for the integrated amp chassis and easy access to the tubes for maintenance. But the sound coming out of the back of an open-backed combo is 180 degrees out of phase with the sound coming out of the front of the cab, and whilst the short wavelengths of mid and high frequencies will reflect around the room a bit and combine usefully with the sound from the front, longer wavelengths will simply wrap around the cabinet and cancel out the frontal output. So, you can have a degree of dispersion but at the expense of losing a significant amount of bass.

As amps became larger and louder with more output tubes, they needed to be housed in separate enclosures, and designers took the opportunity to move to closed-back cabinets to improve the low-end response. Handling the power of the bigger amps, however, also required the cone drivers available at the time to be used in greater multiples, and thus the classic closed-back 4x12 was born. But now we have a different set of problems: the sound from the back of the speakers in a closed-back cab has nowhere to go other than to vibrate the cabinet and, to a lesser degree, interfere with the movement of the speaker cones themselves, thereby

»

Barefaced Audio Reformer 112 \$542

PROS

- More diffuse, bigger sound.
- Greater efficiency.
- Very light weight.
- Twin combi quarter-inch/Speakon sockets.

CONS

- None.

SUMMARY

A lightweight guitar speaker cab, featuring the innovative design of the Augmented Vent Diffraction to make it sound bigger, wider and more balanced than conventional designs of open- or closed-back cabinets.



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» colouring the sound. Perhaps just as significantly, however, the acoustic coupling of the multiple drivers in the cabinet results in an ever-narrowing dispersion at higher frequencies, especially the ones we are interested in as guitar players.

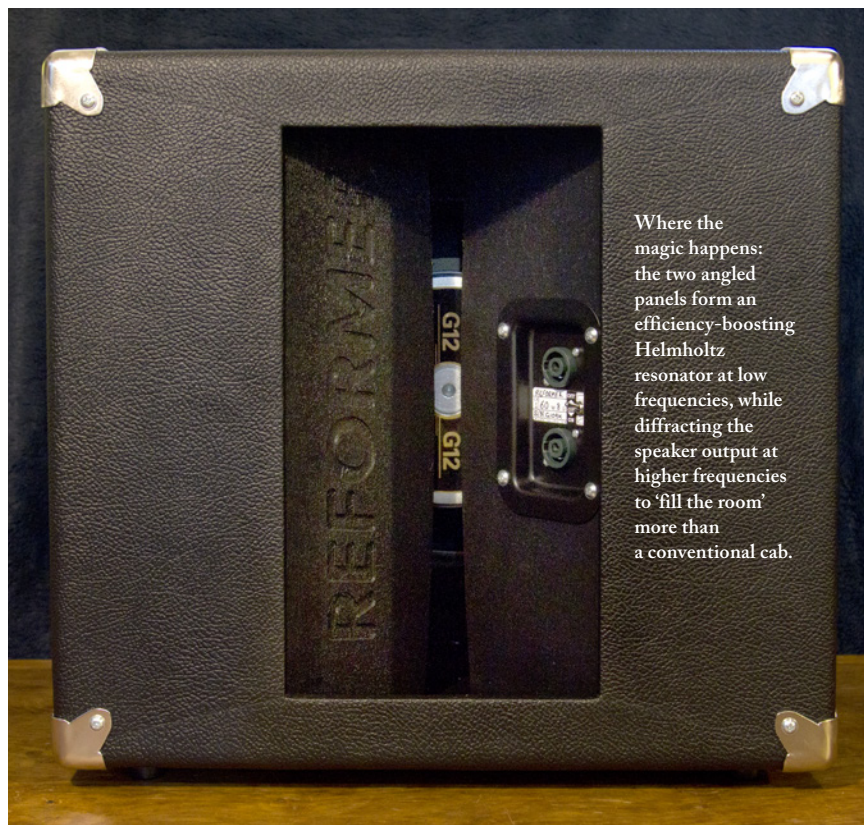
The Design Challenge

The challenge, then, is to design an enclosure that has the useful dispersion of an open-backed cab, but with the bottom-end extension and punch of a closed-back cab. Of course, there are any number of things like horns, waveguides, acoustic lenses and so on that could be used to affect and improve dispersion, but remember what I said about the essential conservatism of guitar players at the outset: all of those will change the sound of the speakers they are employed with, and not in ways that guitar players would find beneficial. Barefaced Audio Managing Director and Chief Engineer Alex Claber started out designing compact bass cabs that sounded bigger than they looked like they should, and his business soon established a reputation for doing innovative things with cabinet designs. His researches, part scientific and part empirical, over several years led him eventually to his patent-pending invention of the Augmented Vent Diffraction (AVD) featured in the 1x12 Reformer model on test.

All the AVD 'action' is round the back — from the front it just looks like a normal cab. At the back you will see two angled baffles that create a narrow vertical slot at the opening between them in the centre of the back of the cabinet. At low frequencies, this acts as a Helmholtz resonator (a tuned port), whilst at mid and high frequencies it works as a diffractor, creating dispersion. Bareface say you get "twice the output of a closed-back cab with an identical speaker and even better dispersion and audibility than an open-backed cab". I'd say that's about right!

This cabinet is also unusually lightweight (22lbs), with its 3/8-inch panels being half the depth of those typically used in speaker cabs of this size. The internal baffles and strategically placed bracing, however, seem to result in a box that is both stiffer and less prone to rattling at high volume than one made of heavier material.

The end result is a sound that is bigger, deeper, wider and louder than any other 1x12 I have ever used. It is still a single 12-inch guitar speaker, so it will 'beam' a bit across the front, but that is now mitigated by the enveloping, room-filling quality of



Where the magic happens: the two angled panels form an efficiency-boosting Helmholtz resonator at low frequencies, while diffracting the speaker output at higher frequencies to 'fill the room' more than a conventional cab.

the sound from the rear integrating with the frontal output. Positioning a guitar speaker to fill a room on an 'un-miked' gig is a bit of an art, and you can still see far too many players cranking up the treble to hear themselves clearly with a cabinet that is firing past their knees whilst lasering a hole in the front row. Some common-sense placement is still required — tilting back is always a good idea — but with an AVD cab, you've got a lot more leeway because the sound is already more diffuse.

Some placements of an AVD cab can actually result in there being too much extra bottom-end, and this is obviously something the manufacturer discovered in testing, too, because this cab features a high-pass filter. It's very gentle but effective at just tightening and cleaning up the bottom end. It can also give you a bit of power-handling safety margin on a loud gig by relieving the cone of a bit of the wide-exursion stuff without affecting the overall sound too much.

Barefaced cabinets are made to order, taking typically about three to four weeks, and you can specify a choice of Tolex colour (black or green) and grille cloth preferences, along with your choice of speaker. The review model came fitted with a Celestion Vintage 30 — not normally one of my favourite Celestions, but here a perfect

complement to the bigger, deeper sound of the AVD cab. You can, however, order a Barefaced cab with any other Celestion 12-inch model you prefer. You can't get an AVD cab empty though — I guess the Barefaced guys want to be sure that they are selling you something that they know will exhibit the expected performance, which is perhaps understandable, if a little frustrating for those like myself with a stack of suitable 'spare' guitar speakers!

Of course, Barefaced aren't the only people to have ever engaged in further development of guitar speaker cabs — Mesa Engineering gave us the extended low-end of their renowned Thiele cabs, and the Nashville-based 3rd Power company in particular are doing interesting design work to eliminate the coloration of cabinet standing waves, for example — but the performance of an AVD cab really is startlingly different in a side-by-side comparison with a conventional design! It is great to see that there is still worthwhile innovation possible in even the fundamentals of the electric guitar world! **///**

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Audeze LCD-1

Planar Magnetic Headphones

Audeze's latest LCD-series cans make the company's impressive planar magnetic technology more accessible than ever.

SAM INGLIS

A tried-and-tested way to build a manufacturing business is to start at the top. If you can establish a reputation for innovative high-end equipment, you're in a good position to exploit the resulting

technological know-how and branding collateral when you move into the mass market. It's a strategy that has paid off for the likes of Focusrite and PreSonus, and it seems to be the path that Audeze have chosen too.

Established by Dr Drag Colich to commercialise his breakthroughs in planar

magnetic driver technology, Audeze have gained a passionate following for their premium LCD range of headphones.

These are popular both in the hi-fi world and amongst well-heeled studio engineers, and I count myself among the fans. Models such as the LCD-X and LCD-MX4 are notable for several things, including their stellar low end, but to my ears their most remarkable feature is their presentation of musical dynamics. Drums are put across with a level of punch and impact that I've never heard on any other headphones, and this doesn't just make >>



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» music more involving to listen to: it also makes it much easier to reliably judge drum levels and compression settings in a mix context.

However, existing models in the LCD series are undeniably pricey, and they are also bulky and heavy. Even if you're willing to take a \$1k+ pair of headphones on the road, it's not always feasible to schlep a large Peli case around with you; and although the LCD-Xs are well padded, not everyone enjoys having so much extra weight clamped to their heads.

The Power Of One

The new LCD-1s will thus seem like manna from heaven to those who have coveted Audeze phones but found them unaffordable or impractical. Audeze already make a lower-cost model called the Mobius, but that is targeted primarily at consumer markets. The LCD-1s are explicitly billed as being "for mixing and mastering applications", and whereas the LCD-Xs cost close to \$2000, these retail at a relatively bargainacious \$399. What's more, they are no larger or heavier than a typical pair of moving-coil headphones, and fold down neatly into a semi-rigid carry case that you can take practically anywhere.

Like most Audeze models, the LCD-1s are open-backed, passive headphones. They are much smaller than the other LCD models and their physical design is a lot more conventional, with rigid plastic

earcups mounted on a padded plastic headband. The earcups can rotate both up and down and fore and aft; they are padded with memory foam covered with "genuine lambskin leather", and as the LCD-1s weigh only 250g, it's easy to achieve a comfortable fit. With so much plastic used in the design, however, they certainly don't have the same luxury vibe as the LCD-Xs, and come across as being rather basic even in comparison to other headphones in their own price bracket. They are supplied with a Y-shaped straight cable that attaches to both earcups using mini-jacks, and which always delivers the right channel to the right earcup regardless of which way round you connect it.

Audeze have put a lot of work into ensuring that their newer models work

"The LCD-1s easily hold their own against open-backed moving-coil designs in the same price bracket, and they have a tonal character that is arguably even more appropriate for mixing than that of the larger LCDs."

well with low-power sources as well as with hi-fi headphone amps, and that shows in the LCD-1s' specifications. With a nominal impedance of just 16Ω and a sensitivity of 99dB/mW, they're plenty loud enough when driven from laptops and smartphones, and Audeze claim a TDH figure of less than 1 percent. They state the frequency response as extending from 10Hz to 50kHz, though no tolerance is given.

Light Touch

The larger LCD models have a tonality that could be described as relaxed, with no obvious presence boost in the upper mids, nor any 10kHz peak as found in many moving-coil designs. This smooth and slightly understated treble no doubt plays a part in allowing the bass and mid-range to shine so spectacularly, but in a mixing context, it can occasionally lead you to underestimate things like tape hiss or vocal sibilance, which might be more apparent on other monitoring systems.

In comparison, the LCD-1s are noticeably more lively, with the presence region being quite a bit more prominent and the middle part of the mid-range a bit less so. They are by no means bright in the scheme of things, though,

and the vast majority of moving-coil designs have a treble boost that is both more obvious and less benign. Of all the other phones I compared them with, the LCD-1s were perhaps closest tonally to my Shure SRH1840s, which have always seemed to me to have a pretty well judged frequency balance. If anything, the Audezes sounded slightly scooped in comparison, with the Shures having more edge in the 2-3 kHz region and the LCD-1s offering richer bass and a little more 5kHz sparkle. This sort of A/B test also highlights the low distortion that is characteristic of a good planar magnetic design: everything seems somehow smoother and less gritty on the LCDs.

In short then, I think the LCD-1s easily hold their own against open-backed moving-coil designs in the same price

bracket, and they have a tonal character that is arguably even more appropriate for mixing than that of the larger LCDs. What they don't have, though, is that unique dynamic quality that sets the LCD-X and related models apart. Playing a rock track with pounding drums through the LCD-Xs is a thrilling experience; you can almost feel the air move as toms are thwacked and snares are pummelled. Switch to the LCD-1s, or indeed any other headphones, and it's as though someone has strapped a brick-wall limiter over the mix bus.

Still, if you buy a Squier Strat, you don't expect it to sound exactly the same as something from the Fender Custom Shop, and ultimately, the LCD-1s need to be judged against competitors in the same price bracket. On this count I think they perform very well indeed, with the major added bonus that they are genuinely portable. If you're in the market for a serious pair of open-backed headphones, then you need to hear these... **///**

Audeze LCD-1 \$399

PROS

- Much more affordable than previous LCD models.
- Easily portable and supplied with a compact carry case.
- Excellent, balanced sound with stronger treble than most planar magnetic designs.
- Easy to drive from any source.

CONS

- They don't have the magical dynamic quality of the more expensive LCD models.
- No-frills plastic construction feels a bit Spartan compared with other phones in the same price bracket.

SUMMARY

With the first affordable headphones in their LCD series, Audeze have thought hard about the needs of mixing engineers. The LCD-1s don't quite have the unique punch of the larger models in the series, but they still sound mighty fine, and you can take them anywhere!

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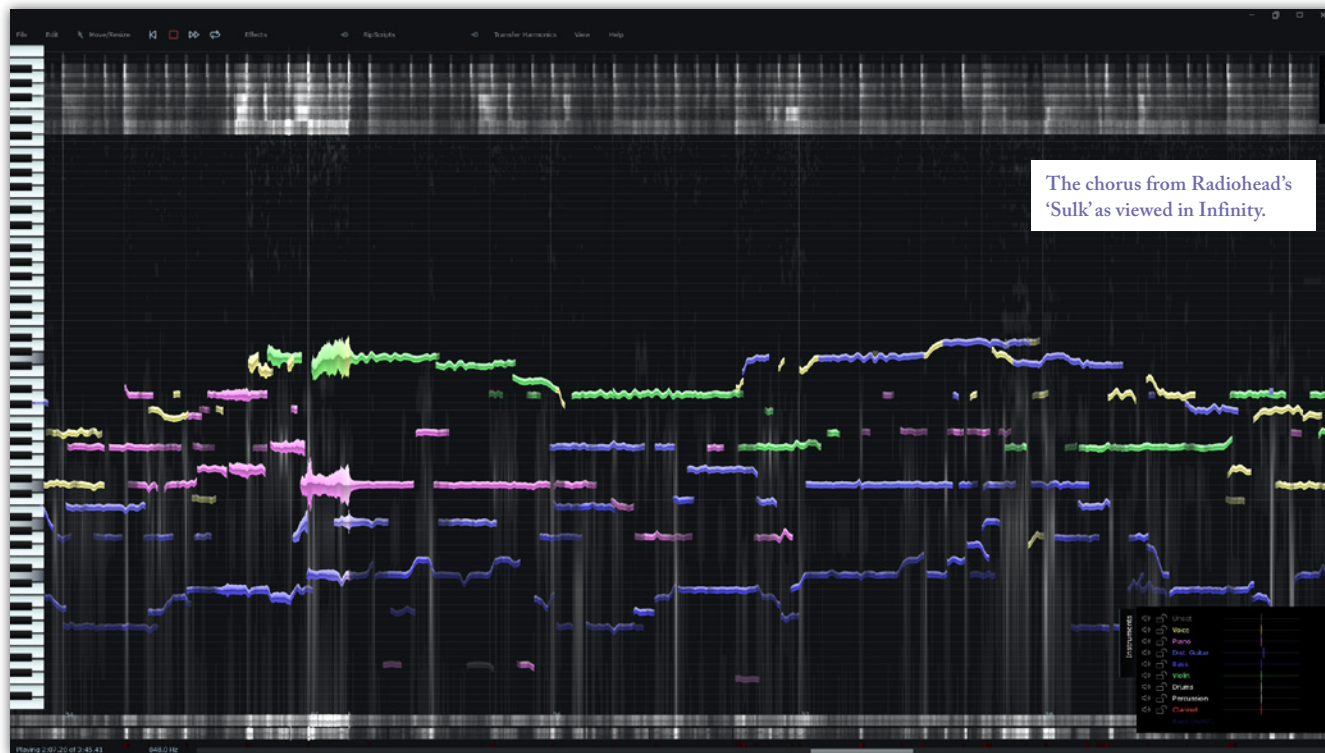
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Hit'n'Mix Infinity

Unmixing & Spectral Resynthesis Software



Infinity is no ordinary audio editor, but one that allows you to manipulate the individual instruments within a mixed audio file in endless ways.

ROBIN VINCENT

It's a remarkable thing to see a mixed piece of music pulled apart and displayed as individually editable instruments. Through the use of its TrueSource source-separation technology, Infinity from Hit'n'Mix can identify the tonal qualities of instruments and split music into fully adjustable notes and percussion. It has the look and feel of Melodyne in some respects, but presents multiple instruments in a single view and allows their individual identities to be blurred, cloned, blended and transplanted. It's not an easy program to explain and it feels like it's yet to fully nail down its own identity, but there's some very clever stuff going on in here.

Hit'n'Mix Infinity does its magic through sinusoidal spectral analysis and resynthesis. What you see on screen are not audio tracks: they are resynthesized representations of musical data created with component sine waves, referred

to by Hit'n'Mix as Rips. The process of ripping is not, as I had initially assumed, one of generating multiple individual audio tracks from mixed audio; rather, it is one of gathering and analysing audio data and identifying musical notes, timbres and tonal qualities. So, the first thing you need to understand is that you are not dealing with audio files but their component ingredients; and until you change something, these recombine perfectly to make an exact copy of the original audio file.

To Rip a file or files, you simply drag and drop them into the Infinity window. Unless you have a fast computer, the process can take a long time, but fortunately the ripper can do its work in the background.

Building Blocks

Developer Martin Dawe declines to explain what's going on behind the scenes; my closest point of reference is the recent electric piano releases from virtual instrument developers Sampleson. Instead

of modelling circuitry or sampling the sound of, say, a Fender Rhodes, they have done deep-level spectral analysis, down to the frequency and amplitude of individual sine waves. The full range of sounds can be recreated by combining the right sine waves in the right way, resulting in a very authentic-sounding Rhodes from only a few megabytes of sine waves.

Martin says that his inspiration came from a David Attenborough piece about the Australian lyrebird, which mimics the sounds of other birds as part of its mating call. The remarkable thing is that the lyrebird will mimic anything it hears — not just birdsong, but all sorts of environmental sounds. The documentary shows it doing strangely convincing imitations of camera shutters, car alarms and chainsaws cutting down trees. Martin was inspired to figure out how to break down any sound to something simple and manipulable like sine waves, creating an algorithmic model based upon the way people perceive notes. He's been rolling

Hit'n'Mix Infinity £299

PROS

- Has the ability to isolate and edit instruments in mixed music.
- Lets you 'paint' with instrument sounds.
- Can transplant harmonic content from one instrument to another.
- Detailed editing of the harmonic structure of sounds.
- Open-ended, with programmable RipScripts that could do anything.

CONS

- Non-intuitive workflow with a challenging learning curve.
- Poor documentation.
- Individual sounds extracted from a full mix don't always sound great.
- It can take a lot of effort to get good results.

SUMMARY

Infinity takes audio apart and resynthesizes it, allowing you to edit it at a fundamental level. It's an intriguing approach to sound that's mysterious, deep and complex, but not always easy to work with.

this around for about 17 years and Infinity sees these ideas made concrete in a piece of software that's incredibly detailed, versatile and unlike anything else out there.

So, what can you do with it? That is the question. In essence, Infinity unlocks the DNA of audio and gives you the flexibility to manipulate it at a molecular level. Sound is

not being emulated and audio is not being streamed or processed: you are applying algorithms to the component elements of sound. These algorithms, or RipScripts as they are called, are written in the Python programming language, and you can get down and dirty with the code directly within Infinity. The RipScripts define the notes and describe the musical elements, dictating how the elements are put back together and what happens when you process them. This could be as simple as adding a vibrato effect to a vocal or as mind-bending as applying the tonal qualities and harmonic content of a clarinet to the sound of a guitar.

Multitrack Remixing

Infinity's ability to pull out individual instruments within mixed music is extraordinary. There's a lot of fun to be had with it, and there's much potential for fixing and tweaking things. It could be useful with live recordings where you want to fix a bum note or maybe remove noise and other artifacts. You can do interesting things like isolating and copying a vocal line from one song and placing it in another. You can then clone the pitch of the existing vocal and apply it to the transplanted one. As a way of isolating previously unobtainable samples it certainly has potential.

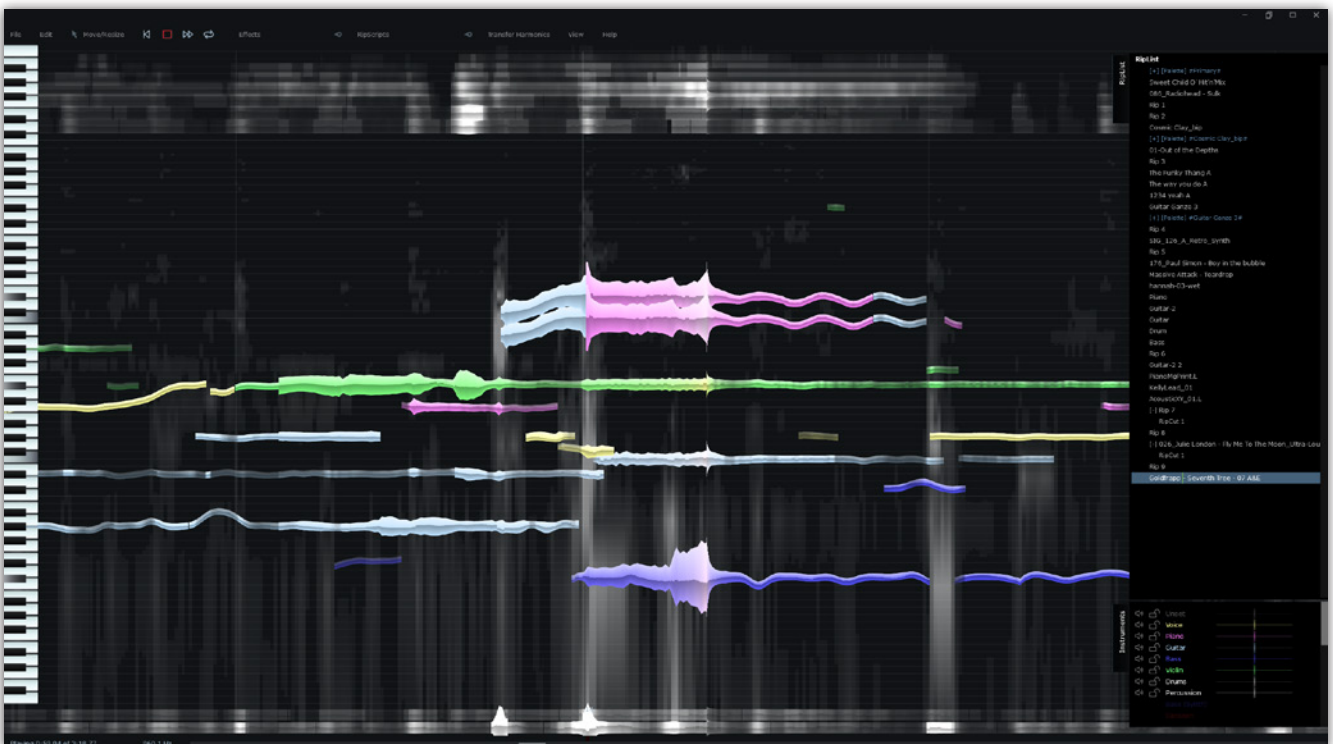
Most of the obvious uses are found in mucking about with mash-ups and remixes, but these have limited real-world

applications in these copyright-twitchy days. Perhaps a more valid application would be to take a recording of you playing live and reharmonise your vocals. But as you use Infinity, a theme emerges: what it does is unbelievably clever, but it's not always clear what you should do with it. Having said that, fiddling around inside mixed music is only one approach to Infinity, and there are a lot more roads to travel down.

Track Editing

The Infinity workspace displays pitch on the vertical axis, in the shape of a piano keyboard, and time on the horizontal axis. It detects tempo and displays bars for you (which can be adjusted) and places a measure-based grid onto the music. Notes are displayed in a colour corresponding to the detected instrument type, with brightness indicating amplitude. Infinity auto-detects from a list of real-life instruments such as violin, piano, bass, trumpet and so on, depending on the harmonic content it identifies within the sound. I found this worked to a degree, but whenever I dragged in a guitar track it always seemed to decide that it was a piano. It's not desperately important to get this right; what matters is more that Infinity correctly separates notes and assigns them consistently to the instruments.

The notes are not always in one piece; Infinity may detect a single note as multiple



Not only can you isolate individual voices and instruments within mixed audio, but you can clone them and add harmony parts, as shown here.

» notes because of the nature of the playing, in which case you'll end up with a number of note fragments. You can stitch these back together manually so that when you are selecting them and moving them about they stay together. Around the notes like ghostly echoes across the whole frequency range is a cloud of grey and white material. These are the more percussive frequencies

that have been detected and are not seen as part of the note. High-frequency energy within this category is labelled as Percussion whereas the low and cross-spectrum frequencies are reported as Drums. These get their own tracks on the instrument list and so you can turn them off to leave you with a pure instrument sound, though that doesn't sound entirely

right; most real-world instruments contain some unpitched material, and killing those elements completely can make it sound a little strange. But it depends on the source material, what you're trying to achieve and the amount of work you want to put in. As an example, Infinity had no problem ripping out all four notes of an electric guitar playing some jazzy chords, and this worked »

RipScripts

RipScripts are scripts or processes that are applied directly to the note and musical information stored in the rip. The Chord Creator is a good example that's easy to grasp, both visually and aurally. Select a note and hit Chord Creator, and Infinity automatically generates a chord around the chosen note. For this process you do get a little panel where you can select major or minor key. You can also specify an arpeggio time, which staggers the notes. It's easy to forget that you're dealing with sinusoidal resynthesis of spectrally and musically analysed audio rather than MIDI.

However, it was also at this point that my unfamiliarity with the Infinity workflow started to turn to frustration. With the Chord Creator panel open, any note I click instantly gets a chord. If I change a setting in the panel it then gets another chord over the top. As I still can't quite work out how to reliably start playback from exactly where I want to, I find myself accidentally generating chords as I try to select something else, before spending a lot of time leaning on Ctrl+Z to undo my way back to something sensible, then getting in a knot as to what is playing back and which chord I'm generating. Martin has said that he wrote this software to enable his own

creativity, but sometimes it's hard to know how he's thinking about the way things work. Pressing Escape seems to be vital in turning RipScripts on and off, a fact that is worth remembering.

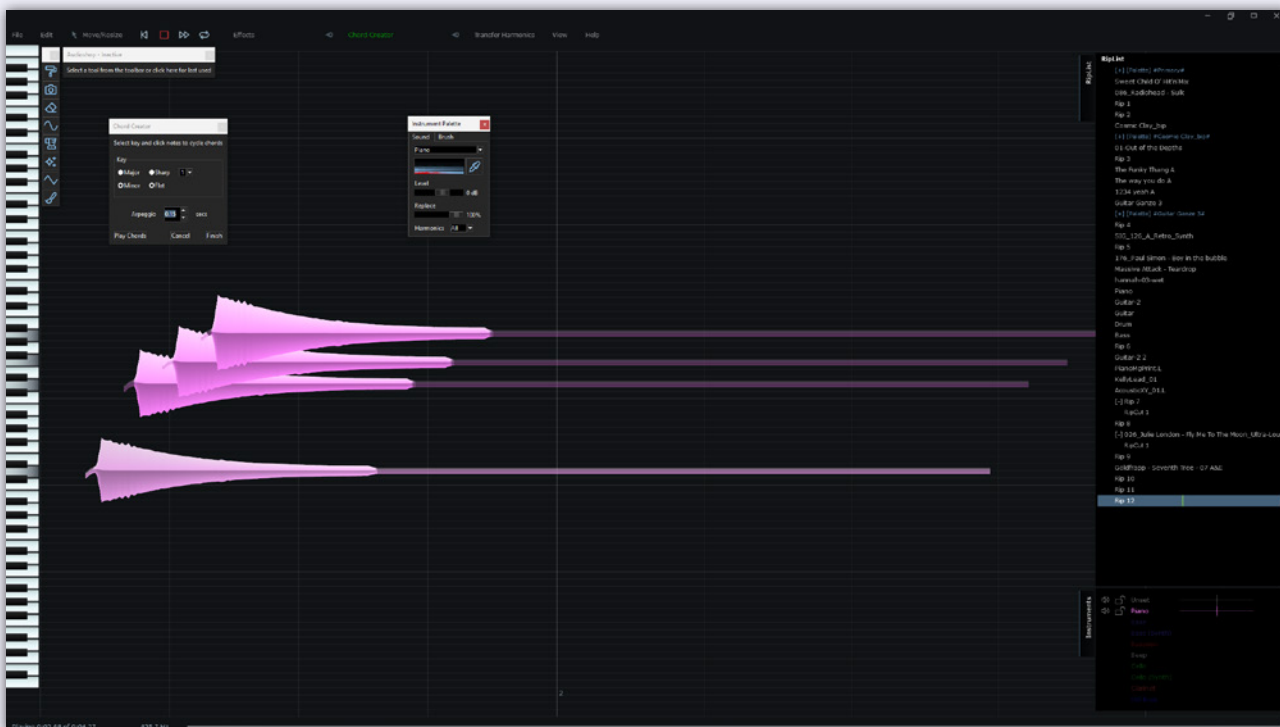
The other supplied RipScripts include tools for cleaning frequencies and phase artifacts, mapping a volume change across a number of notes and various effects. One is called Infinity Scale, and applies the Shepard Tone auditory illusion to the selected notes; whether this is for dramatic or comedic effect, I'm not quite sure. Another such tool is Inharmonicity, which adjusts how far the harmonics depart from whole-number multiples of the fundamental. It sounds really interesting, but is hampered by an interface that lets you enter a single number where apparently something between "0.0 and 0.1 is most useful". Not having any reference for what the number means, you are left experimenting with one hand on the Undo button.

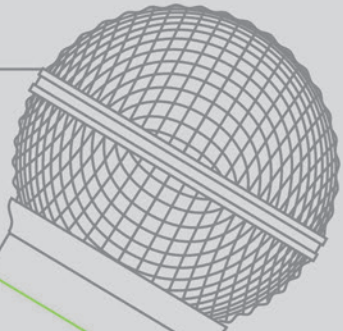
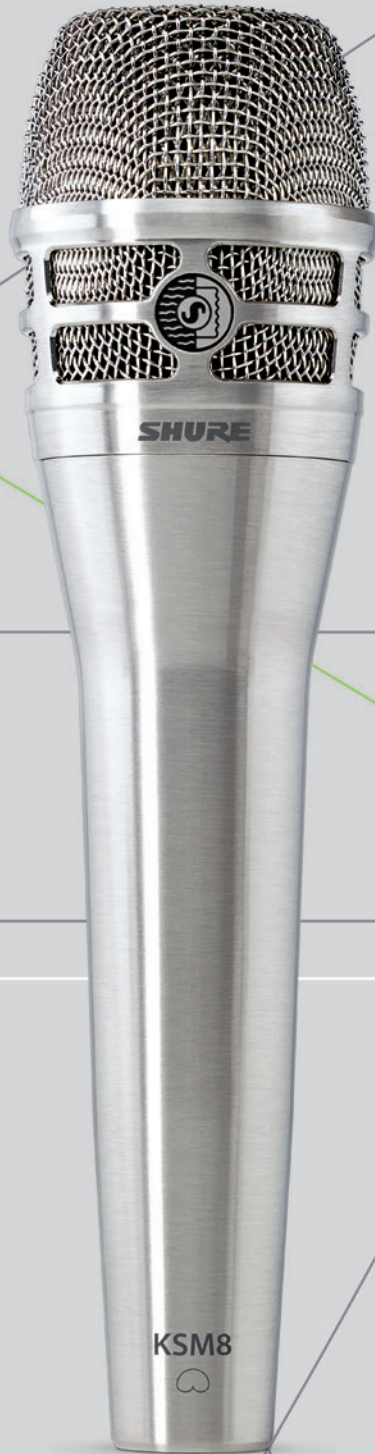
In fact, Infinity often made me feel like I was missing something fundamental. A thorough understanding of harmonics is something we all naturally possess, so the outcomes of these processes, tools and functions should be obvious, or at least discoverable through experimentation.

Yet, although there's nothing wrong in expecting a certain level of intelligence from a user, I was often left scratching my head and wishing that the documentation contained more clues.

On the other hand, this is exactly what RipScripts are all about. They are infinitely variable, developable, and improvable. For instance when I first tried Infinity the effects couldn't be turned off once enabled. If you didn't want the effect you had to remember how many times you'd applied it in order to press Undo the right amount of times. After a discussion with Martin, he edited the script and a Cancel button arrived that undid the effect you were using. I moaned about the lack of tempo information and he conjured up a script to display the tempo. Any problems you have with the usability, feature set, or way of working can probably be solved through writing or improving the scripts. This could be an immensely powerful feature, and Hit'n'Mix hope that a community of users will rise up to write and share their own scripts.

One of the supplied RipScripts is the Chord Creator, which automatically generates a chord to accompany the selected note.





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» quite well when I muted the Drums and Percussion tracks; in fact, the rattling and the hiss on the track were removed very neatly. If I approached things from the other way around and muted the guitar, I was left with quite an unusual percussive guitar sound.

The instrument list gives you a volume control for each instrument, so rather than removing elements entirely, you can just dial them down a bit. You can also delete elements directly from the high- and low-frequency bands as you can with any note. Infinity splits the Percussion up into eight bands of high frequencies and the Drums into two bands of low frequencies, which you can select, audition and delete independently. Clicking any note or percussion element will audition that part in isolation, whereas a double-click will play it back in the context of everything else.

Once you've ripped a track, you can move individual notes in both pitch and time in whatever way you fancy. You can change the chords, change the modes, transpose, fix timing issues, copy/paste, shorten and lengthen the notes... and we haven't opened a single menu yet! There's so much going on within Infinity that it's difficult to explain it all and not get distracted by one element when there are so many other things to talk about.

Effects

Let's rip a vocal track and see what else is possible. Sure, I can correct the pitch and all that jazz, but let's move onto things that other software can't do. Infinity has two main menus: Effects and RipScripts. The Effects are in fact RipScripts but these particular ones are a little bit special because they are applied at the playback/rendering stage so they can be turned on and off, whereas RipScripts are applied directly to the stored notes and would have to be undone.

That's a good point to make at this juncture: Infinity is very destructive. It automatically saves and overwrites itself, so any messing about you've done is written over the Rip, and there's no way back unless you keep clicking Undo until you've got back to where you want to be (if you can remember what that was). When I pointed this out to Martin he created a 'Save copy as...' option, which helps, but with all the subtle changes possible, it can be very difficult to work out if you've undone everything. A Cancel button or the ability to not save your changes would be beneficial.

Under the Effects menu we have 10 mostly pitch-related processes that can be applied to the currently selected

notes. These are Harmony, Quantize, Correct, Flatten and Slide Pitch, Vibrato, Shift Formant, Volume, Panning and Reverb. Working with my vocal track, the Harmony effect throws in a very convincing pitch-shifted second part. As these are generated at playback, the extra part is not shown on screen; nor does it have a control panel, GUI or options, so you can't alter the volume or any other aspect of it other than shifting it up and down an octave. It's the same deal with all the Effects: they all do what they say on the tin but there's no control over how much, other than the odd up and down button to add more or less volume, reverb, vibrato and so on.

For harmony-related editing, it's much better to select what you're trying to harmonise, duplicate it using copy and paste, and move the newly created notes up and down in the main screen. You have far more control, you can apply volume and panning changes, offset it slightly and make use of the remarkable Draw Pitch tool to add slight variations to the pitch so the harmony sounds different to the original. This works so well that you wonder why the option exists under the Effects menu.

Deeper Levels

There's an even deeper rabbit hole we've yet to venture down, which concerns harmonic content, instruments, the Audioshop and Note Editor. All the pitch-shifting, harmony and effects are just so much messing about on the surface: the real depth lies in editing the harmonic content of individual notes and sounds.

This is done through something called the Instrument Palette, which borrows ideas from image-editing software such as Photoshop. Using a combination of tools, you can paint notes into your Rip project, or paint the harmonic content of one sound upon another. This can be a total replacement or a blending of timbre, all the way down to individual amplitude control of individual harmonics.

The Instrument Palette contains a number of pre-ripped sampled instruments, and the Audioshop is a selection of tools that define how the instruments in the palette are used. So, for instance, by selecting Flute in the Instrument Palette and using the brush tool, you can paint notes on the main Infinity screen and compose flute music. It initially feels very MIDI-like but as you start adding other instruments into the same space, it starts to take on a life all of its own. Creating music with multiple instruments in the

same window can be slightly odd because we're so used to the multitrack paradigm, but in practice it feels very natural and organic, perhaps offering a glimpse at the compositional thinking behind the software interface. But, on the other hand, most DAWs now offer multiple track MIDI editing in the piano roll, and you'd have far more control over the sound using virtual instruments or samplers. There's very little tempo structure to the main Infinity page; you can quite cleverly adjust the bar lines, and doing so pushes and pulls the tempo in a very free-flowing way, but there's no snap or metronome to help you. My request for tempo information was fulfilled, but if you change the tempo it time-compresses or stretches any already placed notes so they no longer sound right. I'm sure a rewrite of the script would sort that out.

But we haven't even got to the interesting bit yet. The next tool is the Clone tool, which enables you to copy the pitch, formant, timbre, volume or panning of one note and apply it to another. Ctrl-click on a flute note, clone the timbre and then click on a piano note, and you apply that harmonic content to the piano sound. This opens up a load of sound design and hybrid instrument possibilities, provided you can get your head around the interface — and I confess I found this a struggle. When you're trying to understand what you're hearing and what's going on, there are very few clues that can be gleaned from the process or represented visually on the notes. It's not that Infinity doesn't do what it says; it's just that I can't always work out how to do anything with any accuracy or intention. I did succeed in cloning the vibrato of the flute onto the bass guitar. I could hear the difference and I could see the wobble on the screen. But as for timbre or formant you are left to experiment with different ways of clicking, different level settings and random parameter changes because there's little documentation on how any of it works.

The Remove Timbre tool probably does what it says, but in practice sounds like it just makes the note quieter. The Blend tool is similar to the Clone tool but it takes the pitch, timbre, formant, volume or panning from a sound in the Instrument Palette rather than the main editor. Applying the timbre and formant from a trumpet onto piano notes sort of works. No, it really does work — it's just that the annoying level of experimentation required, the constant undoing and reapplying, the difficulty of placing things accurately and the whole method by which you can audition the note »



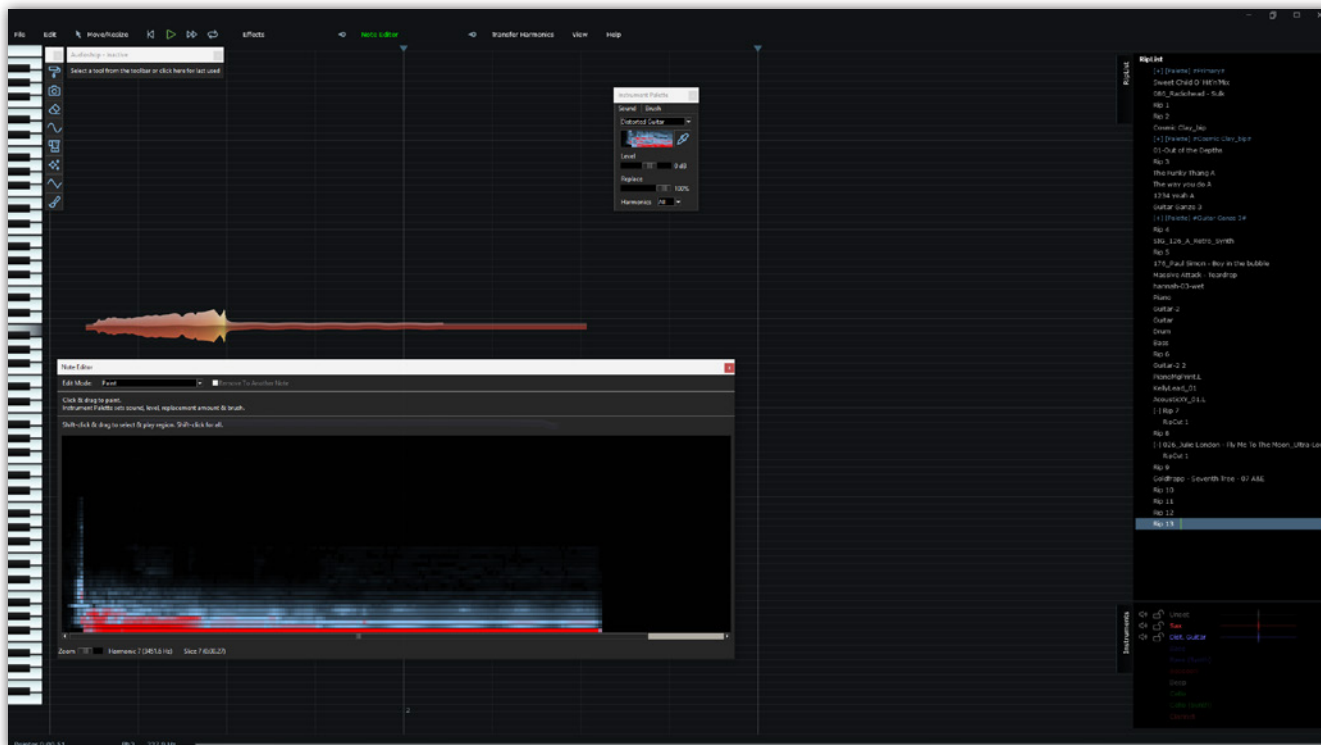
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Toontrack.



Here I've taken the tonality of a distorted guitar from the Instrument Palette and I'm painting it onto a saxophone.

» without accidentally applying the tool again starts to bring me out in hives.

There are, at least, a couple of tools that are wonderfully simple and effective. The Pattern tool lets you apply a sine- or square-wave modulation to formant, pitch, volume or panning. You can set the strength and rate and add some nice tremolo or vibrato to a note, which you can see displayed on the note. You can't see any change when you apply formant modulation, but you can hear it well enough.

Note Editor

The last tool on our Audioshop workbench is the Note Editor. With this tool we get a complete spectral readout of the selected note, which lets us manually edit all of the harmonic slices within that note. If trying to do similar things with the other tools feels inaccurate and frustrating, this is the place to sort it out and really get your hands dirty. You have several edit modes that do things like replace a harmonic slice with the same one from the instrument in the Instrument Palette, delete slices, adjust the level, remove noise spikes or flatten the harmonics, and some other processes I don't really understand. But where this works at my level is that I can take the sound of a saxophone and sketch it over the sound of piano, blending the breath of the sax onto

the thump of the hammer. Or I can make weird sounds by killing all the odd harmonics or scribbling changes about the place. This is perhaps where I should have started, because it does what the Clone and Blend tools do better than they do, and makes you wonder again why those tools exist.

A limitation of the Note Editor is that, as the name suggests, it only deals with a single note rather than an entire instrument or a selection of notes. However, once you've edited the one note, you can suck it into the Instrument Palette and then paint a new tune with your newly harmonically compromised instrument. That throws up a whole other bag of possibilities in creating paintable instruments within Infinity from your own sampled material. It starts to become a sort of sampler but with harmonic and additive synthesis qualities, though this aspect of it is not yet fully developed.

Finite Thoughts

Infinity is hugely clever, but the technology sometimes feels as though it's searching for the right application. It probably needs more time for a bunch of enthusiastic early adopters to work with Martin to shape and hone this undoubtedly remarkable software into something a bit more coherent. The graphics on the screen are friendly and inviting, but the processes involved in achieving anything can sometimes feel clumsy and unforgiving. Perhaps once you've learned the keyboard shortcuts the

workflow will start to reveal itself, or maybe it's because it's so far away from the standard DAW approach that my preconditioned brain can't quite make the leap.

Martin is certainly a very active developer, and incremental releases of Infinity are constantly adding new features. One that will particularly appeal to pro-audio users is that Infinity can now be invoked from an AudioSuite plug-in in Pro Tools.

You can do amazing things within Infinity to craft sounds and alter harmonics, blending the sound of one instrument into another and creating new instruments while all the time painting with sound and with music in ways I've not encountered before. In the right hands, it could offer a very different approach to mixing sounds and blending ideas from unexpected places. It can be launched from within your DAW as an audio editor, opening up potential for doing weird stuff as an experimental tonal editor to add to the toolbox of sound designers and timbre manglers. Or if you feel frustrated with the boundaries and structure of your average DAW then you may find Infinity offers an alternative free-flowing compositional adventure. Infinity could be great! **///**

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Relab Sonsig Rev-A

Algorithmic Reverb Plug-in

Simplicity meets sophistication in Relab's impressive new reverb plug-in.



SAM INGLIS

Danish software company Relab Development have been operating 'under the radar' for over a decade, mostly developing tools for other manufacturers. Relatively few products have been released under the Relab name — but that's changing, and the first fruits of their new focus are a suite of reverb plug-ins.

LX480 and VSR S24 are, respectively, recreations of the Lexicon 480L and TC System 6000 VSS6.1 hardware reverbs, but Sonsig Rev-A is an original design. Relab say that their aim is to combine high-end professional sound quality with the ease of use of guitar stompboxes, enabling users to dial in the right settings as quickly as possible. Sonsig Rev-A thus has a large,

friendly and very purple user interface, with no page flipping, IR loading, menu surfing or other such atrocities to slow you down. It's available both for Mac OS and Windows, in all the major native formats, and is authorised using the iLok system.

Tails You Win

Unlike most algorithmic reverbs, Sonsig Rev-A doesn't differentiate between early reflections and the reverb tail, at least from a control point of view, which is partly why the user interface is so uncluttered. The two largest dials are reverb Time and Size. These interact with one another, so you can, for example, achieve a two-second reverb by turning the Size down and the Time up or vice versa. The RT60 may be the same in both cases, but the sound certainly isn't!

A large-but-short two-second reverb has an expansive and relatively natural room or hall sound, whereas a small-but-long two-second setting produces something more akin to a highly reflective chamber, with that elusive quality that Paul White appropriately describes as sounding 'steamy'.

The Diffusion control ranges from cold, slightly austere and gritty to lush and enveloping, and delivers usable results throughout the range of its travel, which isn't always the case. I'm not sure I can say the same for a Pre-delay function that goes all the way up to 800ms, but it's easy enough to dial in the setting you want, and it can be sync'd to host tempo.

There's also a clear sonic difference between the Character settings. These are adjusted using a simple three-position switch, but there's obviously a lot more going on behind the scenes. In the first position, the reverb tail seems to kick in pretty much straight away, while the other two offer progressively slower build-up of reflections. The all-important modulation in the reverb tail is likewise controlled by a single slider labelled Ensemble, but again, this is clearly adjusting multiple parameters within the reverb algorithm. There seems to be some modulation within the tail even with Ensemble set to zero, but it's relatively subtle and quite natural, suggesting real spaces rather than special effects. As you turn it up, you get progressively more pitch

Relab Sonsig Rev-A \$149

PROS

- A versatile algorithmic reverb that offers great-sounding ambiances, rooms, halls and more.
- Extremely easy to use, with a minimal control set that is nevertheless highly effective.

CONS

- None.

SUMMARY

Relab have drawn ideas from the best algorithmic reverbs of the past to create a plug-in that can cater to almost every application at the turn of a dial. If you appreciate a good-sounding reverb but don't like to be slowed down by complex user interfaces, Sonsig Rev-A could be perfect for you.

variation, which can be distracting on some instruments but adds a welcome charm and thickness to others, and never quite becomes ridiculously over the top.

Tone Quest

Sonsig Rev-A does not have conventional equalisation controls: instead, the tonality of the reverb is set using Brightness, Hi Decay, Tilt and Low Tilt parameters. Brightness introduces a low-pass filter, while Hi Decay determines the degree of high-frequency absorption in the synthesized space — in most natural environments, high-frequency energy is more readily dissipated than low frequencies, so the top end of the reverberation dies away more quickly. Tilt is, as the name suggests, a sort of 'tilt EQ', which rebalances the levels of low and high frequencies around a central fixed point, and Low Tilt is more like a separate damping or absorption control for the bottom end.

All of these work very well, though it isn't immediately obvious which way round some of the controls function. Positive Low Tilt values produce a tilt away from the low frequencies, which wasn't what I expected, but this only took a second to figure out.

On stereo reverbs, I imagine many of us default to leaving any Width setting at its maximum value; here, that delivers a super-wide sound which is certainly impressive, especially on headphones, but I often preferred the results that I got by backing it off somewhat. A good test of an algorithmic reverb is how it fares when the mix is collapsed to mono, and with any Width setting below 75 percent or so, Sonsig Rev-A scores very highly in this regard. Mono reverbs can be surprisingly useful in mixing, and Sonsig Rev-A retains its richness no matter how narrow you make it.

Finally, the three buttons on the right-hand side beneath the meters and gain controls shouldn't be overlooked, because they make a surprisingly large difference to the sound. Each selects a 'render mode', the first two of which are based on the Quantec Room Simulator and Lexicon 224. I particularly liked the QRS setting, which is quite dark, can be somewhat gritty, and seems to fit perfectly in the mix, obliterating at a stroke problems with splashy sibilants and so forth. The 224 setting is also very pleasant, applying a much gentler but still obvious high-frequency roll-off.

Rev-A gives you the full-bandwidth experience and can be much brighter than the other settings.

In Action

They say it's preset number 1 that does the most to sell a synthesizer, and if the same is true of reverb plug-ins, Sonsig Rev-A will fly off the shelves. It actually took me quite a while to start exploring the controls simply because the default patch is so 'right' on almost anything! Once you do start tweaking, you quickly begin to appreciate the sheer range of sounds that is on offer from relatively few parameters. Any parameter can be locked simply by clicking on its name, in case you want to recall other aspects of a preset whilst retaining the same wet/dry balance or other setting.

As is probably apparent from the foregoing description, Sonsig Rev-A belongs to the tradition of algorithmic reverb design that was pioneered by Lexicon in the late '70s and early '80s, and is best seen as a refinement of existing technology rather than striking out in a new direction. As such, it shares the good qualities of those old hardware units, but updates them with a sound that can be more hi-fi if you want it to. Like most algorithmic reverbs, its recreations of mechanical reverberators such as plates are more suggestive than pinpoint, but unless you need a very specific vintage sound, you'll never be stuck for settings that actually work.

I particularly liked Sonsig Rev-A as a vocal reverb, in which role it offers everything from subtle ambiances to dense chambers or expansive halls. Whereas some reverbs become metallic or 'bark-y' at very short settings, this one sounds great, adding thickness and life to the source without overwhelming it. And it's not only the variety of sounds on offer that make this such a useful tool: it's also the speed with which they can be dialled in. With a few quick twists of a knob you can get nuanced studio spaces, splashy tiled rooms, understated theatres and cavernous barns, with no need to switch between algorithms or select different room types. Don't be fooled by its simplicity: this is a thoroughly professional, high-quality algorithmic reverb that also happens to be incredibly easy to use. **///**

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Novation LaunchKey Mini MkIII

Controller Keyboard



Novation's latest LaunchKey Mini offers more features in a smaller package.

ROBIN VINCENT

There's no shortage of mini MIDI controllers knocking around for about \$100 and so any manufacturer hoping to capture this market is going to need to pull something a bit special out of the bag. The previous versions of the Novation LaunchKey Mini have held their own against the likes of the Arturia MiniLab and the Akai APK Mini, so does the updated MkIII offer a creative advantage for mini-keyed music makers?

Form & Function

I find it remarkable how the smallest tweaks to a design can make previous

versions appear dated so quickly. Novation have followed Native Instruments into the neatness of straight lines and square angles. The MkII had the smallest curves on the case and graduations on the knobs, but I wouldn't be seen dead with such old tat now that the MkIII is here. The clean lines are beautiful, the straight down encoders futuristic, and even the keys have been flattened to make them feel nuanced and less chunky. It looks classy and understated with the matte grey/black paint even if it is still made of plastic. It sits over 1cm lower on the desk, and the pads and buttons have been slimmed down so as not to spoil the flow.

Novation LaunchKey Mini MkIII \$109

PROS

- MIDI Output.
- Arpeggiator Mutate and Deviate modes.
- RGB colour pads.
- Pitch and mod strips.
- Perfectly sized.

CONS

- Needs additional power with iOS.
- No TRS MIDI adapter.
- Shift-key shenanigans.

SUMMARY

With the LaunchKey Mini MkIII Novation have brought back some old favourites with pitch/modulation strips, sustain pedal port and a MIDI output while adding an enjoyable arpeggiator, full-colour pads for Ableton Live integration and a slimmer, more stylish look.

The button, knob, key and pad count are almost identical to the MkII with the addition of a couple of extra buttons. But the most prominent physical developments are the pitch and modulation strips, whose presence should be standard on every MIDI controller. Call me old fashioned but I do enjoy a good modulation



wheel. However, if you're looking for miniaturised control and sleek lines then a touchstrip is a small compromise. You don't expect to get too excited about the feel of miniature keys, but these feel OK — there's plenty of spring and resistance in a small amount of travel.

The LaunchKey Mini is lightweight without feeling flimsy, gives good resistance to my twisting and banging it about and has half-decent rubber feet for keeping it still on the desk. If I could find any complaint at all for the build quality of a plastic controller it would be that there's perhaps a bit too much wobble in those very petite encoders.

On the back you get a regular USB port that serves as both power and computer connection and that all-important Kensington security slot for when you're making music in a cafe and have to nip to the loo and don't want anyone to steal it. But that's not all. With the MkIII Novation have added a useful Sustain pedal socket that can be mapped to anything via the Novation Components software. And they've added a MIDI output port which is a cause for

The LaunchKey Mini MkIII's back panel is equipped with a USB-B port, a quarter-inch sustain pedal input and a mini-jack MIDI Out port.

celebration. The MIDI Out is on that increasingly common TRS mini-jack output, for which they've decided not to supply an adapter. No problem, I thought, I'll use the one that came with my Novation Circuit — except I can't because it's not compatible. The LaunchKey uses the recently accepted standard of TRS A whereas their older devices all used the now-defunct TRS B. See the 'Mini MIDI Standard' box for more details. I have a slight twinge of disappointment in not seeing a CV/Gate output, which was a very welcome feature of the SL MkIII, but that would be a lot to expect.

Ableton Live

As it says on the box, this controller is designed for making tracks in Ableton Live and specifically in Live version 10, where you can make use of the MIDI Capture feature to record your playing after you played it when you weren't in record by hitting 'Shift-Record'. Novation have been running this integration for quite some time and the LaunchKey slips effortlessly into Ableton Live controller mode when you launch the DAW.

The main functionality revolves around the pads and the knobs. Holding Shift and hitting the right pad can change the pad mode between Session, Drums and Custom and the knob mode between Device, Volume, Pan, Sends and Custom.

'Session' transforms the pads into a reflection of the Session View in Live. A red box encompassing two rows of clips over eight tracks appears on your computer screen, and that information is shown on the pads in perfect RGB colour, which is a lovely upgrade from the three colours of the MkIII. You can launch any of the visible clips by tapping the pad or you can launch the top row as a scene using the pad/button to the right of the eight pads. The bottom row no longer has its own launch scene button, but instead lets you step through some additional tools. Tap the second row button once and they all become red stop buttons pertaining to the top row, tap again and they become blue solo buttons and on a third time they are yellow mute buttons. This gives a satisfying level of control over your clips

Mini MIDI Standard

TRS mini-jack MIDI works because MIDI only uses three wires within the traditional five-pin DIN cable and so can easily be accommodated by the three connections of a TRS connector. But when TRS mini-jack started turning up on devices no one had thought to declare a standard, and so the three wires of MIDI have been connected in two different ways inside the mini-jack connector. The MIDI Standards overlords caught up, named them TRS A and TRS B and declared that TRS A should be the standard. Novation have been using TRS B on all their products up until now, so any adapters from your older Novation products won't work with the TRS A-fitted LaunchKey Mini MkIII. It's nothing that a bit of soldering or some adapters won't cure, and as this standard persists we'll be able to connect MIDI directly with stereo TRS mini-jack cables — which, to my old-fashioned brain, just seems weird.

while sacrificing the ability to trigger the next scene on the second row.

Track navigation is handled by holding the increasingly popular Shift key and pressing the first and second row buttons for up and down and the Arp and Fixed Chord buttons to move between tracks, which will ultimately move what clips you see on the pads. And this is where something odd happens with that Shift key. Pressing Shift results in a sort of status view displayed on the pads. It shows you what mode the pads and knobs are in. So as you hold Shift to navigate to a different row or track that clip information is no longer displayed on the pads and you are forced to refer to the computer screen. That can be very disconnecting and rattles the vibe of performing from the keyboard. The Shift button's position is on the opposite side of the keyboard from the navigation keys, so both hands are tied up when trying to move tracks or scenes. On the MkII there were dedicated buttons for track navigation, which always left a hand free for doing musical things. This feels like a regressive step. I wonder whether it would be possible to add a latching function for the Shift button? >>





The LaunchKey Mini MkIII measures 330 x 172 x 40 mm.

» 'Drum' mode refers to the Live Drum Rack and helpfully maps itself to the first 16 drum sounds loaded in the rack — assuming you've loaded a kit. You can use Shift and the Scene navigation keys to access all 128 possible sounds within a kit. The pads are smaller than what you'd find on most drum controllers but they are perfectly adequate for finger drumming and give a decent level of velocity control. In fact, the smaller size gives you better access to more sounds without having to stretch your fingers too far.

'Custom' mode lets you map the pads to whatever notes you want using the Novation Components software. By default it drops to a melodic C-minor scale and this creates something interesting — it becomes an extension of the piano keyboard. You have, more or less, an additional two octaves to play with on the pads, which means you could easily be playing some low chords on the keyboard and a higher lead on the pads without having to mess around with the Octave buttons. Using Components you could allocate the pads a different MIDI channel and have the keyboard and pads play different instruments. That's quite handy considering the fairly restrictive two-octave range of the keys.

On to the knobs, then. These map themselves neatly to the first eight parameters of the selected device. That's particularly nice when your track has those eight macro knobs assigned. There

doesn't seem to be any way to move to the next group of parameters or the next device in the chain, which is a shame. Volume and Pan are self-explanatory. With Sends you are able to switch the control between any of the sends you have set up by holding Shift and selecting it from the second row. Seems to me that would be a useful way to enable banks of control for the Device mode.

'Custom' mode for the knobs releases them from Live specific control and lets you map them to any parameter in Live or a third-party plug-in. Their assigned MIDI CC numbers are editable in the Components software.

The Live implementation is comprehensive and for the most part smooth. You can select tracks, record clips, add automation, mix and perform directly from the LaunchKey without much reference to the computer. Other than the annoyance of the disappearing clip view and two-handed navigation the workflow is easy and seamless, unlike my playing, which would really benefit from a sadly lacking undo button.

ARP

This is a function brought over from the SL MkIII that might just have the ability to sell this controller all on its own. The Arpeggiator is thoroughly engaging and makes good use of the layout by spreading settings and modes all over the place.

Like using the Shift button, all the modes and options for the arpeggiator are accessed by holding the ARP button. These modes are spread across the keyboard, starting with seven modes, then five rate options, four octaves, five rhythms and then a Latch on/off. The pads also show a colourful version of the modes and options, with five pages of modes on the top row and then the options selected on the second row. That's very handy for when your hands are on the keyboard, making it difficult to select modes on the keys you are playing. As with Shift, you have to hold the ARP button down to access these settings, which tends to tie up both hands. The Latch key solves this in most cases so that you can take your hands off once you played a chord, but it does make me wish that the ARP, as well as the Shift buttons, had their own ability to latch on.

The modes and other settings are the ones you'd expect from a decent arpeggiator, but there are two unusual ones that make this a bit special. 'Mutate' is a mode and 'Deviate' is a rhythm and they are both enormously fun. Once enabled the intensity of their effect is dictated by knobs 4 and 5. Mutate throws its own notes in there from a gentle, perfectly placed variation to a mess of all sorts of things. Once you've found your melody stop turning the knob and it will repeat indefinitely. Deviate does the

same sort of magic to the rhythm, from cheekily missing out notes here and there to radically overhauling the vibe you were going for.

There's also control over the gate length, swing and tempo. But the tempo control only works when Ableton Live isn't playing or when outside that environment.

Along with the ARP the other new function is Fixed Chord. Hold the button, play some notes and now every note you play will play that chord transposed to the key you are playing. The chord is stored until you hit the button again to store a new one. Simple and effective.

Beyond Live

Novation have provided some similar integration into Logic Pro X. Track selection and navigation is all there along with a drum mode, mixer controls and so on. Reason 10 will map the knobs to useful parameters of any selected instrument. Otherwise HUI support is provided for transport control, Volume, Pan and Send control and navigation in most other DAWs.

“For a mini-keyed controller that’s barely wider than my Microsoft Surface Pro, the LaunchKey Mini MkIII is hard to beat.”

As a regular MIDI controller the knobs are all available for mapping. The Components software gives you a lot of flexibility when matching up to controls on other hardware devices using the MIDI output.

On iOS or Android devices the LaunchKey can be mapped to instrument apps, provided you have a connection that provides power. Unfortunately, the LaunchKey won't work if you try to power it from your phone or iPad. But adding a USB power pack is enough to keep the LaunchKey and tablet combination viably portable. Mapping the knobs is easy and there are plenty of iPad synthesizers that

would benefit from a bit of creatively Mutated and Deviated ARP action.

Conclusion

The LaunchKey Mini MkIII manages to pull together a bunch of creative features in a neat and stylish package that sits deliciously low on the desk. The new look is on point, the return of the pitch and modulation strips is very welcome, the ease of control in Live is as smooth as it should be and the fun-factor arpeggiator and MIDI output set it nicely apart from the crowd. The workflow issues around the holding of the Shift and ARP buttons are small narks and could probably be ironed out with a firmware update. For a mini-keyed controller that's barely wider than my Microsoft Surface Pro, the LaunchKey Mini MkIII is hard to beat. Which leaves me with one question, what are those symbols on the top three keys? **///**

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Yamaha DZR10-D & DXS15XLF-D

Networked Active PA Speakers

Yamaha's newest powered speakers not only offer excellent sound and portability, but their optional Dante connectivity makes them a cinch to set up.

MIKE CROFTS

A new line in self-powered PA speakers always interests me, and as I already have several Yamaha speakers in my inventory I was very keen to get my hands on a pair of their most recent additions.

Yamaha already market two powered 10-inch models — the DBR10 and the DXR10 — and I own and regularly use a number of these little workhorses for all kinds of live work, both as mains and monitors. They don't produce a 10-inch model in their (previously) most expensive range, the DSR series, so the 'top dog' in the 10-inch format has hitherto been the DXR10 — currently at MkII, although mine are all the original type and any reference to them here should take account of this. I like 10-inch speakers

(good ones, anyway), as they tend to have enough bottom-end clout to be used on their own for smaller events, and can also handle the middle and top end of larger systems when using subwoofers. A local sound company that I have worked with on outdoor gigs uses a high-end rig consisting of four 18-inch powered subs and just two 10-inch full-range cabinets, and they regularly put this up for outdoor carnival-type jobs.

The DZR range represents a ground-breaking step forward for portable powered units as, quite apart from their performance, the entire range is available, as an option, with Dante connectivity, in the form of RJ45-type Ethernet ports that not only carry control information but all the audio as well. Thus, in a nutshell, you can daisy-chain the entire PA system together without using a single audio cable between mixer and loudspeakers, provided you're using a compatible digital console. So there's the headline, but the other main features include: plywood cabinets, rotatable horn assembly, latest-generation FIR filtering with DSP crossover and 96kHz internal processing, and comprehensive access to DSP parameters via a clear LCD screen.

The DZR10 reviewed here is the smallest in the DZR range, with 12- and 15-inch two-way, full-range units available, as well as a three-way model with a 15-inch woofer and 8-inch mid-range driver, plus tweeter. There are no DZR-badged subwoofers, but the range includes and is designed to fully integrate with the DXSXLFL units, available in 15- or 18-inch form. Like the DZR range, the DXSXLFL subwoofers are also available with Dante compatibility, those models being distinguished in name by a 'D' suffix. Apart from the



Yamaha

DZR10-D **\$1299**
& DXS15XLF-D **\$1549**

PROS

- Very detailed sound with strong mid-range delivery.
- The DZR10's low-end extension is impressive for its size.
- Editable parameters within a comprehensive DSP.
- Solid build and quality finish.
- Optional Dante connectivity.

CONS

- Slightly less hyped voicing on the DZR10 may take a few minutes to get used to.

SUMMARY

These are powerful, classy and compact speakers, and the Dante options make rigging them as part of a digital system a breeze.



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» digital connectivity and control, the D models are exactly the same in terms of construction and audio performance as the non-Dante versions. I like the idea of Dante being a separate option as, although it may come to pass one day, certainly not everyone is going to need this option within their system, and with this range there's no need to pay for it unnecessarily. For this review, Yamaha provided a pair of the DZR10-D speakers, complete with covers, together with a single DXS15XLF-D subwoofer.

First Look

Leaving aside the digital feature set for the moment, the first impression I gained when unpacking and handling the DZR speakers was that they were solid. They look great in a typical Yamaha way — slightly larger than my DXR10 boxes but still nice and compact, and with an attractive foam-faced front grille in black. Having decent metal front grilles is important for major physical protection of the drivers, and the foam layer provides additional protection against dust and moisture, but having the foam on the outside gives a classy look and in my experience tends to retain its looks more easily than painted steel, especially when speakers spend much time — as mine do — used as floor monitors. The foam actually feels pretty tough and I must admit to snagging one of the speakers on a flight case catch, but leaving no visible mark that I could see. The cabinet material is plywood and is finished in a black poly-coating. It feels like it would withstand a few knocks without getting untidy, and the large handles on the top and on one side have a good, metal-backed recess that has plenty of room to accommodate your average hand. I like the horizontal side handle, as I find this makes pole- or stand-mounting easier, although lifting a DZR10 up to stand height is definitely a two-handed operation as my initial observation of 'solid' translates into a manageable but noticeable 17.9kg (for the D model), compared to the DXR10, which is 4kg lighter.

Control for the DZR10 is via a large panel set into the back of the cabinet and dominated by a large LCD screen at the top, used for reading and setting all parameters in conjunction with a single adjacent data wheel. The basic input/output section is pretty much as you'd expect, with two input channels accessed

through combi XLR/TRS connectors and an old-school analogue level control for each. Output XLRs are provided for direct or post-DSP linking, and (apart from the Dante panel on the D models) that's all you see. A pull-proof IEC power connector and nicely recessed on/off switch are at the bottom, and two ventilation grilles allow convection cooling, which I assume will lead to less or slower fan assistance — I certainly didn't hear any fan noise when the boxes were at idle.

The LCD and data wheel allow adjustment of the various DSP functions, and a 'turn and press' will get you to where you need to be. It took me a few minutes to get used to when I should turn the knob and when I should press to select, but no matter where you get to there is always the friendly back/home button that steps back to the previous screen or wakes the display up if it has gone to sleep. For a detailed explanation of all the available menus and exactly what can be set up it's best to go online and look at or download the full user manual. Suffice it to say that just about everything you could want to set up is accessible here, and the level of detail is impressive: for example, you don't just select preset EQ curves, you can build your own exactly as you need, and even the audio output routing can be controlled to your liking. One thing that is worth noting when making parameter adjustments that directly affect the sound is that changes made using the encoder knob are not applied until the knob is pressed to execute them. This is a great safety feature in that you have to deliberately choose to apply the changes, but it also means that you can't hear the adjustments as you tweak, so sometimes it's a case of trying various settings until perfection is achieved. At the point of perfection for any given application, the DZR speakers have the ability to save your settings to a USB stick connected directly to the panel. This is not only useful for saving settings for a particular venue or band, but also for setting up several speakers — just set up one, and then copy the setup on to the others in the rig.

First Listen

Having selected a basic 'EQ off' preset I fired up the DZR10s in the studio and played some of my favourite tracks through them. My impression of the sound was just like when I unpacked



DZR-series speakers all feature two inputs and outputs, onboard DSP control, and a USB port for saving and recalling settings. The D-suffix versions also have the Dante panel below, which sports two RJ45 network sockets for conveying both audio and control data.

them, the word 'solid' again coming to mind. The DZR10s had a very centred, focused sound with a strong mid-range quite unlike most speakers of this size I've used, and certainly quite different from my DXR and DBR boxes. I turned them up, and down, and played everything from AC/DC to Boccherini, and couldn't make up my mind if I liked them or not.

Resisting the temptation to hook up the subwoofer just yet, I decided to run a direct comparison against my DXR10s, so I set up one of each and ran a mono track through them. (I should make it clear again that my own DXR10s are not the current MkII version). These have been with me for some time now and

»

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Dante-equipped speakers can be given an ID using the menu. This tells upstream and downstream Dante equipment what role the speaker has been assigned, and it will pick up the appropriate output from a mixing console connected to the same network.

use subs, and I was very impressed by the detail and clarity — just like

» I've always considered them to have a good sound for smaller gigs and as stage monitors. With the two speakers side by side, I was able to identify some of the differences, and I also brought in a couple of experienced sound colleagues to validate (or contradict) my impressions. At first, we all thought the DXR sounded a bit fuller, and a bit brighter; the DZR10 definitely sounded 'dark' in comparison, and this was true at various volume levels. The DXR10 was what we were expecting to hear, and the different voicing of the DZR took a little while to get used to. But after about 10 minutes we started playing something smooth and jazzy, and someone said "listen to the brushes", at which point we began to appreciate the level of increased detail in the DZR sound — the snare offbeats were really in sharp focus and I could imagine the individual wire hairs touching the snare head, rather than just hearing a general hit on the backbeat. Having listened 'through the keyhole' we all began to hear more detail throughout the frequency range, and there's no doubt that the DZR speaker was delivering a level of resolution and crispness that brought vocals forward and revealed subtleties in the material that we definitely missed when switching back. One of the most impressive things was that this detail was not lost at higher levels, and the bass extension of this little cabinet is an impressive feature in its own right. I did run them up pretty high, and was very keen to get them out on a live event.

In The Wild

I used the DZR10s for a couple of jobs, one open air and one in a 200-seater community hall. The latter was vocals and acoustic instruments only, so I didn't

back at base — that made mixing vocals an easier job than I'd expected in this less-than-ideal venue.

The outdoor job (nearly cancelled but for a fortunate weather window) required subwoofer deployment, so I took along the single DXS15XLF to handle the low stuff. I'd have preferred a pair of subs, if only to mount the top speakers on, but one was all I had so I positioned it in front of the stage about two-thirds to the right, more or less in front of the bass player. Although all three Yamaha speakers were Dante models (and were driven from a Yamaha TF and Tio stagebox setup), I decided to use conventional cabling mainly because I didn't have time to make up enough digital cables to reach all the DZR units. This gig was not normally something I'd use just a pair of full-range 10-inch speakers for, nor just a single sub, but this little rig performed way above expectation. It delivered a lot more clean output than a system of this physical size really should — that very forward and detailed vocal mid-range was a much commented-upon feature of the sound, and the low end was both deep and very powerful, with great, even coverage. This DXR sub really produced the goods. I enjoyed running the sound for this event a lot more than I'd anticipated, and I was left with the impression that the DZR/DXS system would have run quite a lot louder if I'd wanted, as it didn't show any signs of fatigue covering around 30 x 40 metres of grass!

D-Day

One of the ground-breaking aspects of the DZR and DXS ranges is the option to have Dante connectivity. On the D-suffixed versions there is a Dante panel for incoming and onward digital

connection, and additional menu options for setting individual cabinet ID so that the correct signal is delivered to any particular speaker from your digital console or any point in a Dante network. Those who already use these setups will know exactly what this is all about, but in a very simple nutshell it means that you can wire up the entire rig — mains and monitors — using decent-quality Ethernet cables (and switches if necessary) that carry control data and all the audio as well. You set up the speakers from the back-panel menu, selecting options such as 'ST L' (for front-of-house left) or 'Aux' or 'Sub' and route the appropriate output to them. On Yamaha TF consoles you can use the 'Quick Config' setting, and the devices appear on the console's routing screen alongside your digital stageboxes. I tried this out in the studio and once I had figured out how to set the speaker ID, everything just worked straight away, with three Cat5 patch cables connecting the TF mixer directly to the two DZR10-Ds and the DXS15XLF sub. This has got to represent how things will be going in the future, and has obvious advantages both for installed systems and flexible portable setups. I have a few jobs coming up where I would now feel confident to use the live Dante setup, and I can see myself getting quite hooked on it!

With live sound work, it's obviously all about the sound itself, and the DZR and DXSXL speakers contain much innovative and useful technology — for example, the rotatable horn on the full-range units, and the fact that the subs can be operated in cardioid mode when used in pairs. The lasting impression is that they really have raised the bar in terms of the powerful, clean, detailed and very focused across-the-spectrum output from portable boxes. I only got to try the smallest units in the range (I really want to hear the three-way units!) but I have been very impressed by their performance and would be more than happy to put them up for some challenging applications.

There would be so much more to say about the DZR speakers here if space allowed, but by far the best way is to get your own ears on some and see what you think. **///**

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Thermionic Culture Snow Petrel



Dual-channel Microphone Preamplifier

Thermionic's latest valve preamp offers bags of gain and plenty of analogue warmth. Is this the perfect partner for your ribbon mics?

HUGH ROBJOHNS

Thermionic Culture's chief designer Vic Keary has a fascinating reputation for using valves and circuit topologies that aren't conventionally associated with high-quality audio, yet which always deliver the goods, sonically and technically. His latest design, the Snow Petrel, is a two-channel valve mic preamp. It's a 'high-gain' design with very low noise and distortion, intended primarily for use with low-output ribbon mics.

Vic told me that the inspiration for the Snow Petrel lay in a previous career — he enjoyed using the classic Coles 4038s for brass and drums but found that most mic amps and mixers of the time had too much noise and distortion to enable their use on quieter instruments. So the Snow Petrel was designed specifically to enable low-output mics like the 4038 to be used on quiet sources. That sounds simple enough, but Vic has high standards, and it took him five

years, two unsuccessful previous versions, and a rather unusual circuit design to develop it to his satisfaction!

Of course, the Snow Petrel is not a one-trick Antarctic-gull: it is equipped with both switchable phantom power and input attenuators, and it can work with dynamics and high-output capacitor mics in a wide range of different applications just as competently as it can with passive ribbons.

The Tour

A hefty 2U rackmounting device, the Snow Petrel weighs 6kg and extends 290mm behind the rack ears. It consumes 30W of power but the case is well ventilated at the sides, rear and top, so there's no requirement for a ventilation space above the unit. Rear-panel connections are minimal: two pairs of XLRs for the transformer-balanced mic inputs and line outputs, plus a standard IEC (C14) mains inlet with integrated fuse holder. There's

also a voltage selector switch for 115 or 230 V AC mains supplies.

The visual styling is typical of Thermionic Culture, with the front panel carrying crisp white legends on a glossy black-painted background. Five chicken-head rotary controls and four toggle switches are provided for each channel, and another chunky toggle switch with a large green status lamp switches the unit on/off.

Another pair of toggle switches, each with red status LED, engage each channel's phantom power. These 'secure' switches have to be pulled out before they can be moved, preventing accidental operation — a feature intended to protect delicate ribbon mics. They won't help if phantom was left on accidentally in a previous session, of course, but I spent an entire BBC career 'hot-plugging' Coles 4038 ribbons (phantom power is present permanently on most BBC studio wall-boxes) without them suffering any damage!

The rest of the controls are arrayed



towards the left of the front panel, with channel one's controls directly above channel two's. The first toggle switch introduces a -20dB pad before the custom-designed Sowter input transformer in the signal path, so it avoids unwanted

Thermionic Culture Snow Petrel \$3299

PROS

- Fundamentally clean, but richly analogue and involving sound character.
- Massive 75dB of gain available with a remarkably low noise floor.
- Well-judged tone-shaping options to compensate for typical ribbon mic characteristics.
- Input and output attenuators enable precise control of intended overdrive effects.
- Excellent build quality.

CONS

- No line or instrument input options.

SUMMARY

A very elegant and carefully designed mic preamp specifically intended for low-output ribbon mics — but with enough versatility to be useful with any mic. It offers a range of appealingly analogue sound characters from crisp and clean, to powerfully raunchy.

transformer saturation, as well as preventing overloads of the input stage when using high-output mics and high-level sources. (The user manual recommends using the high-impedance input mode whenever the input Pad is used.)

Controlling that input impedance is a second toggle switch, which alters the transformer's primary wiring configuration to provide alternate Lo/Hi input impedance options of 850Ω or 2.4kΩ. The user manual claims that this low-impedance setting is actually "high compared with some other mic amps". That might be true in comparison with some vintage preamp designs, but the days of 50, 300, or 600 Ω input impedances are long gone, and 850Ω is a low input impedance by modern (voltage-matching) standards. The high-impedance mode isn't particularly high either — standard mic inputs now typically range between 1.5 and 2.5 kΩ. So the high-impedance mode really is just normal; it would need to be above 5kΩ before it could be considered to present a higher-than-normal input impedance.

Moreover, most modern mic preamps specifically intended for use with ribbon mics offer a far higher input impedance to minimise the load on passive ribbon mics.

An input impedance of 15kΩ is common in this application, and some are as high as 30kΩ. Still, the different impedance options will undoubtedly change the transient performance and overall tonality of passive moving-coil and ribbon mics.

Returning to the other operational controls, a red chicken-head knob selects the preamp gain in 7dB increments. The markings indicate 40, 47, 54, 61, 68 dB and 'Max' — this last providing 75dB, and the measured performance using an Audio Precision test set was always within 0.5dB and often within 0.25dB of the stated gain. These nominal gains are only correct when the 'Lo' impedance setting is employed, though, because the adjustable impedance is achieved by altering the transformer's configuration, and the different turns-ratio inherently results in a different voltage gain across the transformer. As a result, the preamp gain is 6dB lower than marked at each setting in the Hi impedance mode.

I'll skip briefly over the next two tone-adjusting controls, as the fourth and fifth chicken-head knobs also affect the signal level. The first is a variable Output Trim for the valve stage driving the Sowter 1:1 output transformer, and its 0 to -10 dB range allows precise level matching when,

»



The case is ventilated at the top, sides and rear, so there's no need to leave a space above this preamp when rackmounted.

» for example, working with a stereo mic array. It can also be used to keep a check on the output levels when the front end is being deliberately overdriven to introduce gentle second-harmonic distortion. Amusingly, or confusingly (depending on your point of view) the control's markings don't coincide accurately with the decibels of attenuation being applied, except at the two extreme ends of the range.

The last rotary control, a three-position Output Attenuator switch, affects the signal after the output transformer. Nominal attenuations of 0, -7 or -15 dB are available, allowing the output transformer to be driven for saturation effects, while maintaining a sensible output level for subsequent equipment. The attenuation varies slightly depending on the connected destination's input impedance, but the figures above are accurate for a typical 10kΩ load. This attenuator is an essential facility, as the Snow Petrel is capable of delivering over +26dBu when running flat out — that's more than most converters, interfaces, or mixing consoles are comfortable receiving! The final toggle switch inverts the polarity of the output.

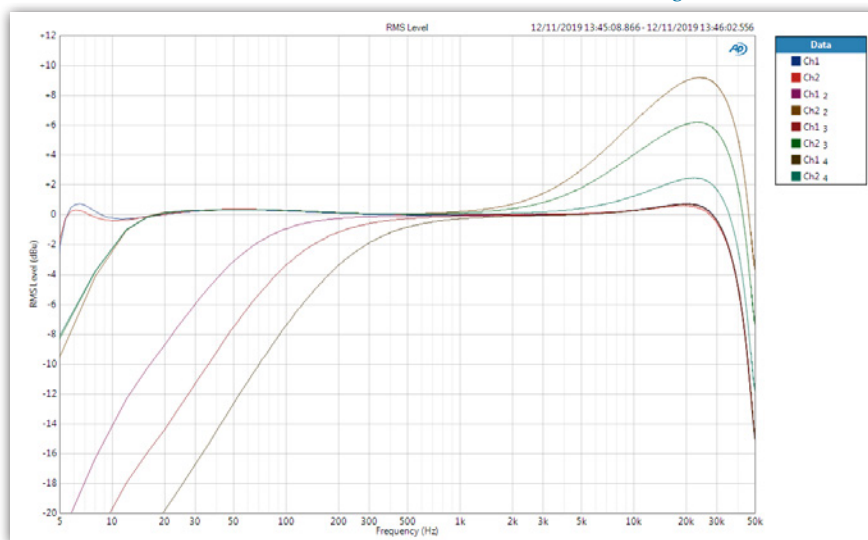
Returning to the two tone controls I skipped past, both are entirely passive and designed to compensate for common characteristics of vintage ribbons. The first (white) knob introduces a first-order high-pass filter with selectable corner frequencies of 60, 120, or 240 Hz. With a slope of only 6dB/octave, this filter is perfect for correcting the proximity-effect bass tip-up associated with bidirectional (fig-8) ribbon mics. (It's not much use in removing unwanted subsonic noises; that generally requires a much steeper slope.)

The second (blue) knob, Air, controls a variable peaking HF boost EQ centred at 22kHz. At maximum, the high-frequency boost reaches +5dB at 10kHz and +7dB at 22kHz, and this can be used to compensate for the HF roll-off associated with vintage 'long ribbon' microphones.

Technology

The mic signal is boosted by the input transformer's high 1:20 ratio into the first gain-stage valve — a Radio Technique RTC5654, which was used very successfully,

A set of response curves measured with an Audio Precision test set to illustrate the Snow Petrel's extended frequency response, along with the effect of the high-pass filter and a range of different Air boost settings.



in Thermionic Culture's Rooster 2. It's a rugged, wide-band RF pentode, featuring distinctive round holes in its anode plates, revealing the control grids within. Apparently the 5654 is a 'special quality' version of the Western Electric 6AK5W or Mullard M8100. The manual lists the latter as a suitable alternative, but Vic estimates stocks of the 5654 to exceed 30,000.

The second half of the gain stage is built around a more familiar, high-gain pentode: the EF86. The Svetlana version is used, but the JJ EF806 is listed as a suitable equivalent. Both these front-end valves are strapped to operate as triodes, with anode voltages of around 240V DC, and the gain is controlled by altering the negative feedback around the pair. Driving the Sowter output transformer is a Tung-Sol 6189, which is a high-output version equivalent of the low-microphony variants of the ECC82 or 12AU7A double-triodes.

The combination of these three valves, along with the step-up gain of the input transformer, permits the unusually high (for a valve preamp) 75dB maximum gain, while maintaining an impressively quiet noise floor and low distortion. The review unit's QC test report cites figures of 0.23 percent THD at maximum gain, with a maximum output level of +21.9dBu, and noise floor at -83dBu. At lower gains, the noise floor falls well below -100dBu, distortion below 0.025 percent, and the maximum output level rises to +26.4dBu. These are impressive figures. Crosstalk varies with gain, as you might expect, but is typically around -70dB at 10kHz. The worst case (-41dB at 10kHz, at maximum gain) is still better than most vinyl pickup cartridges!

In Use

I tried the Snow Petrel with both vintage Coles 4038 and modern AEA R92 passive ribbon mics and a selection of popular capacitor mics, and with a variety of sources and instruments. At just 0.5mV/Pa, the 4038 is about 10dB less sensitive than the R92, but the Snow Petrel was easily able to provide sufficient gain to allow the former

Alternatives

There are several preamps designed for use with ribbon mics, such as AEA's TRP and RPQ and the Integer Audio RMP2. But these are solid-state designs; I know of no other all-valve preamps intended primarily for use with ribbon mics.

to be used on quiet sources, including a cello, without getting lost in noise.

That Keary designed the Snow Petrel specifically for use with 4038s is evident in its creamy-smooth but detailed and naturally transparent sound. Dialling in some Air brings out a beautiful clarity and shine, without any hint of the edginess that's common with capacitor mics, and the high-pass filter cleaned up low-end muddiness nicely when close-miking. The Air control's boost range is very well-judged, bestowing the perfect working range with fine resolution. The turnover frequencies for the high-pass filter are also well-chosen and very effective in taming proximity boosts without sacrificing low-end weight. Similarly attractive sound qualities were obtained with the AEA mic, and while the impedance options did affect the tonality of both mics, I found that I selected different settings in different situations with no clear overall preference.

This preamp's noise floor is impressively low and entirely benign at all gains, but particularly so in the mid-range settings. I had no overload problems when using high-output capacitor mics, either, although I did need to engage the input pad on occasion (that's why it's there!).

In normal use, the Snow Petrel is very clean and neutral but certainly not sterile-sounding, and it produces a lovely clean, but rich and involving analogue sound. If desired, it can also be deliberately pushed into an overdriven character, which is gloriously musical and highly appealing in the right circumstances — although the output attenuator will be needed to avoid overloading downstream equipment. With so much gain available these overdriven effects are still possible when using low-output 4038s, too.

The Snow Petrel doesn't have a dedicated line input, and the input pad is not sufficient to allow use with normal line-level signals as the maximum input before clipping is just +7dBu, even with the pad engaged. This is a shame; it would be great if the preamp could be used for character when mixing, and I'd love to see a version with a dedicated line input. It would also be great to see direct instrument inputs, but Thermionic Culture do make a separate DI box which could add this functionality if required.

Conclusion

Overall, then, Thermionic Culture's Snow Petrel is a really nice, well-engineered preamp, with a clean yet involving sound character, unusually high gain, blissfully low noise floors, and well-judged tone-shaping. It's ideally matched to classic ribbon mics, and in partnership with the Coles 4038 it is capable of some truly magical and impressive results. It's also a very good choice for use with other popular dynamic mics like the Shure SM7B or EV RE20, and can also bring some warmth to the more clinical-sounding capacitor mics. I'm very impressed

with the versatility and quality of the Snow Petrel, and highly recommend auditioning it. **///**

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Nonlinear Labs C15

GORDON REID

Stephan Schmitt was the founder of Native Instruments and its Chief Technical Officer for many years until he realised that he had lost his passion for what NI was doing and moved on to establish Nonlinear Labs. The company's first keyboard, the C15, is a digital polysynth delivered in two sections that you connect using the supplied brackets and cable. The Base Unit contains its five-octave keyboard, a pitch-bender, two long ribbons that you can use as controllers and as data entry devices, a small control panel with an OLED display, plus volume controls for the main and headphone outputs. The Panel Unit contains four panels of 24 parameter selectors (96 buttons to access a rather larger number of parameters) plus a central editing section comprising a further 18 buttons, a rotary encoder and a larger OLED screen. In a novel twist, it's possible to play the C15 without the Panel Unit attached, using the Base Unit as a preset

Polyphonic Synthesizer

The Nonlinear Labs C15 is that rarest of things — a synth that dares to be different.

synth that you can also program using its browser-based editor/librarian (see box). But if you attempt to edit the C15 before doing any homework, you'll probably end up with little more than a selection of weird

noises that will do nothing to enhance your reputation as a sound designer. This is because its 'Phase 22' phase modulation synthesis engine (which is related to, but a significant advance upon, the Kontour engine that Nonlinear Labs developed for NI) eschews common digital synthesis



The Base and Panel units can be separated and the Base unit played as an independent, preset synth.



techniques such as PCM playback and virtual analogue modelling and instead incorporates elements of 2-op FM and its sibling Phase Distortion synthesis, while not being the same as either of them.

Making Noises

There are two initial sound generators for each of the C15's 20 voices. These are called Branch A and Branch B, and each comprises an Oscillator and a Shaper. The Oscillator generates a sine wave whose pitch is controlled by a tuning parameter, an Envelope and key tracking. The initial phase is also programmable, ranging from -180° to $+180^\circ$. However, there are three ways — all controlled by Envelopes — in which you can modulate this wave to obtain complex spectra. The first is by feeding the output from the Oscillator back to modulate itself or by applying the output from its Shaper as the modulator, or a mix of both. The second is cross-modulation derived from the output of the Oscillator or the Shaper of the other Branch, or a mix of both. The third is the modulation generated by a feedback loop tapped from further down the signal path. If this wasn't complex enough, each oscillator offers a parameter called Fluctuation that

applies a random frequency offset to each cycle of the oscillation, creating anything ranging from an unusual smearing of the sound at low values to wide-band noise at higher ones. The amount of Fluctuation can also be controlled by an Envelope, which makes all manner of strange effects possible. A low-pass filter called Chirp lies in the summed modulation path and removes some of the more aggressive consequences of high modulation amounts.

Like the Oscillators, the Shapers generate sine waves that can be modified in various ways. Most obviously, you can overdrive a Shaper to create clipped waveforms with increasing numbers and amplitudes of harmonics, while Fold and Asymmetry (that's not my spelling, I promise) further modify its output in interesting ways. There are then three output level controls within each Shaper panel, and these determine what the Branch sends to the next stage in the audio path. The first mixes the proportions of unshaped and shaped signals within the Branch, while the second (controlled by an Envelope) mixes the output from the first mixer with another tap of the feedback signal before passing the audio to the inputs of the Ring Modulators that lie in each of the Branches. The third control determines

the mix between the input and output of the local Ring Modulator, and this is the signal that is passed down the audio path.

If there were nothing more to the C15, it would still be a powerful synthesizer, but Nonlinear Labs have added two filter sections to shape the audio yet further. The first is called Comb Filter, but offers three functions. Its first parameter determines the mix of Branch A and Branch B accepted at the input of the comb filter itself, while the next sets the initial pitch of the first notch in its spectrum. This is followed by a 2-pole all-pass filter that affects the phases of the signal components passing through it, with Tune and Resonance parameters determining the shape of the phase-shifting curve. The third is a high-cut filter that tames the high frequencies in the resulting signals. This section also contains an internal feedback path that can turn it into a resonator excited by the signals generated by the Branches. Its Decay parameter helps to determine the nature of the resulting signal and how long it takes to decay when the keyboard gate is removed, while the Gate parameter determines whether its oscillation decays naturally or is curtailed by a shorter release. All three filters can be modified by key tracking and Envelope C, while the Comb Filter



I/O

The C15 boasts surprisingly few connections. There's a headphone output on the front of the Base Unit, while the I/O on its rear consists of a stereo pair of balanced/unbalanced line-level audio outputs, four sockets for pedal switches, expression pedals

and other suitable controllers, plus a USB A socket for memory backup and updating the OS. Two further sockets allow you to connect the Base and Panel Units together using the supplied ribbon cable. The final hole is for its power supply. Unfortunately,

this is another external AC/DC converter connected using a thin cable terminated with a barrel plug, and there's no stress relief. This is not a suitable way to power professional equipment and it's time that manufacturers stopped doing it.



» can also be modulated by the outputs of Branches A and B.

The second filter section is a State Variable Filter that you can place in parallel with the Comb Filter section, or in series, or in a mix of both. It comprises two 12dB/octave filters whose configurations

range from LP>LP, though BP>LP, HP>LP, HP>BP and finally HP>HP. The cutoff/centre frequencies of the two filters can be separated by a Spread parameter that provides all manner of additional profiles, while the Parallel parameter allows you to configure the two filters in series or in parallel, or anywhere between the two. You can determine the initial cut-off/centre frequency and resonance of the pair, and control these using key tracking and Envelope C. Filter FM is again possible using the outputs from the Branches.

The audio now reaches the Output Mixer, which accepts signals from Branches A and B plus the two filter sections, with individual pans for each. There's a master Level control, and Key Panning allows you to spread the output with low-pitched notes to the left and high ones to the right. If this seems straightforward, it isn't, because the Output Mixer contains another Shaper whose Drive, Fold and Asymetry parameters perform the same actions here as they do in the Branches.

Further Noises

Next come five effects sections in a predetermined series. The first is a stereo Flanger. The delay that creates the effect can be modulated by contributions from its internal oscillator and Decay contour, with parameters to determine the initial delay time, the modulation depth and (if desired) an offset in the delay times in the left and right channels. The signal then passes to another all-pass filter that creates phasing effects and the output from this can then be fed back to the flanger's input with either positive or negative polarity, and (again if desired) crossed so that the output from the right channel is fed back into the input of the left, and vice-versa.

A high-cut filter then attenuates some of the higher frequencies in the effected signal, and a wet/dry mix parameter does what you might expect, but with the option of inverting the delayed signal for yet another flavour of effect.

The audio now passes to the Cabinet effect, which is yet another Shaper with an additional parameter that allows you to bias the shaping of the audio toward high or low frequencies. It also includes a band-pass filter with independent high-cut and low-cut controls, plus a saturation stage that adds yet more distortion.

Next, the Gap Filter comprises independent low-pass/high-pass filter pairs for the left and right channels. You can control the centre frequencies for these, the offset between the channels, the gap between the high-pass and low-pass cutoff frequencies, and the resonances of all four filters. There's also a Balance control that emphasises the low or the high frequencies of the effected signal. The Mix control then determines the wet/dry mix, with positive values configuring the filters in parallel, and »

Nonlinear Labs C15 €3361

PROS

- It introduces a powerful sound engine capable of generating remarkable new sounds.
- The choice of keyboard action makes it a pleasure to play.
- The designers really cared about the look and feel.
- Editing is possible from any connected browser.
- There's an excellent flight case for the synth and its accessories.
- The manufacturer offers innovative purchasing and returns plans.
- There's a significant roadmap of future enhancements.

CONS

- It's complex and will take time and effort to master.
- Parameter-access programming systems have long been superseded.
- There are too few Oscillators or Shapers to create some of the sounds that I attempted.
- It offers no dedicated LFOs.
- There are some unfinished features.
- It lacks MIDI.
- Another flimsy wall-wart!

SUMMARY

The C15 is a powerful but eccentric polysynth that does some things very well and other things not at all. Some people will love it for its radical and sometimes unique sounds, but for many it will remain a curio if only because its lack of MIDI will mean that they can't fit it into the ways that they compose and play music.

Sound Management

Although the patch memory system is divided into banks and presets, there are no fixed numbers here; you can create as many banks as you like, and place in each as many patches as you choose. This is great if you work on multiple projects simultaneously and want to create sound banks for each when you have no prior knowledge about how many patches you'll need. Once stored, you can move patches within and between banks, and search for specific sounds using the keywords and hashtags that you can save within each. In the absence of MIDI, all backing up and restoring of patch banks is carried out using USB memory sticks.



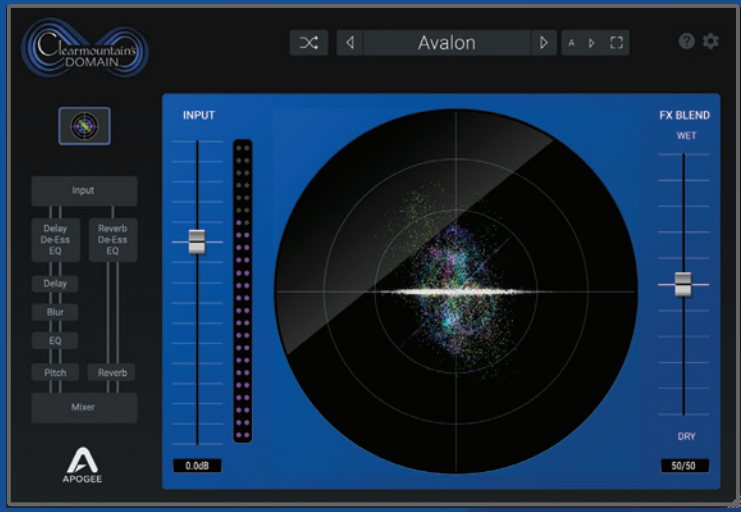
Bob Clearmountain's Personalized FX Signal Chain Recreated in a Simple, Powerful Plugin

Where Your Mix Lives™

The Clearmountain's Domain plugin by Apogee reproduces Bob's personalized FX signal chain for creating the cohesive spaces, expansive dimensions, and rich atmospheres where his mixes live. Clearmountain's Classic presets help you to recreate the sonic environment of your favorite Clearmountain mixes. Advanced views reveal Clearmountain's decades of expertise and empower you to create your own musical mix domain.

Highlights

- Clearmountain's Personalized FX Signal Flow
- Classic Presets based on Clearmountain mixes
- Clearmountain Spaces, Convolution Reverbs
- Real Time FX Visualizer



Simple home screen with FX Visualizer makes it easy to find a unique sound using Clearmountain's Classic presets.

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The Editor/Librarian

Nowadays, you can control most synthesizers using editors that transmit and receive MIDI CCs and NRPNs. The C15 adopts a different approach, making the synth's editing system and patch librarian accessible via a GUI that you can view, zoom and edit using any browser connected to the synth over Wi-Fi. I launched Safari on my iPad, connected it to the C15's network, typed in the address and, within seconds, the GUI appeared and I could edit the synth using the touchscreen. The same exercise on my MacBook Pro required a WPA2 password but, once this was entered, the GUI appeared as before. The larger screen on the laptop was a boon because there's a lot contained within the GUI, so the more screen space available, the better. (If I were to make a C15 the centrepiece of my studio or program it extensively for live use, I would buy a huge monitor to edit it.) At some point, I noticed that I had two devices connected to the C15 and that I could use both simultaneously. Once I had started programming the C15 this way, I realised that I would never choose to use the Panel Unit again.

The State Variable Filter allows you to create a much wider range of cut-off profiles than you might imagine.



Each of the two oscillators generates just a sine wave, but there are many ways that this can be bent into fiendishly complex waveforms.



» negative values configuring them in series. Nothing in the C15 is as simple as it seems!

The signal now reaches the Echo and Reverb sections. You can set the delay time for the Echo and determine an offset for the left and right channels for yet more stereo effects. There are controls for feedback, for cross-feedback from right to left and vice-versa, for a damping filter, and for the wet/dry mix. The maximum gain of the feedback loops is a tad under unity which means that, while the echoes can last for a long time, wacky sci-fi effects are beyond it. The output from this is then passed to the surprisingly good Reverb which offers control over pre-delay and reverb time, low-frequency damping and a Chorus effect that modulates the delay times of the reverberated signals. The audio then passes through a stereo master volume and tuning stage that includes a soft clipper to tame things if the signal gets a bit too wild.

At this point, you may want to breathe a huge sigh of relief at having reached the end of the C15's complexities, but you can't because you haven't. This is because, as already mentioned, you can feed the mixed outputs from the two filter sections and the effects (with a programmable contribution from the Reverb, or not, as you choose) back into the modulation and mix inputs of both Branches. Of course, this wouldn't be the C15 without yet another Shaper integrated into the Feedback Mixer, offering the now usual Drive, Fold and Asymetry controls. You can also set the base level of the signal in the feedback loop and use key tracking to bias this toward low or high frequencies. Inevitably, the choice of positions from which you derive the feedback signal and the relative amplitudes

of the components can dramatically alter the sound that you hear.

Next, we come to a Unison section that can stack up to 12 voices per note with detuning of up to 120 semitones. You can also spread the start phases of the stacked voices to create phase cancellations and you can spread their positions in the stereo field for an expansive sound. Finally, we reach the Scale section, which acts on a per-patch basis and allows you to determine an offset of up to eight semitones for each note in the octave.

Throughout this description, I've referred to the C15's Envelopes. There are three of these, called A, B and C, with the first two acting upon parameters within Branches A and B, while the third controls the filter sections as well as the Feedback levels. They each offer five stages — Attack, Decay 1, [Breakpoint], Decay 2, Sustain, and Release — and allow you to control the Attack time and level as well as the Release time using velocity, while a Curve parameter allows you to shape the Attack from logarithmic to exponential responses. The durations of the Attack and Decay stages of all three Envelopes can range from near instantaneous to 16 seconds, while their Releases can range from near instantaneous to infinite, so sounds can be sustained indefinitely. You can control each contour's amplitude using velocity and key tracking, and shorten its duration as you play higher up the keyboard, which is vital for emulating many natural sounds. Envelopes A and B offer an overall gain control that Envelope C lacks, but C's Breakpoint and Sustain levels can be negative, so it can generate a much wider range of shapes.

The C15 also offers a small modulation matrix that allows you to direct your choice of eight hardware controllers — its four pedal inputs, the bender, the dual ribbons and channel aftertouch — to four programmable macros that can be directed to any combination of 86 possible destinations. I had some problems with the pedals during the review, but the manual admits that this section is "still in development and proper functionality is not guaranteed for now", so I'm not going to make a fuss.

In Use

I love the look of the C15 (which recalls the iconic Synclavier II ORK) when I'm playing it, although the view from the side is less attractive, and I like the fact that you can choose from four woods to best fit the look of your studio or stage rig. Playing it is also a pleasure; while I would have liked the Base Unit to incorporate pitch-bend and modulation wheels, the case, keyboard, ribbons and pitch stick all give the impression that somebody really cared about how the instrument would look and feel. Whether the programming system accessed via the Panel Unit will be widely liked is a different matter. The idea of stepping through related parameters using a button and then editing the chosen one using a single encoder was quite neat in the 1980s, but is much less so in 2020.

Whatever players think about editing the C15, I suspect that most will merely stumble across interesting sounds because trying to program complex timbres deterministically often feels like a job for a mathematician. There may be a small number of modules within

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» Phase 22, but there's a huge amount of complexity and interaction available, and in addition to everything described here there are numerous facilities — parameter smoothing, patch morphing, undo trees, additional pages in the editor, and more — that I've had no room to discuss. Furthermore, patches will often collapse into noise if you don't keep a rein on things. But if you take the time to understand the system, learning (for example) how to obtain waveforms such as sawtooth and square waves, you can program strings, brass and some surprisingly organic patches alongside the myriad percussive, plucked, hammered, distorted, and twisty-noisy sounds at which the C15 excels. I even coaxed some pleasing lead synth patches from it, although the lack of portamento was a limitation. On the other hand, if you're looking for lush sounds or patches with complex modulation of tone and loudness, you can forget it. That's not what the C15 was designed for, and no amount of persuasion will cajole it into producing them.

You could make a case, therefore, that the C15 is an expensive way to obtain a constrained palette of sounds, but I don't accept that argument. Some people will pay just as much for a second-hand Minimoog that they use for one thing, and the Prophet 5's inability to produce the sounds of a Logan string ensemble doesn't make it a bad synth. Perhaps a better observation would be that, for a similar price, instruments such as the Kronos and Montage can sound similar to the C15 and do much more in addition. Nonetheless, I suspect that you'll obtain sounds from the C15 that you've not heard before. The results may not always be musical, but they are often remarkable.

Despite its complexity, there are a number of other things about Phase 22 that I find restrictive. I can live without arpeggiators and sequencers, but I'm much more concerned by the fact that you can't shape the amplitude and tone of a given Branch independently, and that you can't use the Envelopes to shape the amplitudes of the audio signals presented to the output mixer or final output stages. Furthermore, the lack of conventional LFOs means that you have to use a Branch Shaper to create simple modulation effects such as vibrato, which often means that you have just one Branch available for signal generation. As for tremolo... forget it; the C15 can't create

Buying a C15

Nonlinear Labs is also innovative in the ways that it will sell the C15 to you. You can buy it outright in the conventional fashion or pay a minimum of 2 percent per month, interest-free, so that you own it after 50 months. If you choose not to keep it, you can return it at any time and no further payments are due, and the company will even refund any payments made above the minimum amount.

Similarly, if you decide not to keep one that you bought outright, you can return it for a partial refund calculated on the same basis. The price of the C15 itself is €4000 including VAT, and a superb custom-made flight case is available for an additional €300. Nonlinear Labs will even include two Roland EV-5 expression pedals and a Roland DP10 damper pedal at cost.

it. I'm relieved, therefore, that Stephan Schmitt has confirmed that many of these shortcomings will be addressed in future upgrades.

But the huge — and I mean *huge* — omission is the lack of MIDI. I understand the philosophy behind this; if you can't sequence the C15, you're forced to play it. Indeed, Stephan Schmitt told me that, "With MIDI keyboards and expanders, the studio becomes the instrument. I wanted the instrument to be the instrument — sit down, switch it on, and play." But so much of our musical lives is now bound up in multitrack sequencing, beat synchronisation, repeatability and tweakability that it's hard to justify its omission. Sure, you can treat the C15 like an acoustic instrument and, if necessary, edit performances in the audio domain, but I can't see that incorporating MIDI would have deterred players who want to use it this way. If there's a silver lining it's that, when unconstrained by MIDI wordlengths, parameter and controller values can be of significantly higher resolution, so things like velocity and aftertouch can be defined with much greater precision.

The Future

Despite its current limitations, the C15 has great potential because it has been designed to be upgradeable, and not just with minor updates. Nowhere is this more apparent than on the Panel Unit's control surface; its magnetic inlays are removable and can be replaced by new ones with different legends. Should the promised new sound engines appear this means that you could, in effect, give the synth a new panel to reflect its changed facilities.

During the course of this review, Stephan Schmitt was kind enough to let me see the development roadmap and, even without the introduction of new synth engines, this includes proposed improvements to the voicing, sample rates and latency, the addition of splits and layers, a monophonic mode, additional

macros and destinations, extended parameter ranges, chord memory, improvements in the GUI, seamless patch changes, improved morphing, and more. It's an ambitious list, but one that should be within the range of the existing hardware and, therefore, of all of the instruments already sold. Other ideas under consideration include new models with extended keyboards, polyphonic aftertouch, and additional control capabilities. But however the C15 evolves, the Nonlinear Labs website promises that the company will keep existing instruments up to date, which is very much to the company's credit.

Conclusions

It's been a long time since I encountered a synthesizer that made me think, 'wow, that's different!', but there are many aspects of the C15 that have surprised me. Given everything that Stephan Schmitt has achieved in our industry over the past 25 years, I think that he has every right to design and build something eccentric that ticks all of his boxes, if not yours. Inevitably, it will be lambasted by people who don't want to learn a new way to create sounds and who will never be able to play well enough to fully utilise its expressive capabilities, but there's nothing in the rules to say that all synths should cater for all tastes; the C15 is a unique synthesizer that simply happens to be very strong in some areas and very weak in others. Sure, it's not cheap, but it has immense character, so a handful of players are going to love it from the outset. Others will consider buying one because it's so different from anything else that they own. But I suspect that most of us will stick with what we already know. It therefore remains to be seen whether there is a place in our lives for the C15. ■■■

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Modal Electronics Craft Synth 2.0

Monophonic Wavetable Synthesizer



Modal's revisited Craft Synth offers complex sound design in a compact package.

RORY DOW

In the world of software, major updates to instruments are commonplace; less so in the hardware world. Yet here we are with a second generation of the Modal Electronics Craft Synth. The original was a budget-conscious, snap-together DIY effort, resulting in a zany-looking two oscillator digital monosynth. The Craft Synth 2.0 moves away from the DIY approach with a more robust build and a matured synthesis engine, yet keeps the price almost as low as the original.

The synthesis engine under Craft Synth 2.0's hood is impressive. Eight wavetable oscillators, 40 morphable waveforms, multi-mode filter, frequency modulation, phase modulation, sync, ring modulation, waveshaping, waveshaping, unison,

three envelopes, two audio rate LFOs, arpeggiator, sequencer and effects. It is a lot to fit in a small space.

She's Crafty

And make no bones about it, the Craft Synth 2.0 is small. At 150 x 135 x 68 mm, it easily fits in one hand, yet manages to pack in a dozen encoders, a touch-sensitive keyboard and even proper MIDI DIN ports. A tubed section underneath holds three AA batteries and tilts the entire instrument forwards slightly, making it an ideal tabletop synth.

Basic operation is simple to grasp. Each of the dozen encoders has three functions it can control, with each function being labelled in a different colour underneath. On the left and right of the keyboard is a Shift and Preset key. Holding down one

**Modal Electronics
Craft Synth 2.0 \$149**

P R O S

- A very capable wavetable synthesizer.
- Backpack friendly.
- Excellent editor/librarian included for free.
- The price.

C O N S

- A reliance on the editor app for some basic functions means you'll be using it whether you want to or not.
- Despite having eight oscillators, it's monophonic.

S U M M A R Y

The Craft Synth 2.0 is a big upgrade from the original Craft Synth. The synthesis engine is hugely improved, bringing wavetables, FM, phase modulation, effects and more. It feels, in fact, like a completely new synthesizer. The price-to-synth ratio is high, but it is not without compromises.

of these whilst adjusting an encoder will access its alternative functions. In this way, you have direct access to 36 different parameters using 12 encoders. The Preset key is also used in conjunction with the eight touch-sensitive 'keys' to store and recall up to 64 presets arranged in eight banks of eight.

The touch keys can be used to play the synth too, and are flanked with buttons to increase or decrease the octave. With only eight keys, Modal decided that the best approach was to have them permanently locked to a root note and scale. There are 29 scales to choose from, with one slot reserved for a user-programmable scale, which requires the MODALapp editor. However, without any kind of screen, it is impossible to tell which key or scale you are selecting. Coupled with the fact that the keys are not velocity sensitive, it's difficult to see how the keys could be used for anything other than simple triggers during sound design. Any serious playing will need to be done via MIDI.

Above the eight keys are dedicated buttons for various functions. Alongside Shift and Preset, there are buttons for selecting which LFO or envelope will be controlled by the shared encoders, as well as an on/off switch for the arpeggiator/sequencer. Along with these more obvious functions, there are a reasonable number of hidden key combinations which can only be learned by reading the manual. For example, holding Shift and Preset buttons for four seconds allows you to set the synth's MIDI receive channel. I'm not a big fan of esoteric key commands like this. It is rare that one can remember them all, but given the size of the Craft Synth 2.0 and its relative complexity, it is difficult to see how they can be avoided. Worse still, some functions are only available by using MODALapp, an application available for OS X, iOS, Windows and Android (see the 'MODALapp' box).

The design is striking enough. The diagonal lines on the front panel give it a certain '80s feel, while the rounded white plastic put me in mind of early iPod designs. I do question making the knobs almost the same colour as the panel. In a dark studio, it is difficult to see the knobs at all. Some of the labelling can be hard to read in low light also.

Round the back, we find MIDI In and Out ports and a micro-USB connector for power and MIDI. There is also line out, headphone socket, and sync in and out, all on mini-jack sockets. I was particularly pleased to see the full-sized MIDI sockets, as many instruments



The big news on the Craft Synth 2.0's back panel is the full-size MIDI I/O ports. The accompanying sync I/O, line out and headphone ports are all on mini-jack sockets and there's a mini-USB port for power and computer connectivity.

at this price resort to mini-jacks for MIDI in order to save a few pennies.

The Sound Engine

On to the important stuff. The synthesis engine is comprised of two wavetable oscillators. There are nine wavetables available, each containing five waveforms, and both oscillators can use a different wavetable. The wavetables will morph smoothly from one waveform to another and you can automate this using a number of different modulation sources, after which you can add further interest by applying one of 16 different 'oscillator modifiers'.

Examples of oscillator modifiers are ring modulation, oscillator sync, downsampling, wave-shaping and frequency modulation (for

a full list, see the 'Oscillator Modifiers' box). They each work using just one control, which can also be automated using the modulation matrix. Some use cross-modulation between the two oscillators, whilst others use a simple effect like bit-crushing on one or both oscillators. Clearly, Modal have done a fair amount of work in this area, but the one downfall is that there is no way to know which Oscillator Modifier you are applying unless you either memorise the 16 options (and their relative positions in the list) or use the MODALapp for sound design. Unfortunately, this reliance on the app becomes a running theme throughout. Some users will be happy adjusting this by ear, but others may want to know exactly what they are listening to. »

Oscillator Modifiers

Here is a full list of the Oscillator Modifiers, which allow you to cross-modulate and effect the two oscillators in different ways.

1. **Phase Modulation:** Yamaha FM-style phase modulation where Wave 1 phase is modulated by Wave 2.
2. **Window Sync:** Wave 1 oscillator sync with internal sync pitch.
3. **Ring Modulation:** Classic ring mod, Wave 1 and 2 are multiplied together.
4. **Triangle Wavefolder:** Applies variable gain to Wave 1 which results in wave folding.
5. **De-Rez:** Reduces the sample-rate, for a more lo-fi sound.
6. **Rise-Over-Run Phaseshaper:** Modifies the phase and resulting waveform from Wave 1, allowing PWM-style sounds as the waveform is modified.
7. **Glitchy Sine Waveshaper:** A subtle bit-crushed sine-wave waveshaper applied to Wave 1.
8. **Window Amp Sync:** An alternative to Window Sync.
9. **Frequency Modulation: Wave 2** modulates the frequency of Wave 1. Similar to Phase Modulation, but more extreme.
10. **Hard Sync:** Classic hard sync as found on many synths. It uses an internal oscillator to sync Wave 1 to, rather than using Wave 2.
11. **Min Modulation:** A type of amplitude modulation. It compares the amplitude of Wave 1 and Wave 2 and outputs whichever is lowest.
12. **Sine Wavefolder:** Same as the Triangle Wavefolder, but wraps waveforms using a sinusoidal response instead.
13. **Bit-Crush:** Classic bitrate reduction. Great for 8-bit chip tune sounds.
14. **Scrunch Phaseshaper:** Messes with the way the wavetables are read.
15. **Lo-Fi Phaseshaper:** An alternative version of above.
16. **Vocalized Sync:** Combines Window Sync and Phaseshaping to create formant-like tones and strange sync sounds.

» Up to eight additional oscillators can be added using the Spread function, which will achieve unison, stacked octaves and chords. This is done with a single physical encoder, which makes selecting specific chords or intervals quite difficult. Once again, unless you have excellent hearing and pitch detection, it's off to the MODALapp to pick something specific. Considering the Craft Synth 2.0 can generate eight oscillators, I do wonder why Modal did not include a simple paraphonic mode where you could play all available voices via MIDI. This would broaden its appeal to a great extent I think. To be fair, there is the possibility to polychain multiple Craft Synths together and have them operate as a polyphonic whole (see the 'Polychain' box), but it would still be neat to get something close without buying multiple synths.

The filter is a state variable 2-pole design which can morph through low-pass, band-pass and high-pass. It doesn't self-resonate. It's a decent digital filter and the ability to morph between filter responses means you have many tonal options. The morph parameter is a destination in the modulation matrix too, which can add a wonderful sense of movement to a patch when assigned to an envelope or LFO. Talking of envelopes, there is a dedicated filter ADSR envelope with amount control along with one for amplitude and one which can be freely assigned through the modulation matrix. All envelope stages, plus the amount, are also available as modulation destinations.

To add further movement to your sounds, there are two LFOs. Each has four waveforms; sine, sawtooth, square and sample & hold, and you can morph seamlessly through them using a single

Alternatives

Wavetable synthesizers at this price are thin on the ground. The brand-new **Korg NTS-1** is a build-it-yourself synth which comes as a kit. The synth engine is a single monophonic voice taken from the Prologue and Minilogue XD, which means it can do wavetable synthesis, but retains the user-programmable oscillators of its older siblings. It is even smaller and cheaper than the Craft Synth 2.0. Alternatively, if you save a few more pennies, the **Arturia MicroFreak** (reviewed by Simon Sherbourne in the June 2019's SOS) will give you a larger synth with more hands-on control, a two-octave velocity-sensitive keyboard with polyphonic aftertouch and multiple synthesis types including wavetable, additive, karplus strong, superwave, FM and granular formant. It is a more fully featured (but less backpackable) synth, but will cost you more than double the price.

MODALapp

Connect the Craft Synth 2.0 to a computer or mobile device via USB and you can use MODALapp to control every aspect of the synth. Not only is there a fully featured patch editor and librarian, but firmware updates are done here also. MODALapp is free and available for OS X, iOS, Windows and Android. If you are lucky enough to own any other Modal synths, it will also perform the same functions for those too.

The app is comprehensive and rather well designed. The patch editor is logically laid out, and very easy to get to grips with. The patch librarian offers useful functions such as patch renaming (not possible on the synth itself) as

well as options to backup, reorder, initialise and randomise patches. Various settings are also on offer such as MIDI channel, MIDI filters, clock source, pitch-bend range and so on.

Some functions, which I would regard as essential to good sound design, like assigning velocity, mod wheel or aftertouch, can only be done using the app. This is rather a major flaw because it means that, regardless of your thoughts on using computer patch editors, you'll have to. I can't think of the last time I made a patch which didn't use either velocity or mod wheel. It is rather impossible, therefore, to finish a patch without loading up the editor.



control. Both LFOs can either be free running or sync'ed to master tempo, but LFO2 can also work at a division of the current note's frequency. I think this is the first time I've come across this feature in a synth (I suspect it has lineage in Modal's bigger synths) and it's very cool. It can produce audio-rate, note-dependant effects and it can almost sound like an additional oscillator at times. There are options to have the LFOs retrigger, free run or run through a single cycle, effectively turning them into an extra envelope.

To help you assign all this modulation, the Craft Synth 2.0 offers an eight-slot modulation matrix. Each slot can hold a single modulation source and destination. The list of modulation destinations is comprehensive. It includes obvious ones such as wavetable position, pitch, oscillator modifier, filter cutoff and resonance, etc. But it also includes other useful destinations such as all envelope stages, delay time and feedback, glide time, oscillator spread and

filter morph, which allows for some very creative patching. The modulation sources are sadly more restricted. There are eight fixed sources: LFO1, LFO2, mod envelope, note pitch, velocity, mod wheel, expression (CC11) and aftertouch. Only the first four in that list are available to assign from the synth itself. Velocity, modulation wheel, expression and aftertouch can only be assigned from the MODALapp. This is probably my biggest gripe with the Craft Synth 2.0. If you want to use the synth as a stand-alone instrument (without a computer), you'll have to learn to deal without velocity, modulation wheel or aftertouch. I suspect this will be a deal breaker for some.

On to the programmable arpeggiator/sequencer. A single on/off switch controls the arpeggiator. You might expect more options, but that's really it. It's a basic 'as played' pattern, meaning notes will be repeated in a 16th pattern in the order in which you hold them. There are no other patterns, no octave settings, no time



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MIDI Spec

When the Craft Synth 2.0 is plugged into a computer via USB it is fully class-compliant, so no drivers necessary. The touch-sensitive keyboard, arpeggiator and sequencer will send MIDI notes, and the encoders will send MIDI CC. Likewise, the synth will respond to incoming notes and MIDI CC so you can fully automate it from an external sequencer. A full list of the MIDI CCs that the synth responds to is available in the excellent manual. The sequencer, arpeggiator, delays and LFOs will sync to MIDI clock automatically if detected. The global MIDI channel can be set from the synth itself or from the MODALapp. You can also recall presets using MIDI Program Changes. Overall, Modal should be applauded for a good MIDI implementation, something which many electronic instruments in this price bracket don't manage.

» divisions and no repeats. There is, however, an SH-101-style sequencer. Hold the Arp/Seq button and input notes from either the touch-sensitive keyboard or MIDI. The sequencer will record the notes at 16th note intervals. You can insert rests too if desired. When you're done, hit a note and the sequence will play back from the beginning, transposed according to which note you play. Both the arpeggiator and sequencer can be clocked either to the internal tempo, MIDI clock or the analogue Sync input. The arpeggiator, it is fair to say, won't win any prizes, but a simple step sequencer is useful to have and can spark new ideas.

A distortion and delay effect are the final piece of the sonic puzzle. The distortion is a waveshaper, so can sound quite extreme unless used carefully. There is only a single control for amount. The delay is a digital delay with dry/wet, time and feedback controls. The delay can range from around 0.02 to 750 ms, making it useful for modulation effects such as chorus and flanging (you can automate the delay time

Polychain

Up to four Craft Synth 2.0s can be chained together to make a four-note polyphonic group. It works through the MIDI din connectors and requires each Craft Synth 2.0 to be attached via MIDI In and Out to the next in the chain. The master unit is at the front of the chain and, with a secret button combination, all the units will snap into the correct mode, allowing you to control all four from the master. I only had one review unit so didn't get to try it, but I'm having fun imagining the 32-oscillator mayhem that would be possible.



The Craft Synth 2.0's front panel measures just 150 x 135 mm, making it perfect for pockets, small desks and those little fold-out tables you get on the back of train seats.

in the modulation matrix), as well as normal delay duties. It can also sync to clock at various subdivisions of either internal, MIDI or Sync tempo.

Conclusion

There's a lot of fun to be had with the Craft Synth 2.0. The synthesis engine is surprisingly deep. If judging this synth purely by its sound design potential, it would get a resounding thumbs up. Wavetable synthesis is a gateway to all kinds of unique and interesting sounds that would have an analogue synth flummoxed. It seems to excel at sharp leads, cutting bass sounds, Detroit-style chord stabs and thick detuned unison sounds. There is plenty to like about the sounds this thing makes.

A hardware synthesizer is more than just sounds, though. The physical interface is important and this is where I feel the synth struggles to deliver. The density of the knob layout, the very small printing on the front panel and the colour choices mean I was constantly squinting to see which button and encoder combo I needed for a particular parameter. To make things worse, a number of parameters have specific lists of items to choose from, but because there is no screen, you are left scrolling in the dark. Choosing things like key, scale, specific chords and oscillator modifiers can feel like pot luck. Then there's the inability to assign

velocity, mod wheel or aftertouch without using the software editor. All these problems combined, for me at least, result in the sound design being more pleasurable on the editor than it is on the synth itself. That could be a problem for some. One positive is that the editor is excellent, the layout is good and, when using it, sound design becomes easy and rewarding, but you have to decide whether that is a compromise worth accepting.

If you are happy to do some sound design on the computer and you want an inexpensive and portable package, the Craft Synth 2.0 will do you well. I can imagine it being useful in a live rig, where you program your sounds back at the studio, then take the synth out to a gig. It won't break your back or your bank, and if it does get beer split on it, then it won't be the end of the world. A synth of this price is always going to compromise somewhere. Perhaps Modal Electronics tried to cram too much in, resulting in a necessary reliance on the accompanying app, or maybe they've come up with a deep synthesizer with just the right combination of price and features to appeal. The answer, of course, will depend entirely on the buyer. **///**

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THE RACK CONSOLE



SAM INGLIS

Nowadays, ribbon mics are often perceived as specialist tools for miking things like guitar amps, drum kits and brass instruments, but in their heyday, they served all sorts of applications. Ribbons were used for Foley capture, stage vocals, radio DJ miking, and there were even ribbon lavalier mics for TV interviews.

In many of these roles, the ribbon mic's native figure-8 polar pattern was more of a hindrance than a help, and manufacturers used all sorts of ingenious tricks to try to create a ribbon mic with a 'unidirectional' (cardioid) pickup. Mics such as the Western Electric/Altec 'birdcage' and STC 4033 tackled this challenge by combining a figure-8 ribbon element with an omnidirectional moving-coil capsule, while RCA's chief designer Harry F Olson did so using acoustic means.

The most famous of RCA's unidirectional ribbon mics are probably the iconic 77, with its mechanically switchable patterns, and the rare KU3A 'skunk mic', one of the best ribbon mics ever made. However, there's another, less well-known RCA design that now attracts a cult following. Said to have been designed, among other things, for recording gunshots for movie soundtracks, the BK5 was notable both for its somewhat cardioid pickup pattern and its ruggedness. On neither front does it score highly against today's mics, but what the BK5 does have is a very characteristic mid-forward sound. (Indeed, we often assume today that ribbon mics are universally dark and smooth, but unidirectional models all tend to be brighter and more mid-rangey.)

The Fifth Element

No-one knows more about RCA microphones than Wes Dooley, the founder of AEA. After many years spent servicing the originals, he eventually learned to make every single part of the classic RCA 44 and decided to introduce a modern version of it. This has been followed by original AEA designs such as the R84 and N22, and also by an equally meticulous recreation of the KU3A.

The latest mic to emerge from AEA's Pasadena HQ is a unidirectional ribbon that is designed to be robust, has the number 5 in its name and is housed in a large cylindrical metal body with radio DJ-style yoke suspension. However, the KU5A is meant less as a faithful recreation of the BK5, and more as a new design which takes that mic as its inspiration. Unlike the BK5,

AEA KU5A

Unidirectional Ribbon Microphone

Inspired by RCA's BK5, AEA's latest mic combines classic ribbon tone with a supercardioid polar pattern.

the KU5A is an active microphone, requiring phantom power and delivering a sensitivity of 7mV/Pa. It's not designed to share the BK5's unique tonality, but rather, to sound good in the same sorts of roles for which the BK5 was intended.

The KU5A is thus optimised for use close up and on loud sources, with a maximum SPL of 135 or 141 dB depending on whether you believe the spec sheet or AEA's website. There's no pad, but a button on the end of the mic engages a 6dB/octave high-pass filter turning over at 283Hz, for taming proximity effect when used up close. The output is balanced using a custom transformer.

AEA describe the KU5A as a supercardioid mic, but the detailed polar pattern in the specification suggests some variability with frequency, with the mic appearing perfectly cardioid at 2kHz and narrowing to supercardioid below this. At high frequencies the pattern seems looser

and, being a ribbon transducer, somewhat asymmetric (presumably the two halves of the plot represent measurements taken at a 90-degree mutual angle, though this isn't explicitly stated). The on-axis frequency chart is also cheerfully uneven, with a peak at 2kHz, a dip around 4.5kHz and a fairly steep roll-off from about 9kHz upwards.

We were sent a pair of KU5As for review, housed in smart, lightweight and sensibly sized hard cases. As you'd expect given AEA's pedigree and the cost of the mics, build quality is first-rate, and there's something pleasingly businesslike about the KU5A's chunky metal housing. At 20cm in length and 8cm in diameter, it's not a small mic, and it weighs in at a substantial 1.2kg, but the built-in yoke holds it securely over a reasonably wide angle of adjustment. I'm on the fence about the 3m-long captive cable: it makes the mic super-quick to set up, but annoying to pack away, and in the long run, it presents another potential failure point. And to risk stating the obvious, it's also fixed in length, and I often found myself needing to extend it.

In Use

It's pretty obvious from the specs that the KU5A isn't intended as the last word in uncoloured sound capture: this is a mic that's designed to have its fair share of character. Importantly, though, it's not an out-and-out 'character mic' in the way that the BK5 is. With its massive 3kHz peak, rolled-off highs and non-existent low bass, the BK5 can be a great mic for guitar amps, partly because their bandwidth is restricted in a fairly similar fashion. On most other sources it delivers an instant '50s vibe which is more of a special effect than something you'd want to use all the time.

The KU5A is, to be blunt, a better microphone in almost every respect, and a much smoother-sounding one. Perhaps

AEA KU5A \$1199

PROS

- Delivers the warm tone often associated with high-class ribbons, in a heavy-duty supercardioid mic that can be worked up close.
- Beautifully made.
- Active design means it's not picky about preamps.

CONS

- Expensive.
- Limited high-frequency response won't suit all applications.
- Not everyone will appreciate the captive cable.

SUMMARY

The KU5A combines classic 'ribbon mic' tone with a supercardioid pickup pattern and stage-ready robustness — but it ain't cheap.

it's the form factor, or possibly the examples shown on AEA's website, but I came to view it as an excellent candidate for roles that the Shure SM7B usually fills. Ask a rock singer to scream directly into it from an inch or so away, and it'll shrug its shoulders and say 'Yeah, is that all you got?' On guitar amps, its 2kHz bump is often exactly what's needed, while its 4.5kHz dip tames the annoying unmusical hash that tends to accompany a good distorted tone. I liked it a lot on tenor sax, and it made an interesting snare mic that delivered a really meaty 'thunk'. Although the ribbon is well protected, AEA caution against using it inside kick drums, so I didn't try that.

Because the tail of the supercardioid pattern extends right down to 200Hz and below, some care is needed to avoid feedback in a live context, but the KU5A achieves the rare trick of being articulate without being bright. The subdued high-frequency response means off-axis pickup from cymbals and so on is actually quite benign, and certainly softer-sounding than it is on the SM7. There's enough proximity effect to be useful on thin-sounding vocalists, but not so much that you need to put the high-pass filter on for all close-miking situations.

With little usable frequency response above 10kHz or so, there's no getting away from the fact that the KU5A is a dark microphone by modern standards; nor does it display the sort of flat frequency response and uniform polar pattern you'd expect if you spent the same money on a stage capacitor mic. But you can't sing into a specification, and what matters is not how a mic performs in the test chamber but how it fares in front of musicians. From that perspective, I think AEA have judged

this design very well, creating something that can be used close up and in noisy environments, yet which retains the smooth, warm sound for which ribbon mics are often appreciated. The KU5A is also, as far as I'm aware, the only mic of its kind on the market at present. Beyer used to offer a range of ribbon mics for stage vocal use, but the last of these now seems to have been discontinued, as has the Silvia Classics SC5C/Telefunken RM-5C.

That leaves the question of whether the KU5A's uniqueness and smooth sound justify its cost. The price is hardly surprising for a boutique product handmade to very high standards in California, but the fact remains that for all the KU5A's distinctive qualities, its intended roles are already filled very well by more affordable mics. If you're satisfied with what you get from a high-quality moving-coil design such as the SM7 or RE20 on vocals, guitar amps and so on, it might be hard to contemplate shelling out three or four times as much to add a KU5A to the locker.

That, of course, is your call, but what I can be pretty sure of is that if you do get a KU5A, it won't spend much time sitting idle. It may be pricey, but it's also a seriously useful mic. ■■■



The KU5A has a switchable high-pass filter, which attenuates frequencies below 283Hz by 6dB/octave.

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Win! Arturia AudioFuse 8Pre & Software Bundle

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In a relatively short space of time, French company Arturia's product line-up has grown from offering software emulations of classic analogue synthesizers to making some classic hardware synths of their own. Along the way, they've offered MIDI controllers, drum machines, sequencers and a range of high-quality USB interfaces.

The first two models in the AudioFuse range were desktop interfaces, but the newest — the AudioFuse 8Pre — adopts the professional 19-inch rackmounting standard. And it's not just the form-factor that's gone pro: the 8Pre lives up to its name by including eight of Arturia's mighty DiscretePro preamps. These represent a serious step up in quality compared to most interface preamps, by providing up to 77dB of gain and having an impressively low noise figure of -129dBu. Whether you're recording the quietest sources or using vintage passive microphones, these preamps will handle it all!

Other features of the AudioFuse 8Pre include two front-panel instrument inputs, 10 analogue outputs,



a monitor control section (including speaker switching and a headphone output), and ADAT I/O for easy expandability. And in addition to its USB connectivity, the 8Pre can also work as a stand-alone ADAT expander, so you can easily plumb its premium preamps into your existing setup.

For this month's competition, Arturia are kindly giving away an 8Pre interface and a huge selection of software to go with it. Arturia's V Collection 7 is the culmination of their expertise in modelling analogue instruments, and it includes no fewer than 24 emulations of classic

synths and keyboards, from the likes of ARP, Fairlight, Buchla, Yamaha, Oberheim, Roland and many, many more.

The cherry on top of this well-iced prize cake is Arturia's 3 Delays You'll Actually Use collection. This includes two emulations of classic hardware delays (the Tape-201 and bucket-brigade-based Memory Brigade), plus the all-new Eternity plug-in. All these delays feature extensive modulation and tone-shaping options, and between them should cover every delay sound you'll ever need!

To be in with a chance of winning this fantastic bundle, all you have to do is answer the questions at the link shown, by Friday 31st January. Good luck!

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LP Series





Elektron Digitone Keys

SIMON SHERBOURNE

FM Synthesizer & Sequencer

Elektron's FM powerhouse gains a keyboard and a lot more besides.

The Digitone was easily my favourite bit of new hardware of 2018. It's a four-part FM synth workstation built around Elektron's distinctive sequencing environment. The Digitone Keys takes the desktop module and turns it into an expressive instrument with the addition of a 37-key keyboard, dedicated performance controls and a few surprise extras.

The Digitone Keys is an unusual-looking synth. Elektron have not followed the convention of putting a keyboard in front of a panel of synth controls. As with their 2003 Monomachine keyboard, they've kept the original desktop unit more or less intact and grafted the keyboard onto the side. Unlike the Monomachine the DTK has a raked synth panel, making for a better viewing angle. Overall, this narrow footprint is rather practical, leaving room at the rear for more gear, or space in front for a computer keyboard.

There are a couple of drawbacks to this layout, though. With the desktop machine in front of you, your right hand operates the main encoders while the display shows you what they're doing. On the keyboard version, you're offset to the right. It's now your left hand on the encoders, and the display gets obscured.

The second compromise is that for the width of the instrument you get a comparatively short keyboard range. Three octaves of keys puts the DTK in the same 'semi-compact' synth class as, say,

the Moog Subsequent 37 or Access Virus Polar/Darkstar, which may feel restrictive to 'proper' players. The keys are full-sized and unweighted, with a springy 'synth' action. The keyboard features aftertouch, although not polyphonic.

It's not just keys that have been added. To the left of the original section you get lovely weighted Pitch and Mod wheels with assign buttons, and keyboard octave shifters. Above the keyboard you have eight performance encoders that can be toggled between a default control set and a user bank. Five new buttons engage the various keyboard features such as Multi-Map and MIDI Controller mode. All the controls have that silky smooth feel, and it all adds up to the DTK feeling like a top-quality instrument.

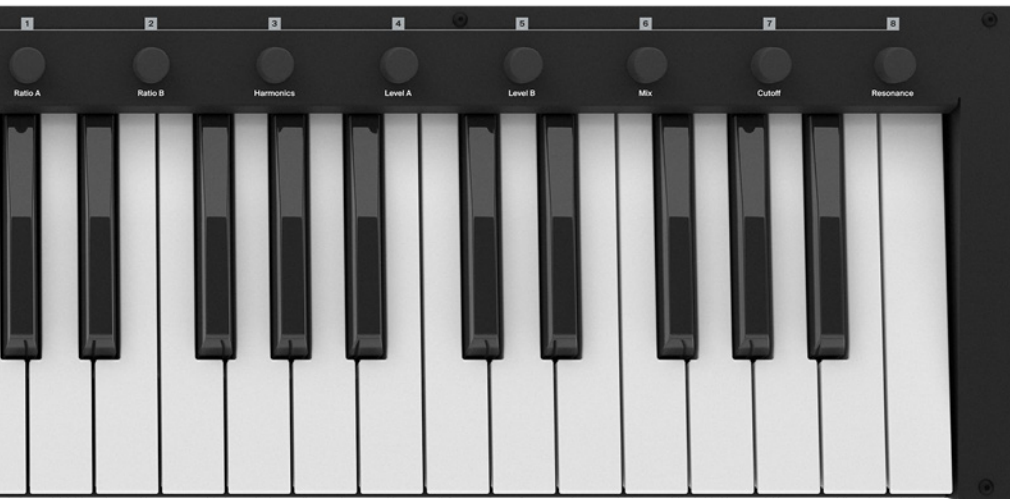
The main volume knob has been moved across into a prominent position above the wheels. The space it leaves behind has been put to excellent use with the addition of a dedicated Sound Browser button. On the desktop machine access to the patch list is a Shifted function.

The original collection of glorious sounds is still intact, but has been bumped down the list by a new bank of presets that are somewhat more keyboard focused. This means that some of the more traditional FM-associated sounds that I was happy to see omitted in the original make a return here. Yes there's a flute, and lots of electric pianos, but they do sound gorgeous coming from the Digitone's pristine synth engine and smooth effects. Compared to the velocity-insensitive input buttons, the

MIDI Control

The Digitone Keys offers MIDI keyboard control of external devices alongside its stand-alone synth capabilities. By default this is a discrete mode, toggled by the MIDI EXT button: in normal operation the keys and controls only route to the internal synth, while in External mode they only send MIDI. However, a collection of settings allow you to share various control sources across both MIDI and the synth.

The eight main encoders send CC data, with the parameter numbers user-definable in the settings. By default the eight encoders on the synth section always stay tied to the internal Digitone controls, but another preference can open these up as extra external CC sources. Likewise the transport buttons can be drafted in to send MIDI Start/Stop/Continue messages to the outside world. Up to eight different setups can be stored and recalled as MIDI EXT 'slots'.



Elektron Digitone Keys \$1299

PROS

- Sounds fantastic.
- 12-track Elektron sequencer.
- Versatile Multi Map and MIDI modes.
- Solid build.
- Separate track outputs.
- External control inputs.

CONS

- Screen gets obscured when using synth panel encoders.
- Three octaves might not be enough as a master keyboard.
- Eight voices is tight for splits/layers.

SUMMARY

The Digitone Keys turns an already brilliant synth into a true instrument. It's also packed with features that would make it great in a live setting if you don't mind the short keyboard range.

Digitone Keys is bristling with expressive possibilities and the sounds are set up to take advantage.

FM Engine

The original review covers the details of the Digitone's FM synthesis, but to recap, Elektron have taken the heart of the classic Yamaha DX concept and re-imagined how

you can interact with the sound generators and shape their output.

Four oscillators (operators) are employed in eight routing schemes (algorithms), offering different FM mappings, feedback loops and audio output points. One of the design decisions I discussed in the original review was the limiting of frequency ratios to quarter-integer steps, which steers

your programming toward more harmonic outcomes. The original firmware did have a detune option which offset two operators and helped you reach harsher terrain. But >>

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Overbridge

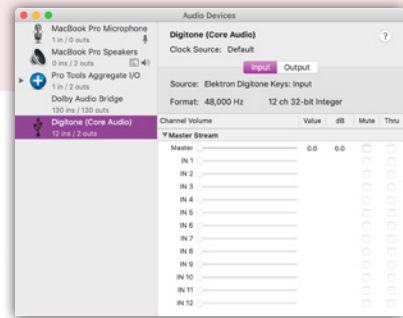
At the time of the original review Overbridge for Digitone was still in development. It's now available as part of the public beta of Overbridge 2.0. Overbridge provides USB-connected features between the Digitone and your computer/DAW. A companion plug-in provides remote control of your hardware and multi-channel audio streaming into your DAW.

Even without the plug-in the Digitone appears as a Core Audio/ASIO Device, which is handy if, like me, you run a host that doesn't support VST/AU. Unless you actually use the Digitone as your main or only audio interface, this is generally going to require setting up an aggregate interface. This always makes me a little nervous in terms of what this is doing for your latency.

When sync'ing up with a DAW, Overbridge gives you 'plug-out' functionality: letting you use the DAW's regular automation system to control the hardware, storing/recalling patch information with your session file, and providing audio I/O in-line with your tracks. Really clever stuff!



The Digitone Overbridge editor runs standalone or as a plug-in, providing control, sync, total recall and audio inputs to your DAW.



Even without the plug-in, the Digitone will appear as a 12-in, 2-out audio interface.

» in the latest iteration you'll now find ratio offsets for all the operators, opening up more sonic potential.

Moving away from DX territory, the Digitones use filters to shape the raw FM source sounds. They have the filter design common to several Elektron devices, teaming a simple high-/low-pass band limiting stage with a more creative resonant filter.

A total of eight voices are available, shared by the four tracks. As before, the tracks all have their own sequencer and arpeggiator which you can use to create multi-part patterns, or now you could use tracks to create split zones on the keyboard.

sound parameters accessible from a small number of controls. The DT Keys takes this to another level, by cherry-picking these controls from their various pages and placing them along the top of the keyboard.

In my original review I said I wished that the FM Ratio and Level controls were in the same place; after all, the ratios have no effect if you don't actually apply some frequency modulation. On the Keys model my wish is granted. The new performance encoders give you Ratios A and B, waveform harmonics, FM levels for A and B, and the Mix control that blends between the two audio tap points in each algorithm. Knobs 7 and 8 default to

a master macro page which could collect controls from different sounds in one place.

Hands-on modulation can be accessed per Sound via the Mod and Pitch Wheels. Both have a dedicated edit button which opens up a familiar Elektron assign page



The Digitone Keys' rear panel is a veritable Elysium of quarter-inch jack sockets and full-size MIDI ports — and it has even gained a sustain pedal input over its keyboardless sibling.

The Digitone Keys has all the Elektron sequencer goodness. On the synth tracks you can do motion sequencing, parameter locks, sound locks, conditional events, etc. And like the Digitakt and Octatrack you get eight MIDI tracks for sequencing external gear. One of my favourite things to do with the Digitone is hook up another synth via the external audio inputs and include it in my patterns.

Control

A major part of the ingenuity in the Digitone design was making all the key

Filter Cutoff and Resonance. Thus the core of the Digitone's sound is laid out for you on just eight encoders.

A long press of the User Assign button flips the encoders to a second page of controls that can be mapped to any of the synth's parameters. Surprisingly, these allocations are saved as part of the current Pattern rather than the sound patch, and are shared by all four sounds in the pattern. This could be useful, with different variations of a pattern having different controls, but I'd like to have different controls for each sound, and maybe

where up to four parameters per wheel can be added with custom ranges. (The desktop DT uses this same system for Velocity assignments). It's unusual for a Pitch Wheel to do double-duty as a second Mod Wheel; it's a cool feature that provides a bi-polar mod control.

New Additions

While adding the keyboard and performance controls, Elektron have taken the opportunity to make some other hardware upgrades round the back of the unit. In addition to the main stereo mix output, there are now four stereo pairs providing discrete outputs for each synth

track. This is fantastic news for both live and studio work. I often want to record out an improvised arrangement from the sequencer into a DAW, without having to fiddle with Overbridge. You can set a global routing scheme, but can choose to store different routings on a per-pattern basis.

The desktop Digitone is missing footpedal inputs for external sustain or expression controllers. The Digitone Keys rights this wrong with two quarter-inch control inputs that can be used for footpedals or CV modulation. Expression pedals or CV inputs are set up in the same way as the wheels, with a config page that lets you assign up to four modulation targets.

The main missing feature when Digitone was launched was portamento. This has now been introduced for both desktop and keyboard versions. In fact you'd almost think Elektron are overcompensating as it's one of the most comprehensive porta implementations I've seen! Among many options you can choose between a smooth glide or stepped glissando and constant rate or constant time. You can also choose to glide between notes in tracks, or between notes using the same synth voice, which could have some interesting results when Patterns start voice stealing across tracks.

Doing The Splits

Unless told otherwise, the Digitone's keyboard plays whichever synth track is currently selected. For things like splits there's a Multi Map mode, with an Edit page that lets you divide the keyboard into zones or 'ranges'. You can assign each of the internal synth tracks to a different range, and offset the actual notes that each range plays. I was a bit surprised that you can't make overlapping zones. Digitone does have a dedicated track layering function as part of its voice options, but you won't be able to, say, layer with external sounds.

There's actually a lot more that you can do with keyboard ranges than simply mapping to the tracks. You can, for example, trigger playback of a pattern or a set of patterns across several keys. Keys can also be set to play a specific sound slot from your current project rather than the main patch selected in each track. This can be set to increment across a range. So, for example, you can create a drum kit within a key range, either by hard mapping a set of keys to specific slots, or storing the sounds in successive slots. If you then record a performance into the track, the different sounds will be written into the sequence using the Digitone's Sound Lock feature.

I was especially intrigued by the ability to make fluid assignments, such as setting a zone to play whichever track is currently selected, or trigger the current pattern. With some thought and practice you could set up a consistent live performance configuration into which your songs and sounds instantly load as you change banks. This is classic Elektron power-user design — the kind of deep feature that really repays an initial investment in learning time.

Conclusion

The Digitone Keys might look like a Digitone with a keyboard stuck on the side, and it is, but it's quite a lot more besides. Discrete track outputs, customisable control and modulation mappings, and deep multi-zone control and trigger functionality for the keyboard make this a real powerhouse on stage or in the studio. And a year on

from my original review, the Digitone continues to be my go-to place for captivating sounds and song ideas. ■■■

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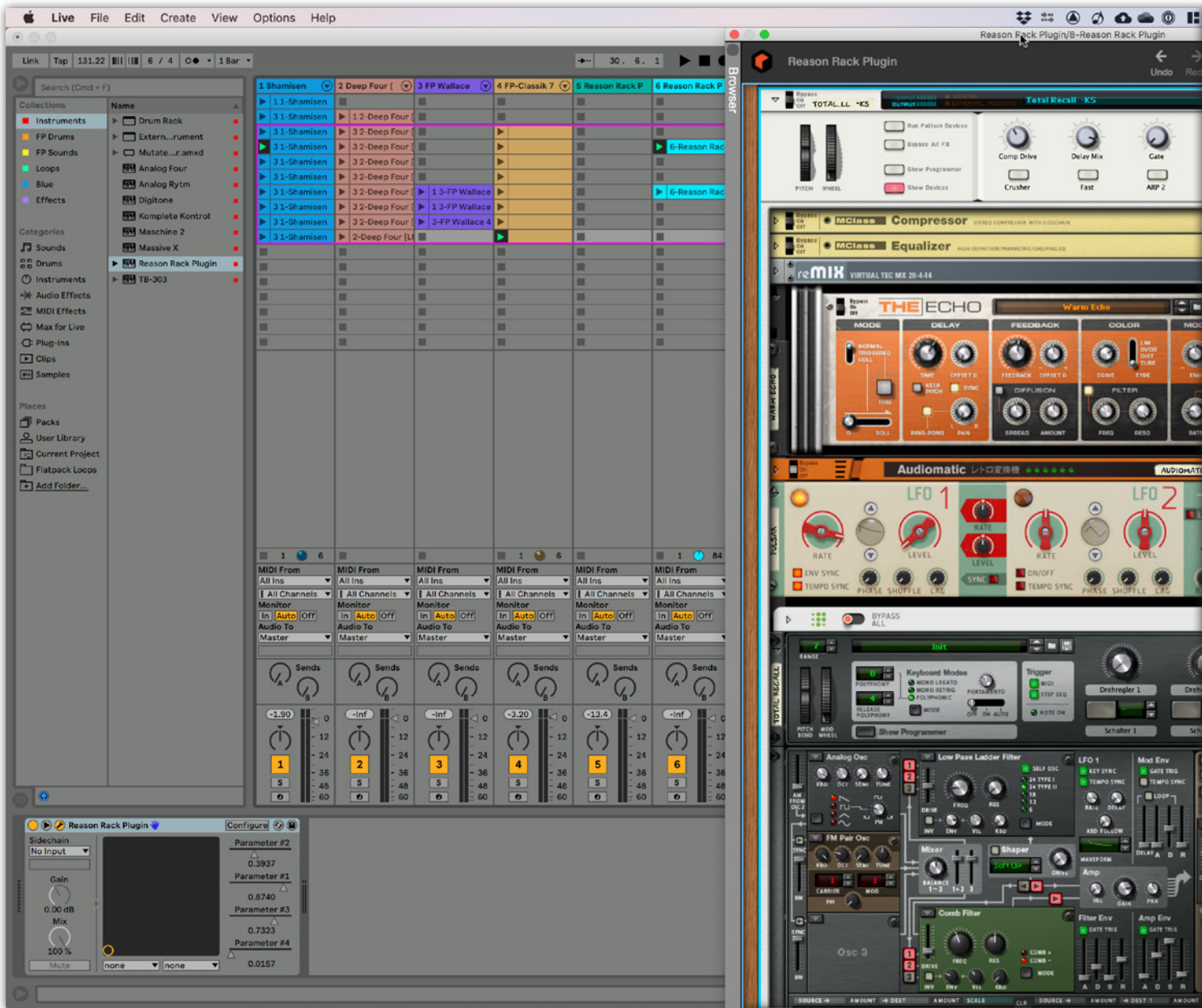
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SIMON SHERBOURNE

If you were asked to explain what Reason is, you'd probably describe a virtual rack of virtual music gear where you can put together synths and samplers, drum machines and effects, sequencers and arpeggiators. This virtual rack has been at the core of the program throughout its 20-year evolution into a complete DAW and mixing package.

That evolution has seen the Rack joined by a multitrack recording timeline and an epic SSL-esque mixer. In the process, Propellerhead — now Reason Studios — have done their best to keep the mixer linked into the original virtual studio rack, but I've often felt part of

Reason 11

Modular Music Production Software

Version 11 introduces the most radical development in Reason's long history: the Reason Rack now runs as a plug-in in other DAWs.

the compelling intuitiveness of the original experience got clouded along the way. So, while I enjoy working in

Reason stand-alone, as often as not I use it as a sound source and sonic playground running as a Rewire client



The Reason Rack running as a plug-in in Ableton Live.

alongside Live or Pro Tools. However, linking up two applications in this way makes for a complicated workflow, and can be off-putting for beginners. Reason 11 thus introduces a new way to work: using Reason, or more specifically the Reason Rack, as a plug-in, placing Reason's devices and modular patching environment inside your favourite DAW.

The ability to run Reason as a plug-in is not the only thing new in version 11, but it certainly dominates the release. The feature list does also include a couple of new premium effects devices, independent modules with the signal processing from the Reason mixer, and



Reason's auto-cabling takes care of most patching for you, but you can flip the Rack plug-in to the rear view and manually patch different input and output busses.

a few updates to the Sequencer, but if Reason is your primary DAW and you don't need the plug-in, you're probably wondering if this is enough to justify updating. However, alongside the Reason 11 launch, Reason Studios are introducing Reason Suite, a collection which includes nearly every extra instrument, effect and

Player they've ever made. If you're an existing Reason user who's not already invested in most of the awesome Rack Extensions like Complex-1 or Parsec then the Suite upgrade price should look tempting.

Reason Within

When I ran the Reason 11 installer, both the stand-alone Reason program and the plug-in were installed; there wasn't an option to install one without the other. Currently, only a VST3 version of the plug-in is included; AU will follow later for Logic Pro users, and I hope Reason Studios consider an AAX version.

In Live the plug-in showed up as both instrument and effect variants. After I dropped the instrument version onto a MIDI track, the plug-in window popped up and presented me with a start screen showing thumbnails of a few Reason instruments. I clicked on my favourite Reason synth, Grain, and it appeared in the Rack and was immediately playable in Live. This is perfect for anyone who simply wants to use individual Reason devices in their DAW without thinking about patching.

Reason Studios Reason 11 \$399

PROS

- It's Reason. As a plug-in.
- New premium effect devices.
- Great value in Suite version.

CONS

- Not much new for pure stand-alone users.
- Rewire is no more.
- No AAX.
- Single-channel MIDI in and no MIDI out in plug-in version.
- Plug-in has limited MIDI controller support.

SUMMARY

With its virtual rack now available as a plug-in, everyone can get a much needed dose of Reason.

» To get at the full menu of Reason devices, there's a pop-up floating in the Rack, or you can open the Browser. Just like in Reason stand-alone, the Browser appears as an extra pane to the left of the Rack, and provides drag-and-drop access to all devices, patches, samples and system-level files. The Browser is the same whether you're in the plug-in or stand-alone app. All my patches, Refills, and favourites were how I'd organised them before. Add-on devices in the Reason's native Rack Extension format are all available inside the Rack plug-in: this is not a cut-down or simplified version of Reason. An exception, though, is that although VST plug-ins are supported in the stand-alone version, they cannot be hosted (nested?) inside the plug-in.

The Reason plug-in uses auto-cabling in much the same way as Reason always has: drop in an effect device after the first instrument, and audio connections through the device will be organised for you. Of course, Reason is famous for its manual patching with virtual cables in the iconic back-of-the-rack view. In the stand-alone version, you hit the Tab key to flip the Rack. In the plug-in there's a Flip Rack button instead, sitting on a toolbar along with Undo/Redo. My Reason muscle memory refuses to update itself here, so after weeks I'm still inadvertently switching Live between Session and Arrange views. Plug-in developers can choose to intercept keyboard commands when their plug-in is focused (for example, the Maschine's plug-in uses Tab to toggle its mixer view). However, I think Reason Studios made the right call here, as this way is consistent within the DAW and avoids confusion.

Presets & Portability

Adding and using individual Reason devices is a piece of cake, but if that's all you ever did, you'd be missing out on Reason's power (and fun) as a modular environment. However, it might not be apparent to a new user how to see examples of multi-device patches: the Reason plug-in browser doesn't have a way to save the whole rack, equivalent to a Reason Song File. Instead, the plug-in works with the standard VST plug-in preset system, so snapshots of the plug-in are saved into your DAW's library, but there are no factory presets in this format.

Reason has a special device called the Combinator that wraps up groups of other devices into a container and provides

a custom macro panel. Many of the patches in Reason's fantastic sound library are stored as Combinators, and these are available in the plug-in. 'Combis' also provide a means of moving patches between the plug-in and stand-alone versions, but Reason Studios could probably make the library more accessible to someone coming fresh to the plug-in. Perhaps there needs to be a VST preset version of the Reason Sounds bank, or maybe it needs to be standardised around the Combinator — or, ideally, an updated Combinator with more macro controls?

Patch & Tweak

Reason has always had a virtual Hardware Interface at the top of its Rack, which is used to route audio connections in and out of the software. Usually the Mixer sits between this I/O and the Rack devices, although if you've ever used Reason as a Rewire client, you've probably bypassed this and connected devices directly to the virtual patchbay.

The Rack plug-in (which doesn't have the master mixer) uses this same

hardware interface concept to bus audio in and out of your DAW. The 'I/O Device' sits at the top of the Rack and provides four inputs and 32 outputs. Again, much of the time, connections here are automated for both instrument and effect variants of the plug-in, but you can patch manually to route multiple outputs from the Rack, or bring in a side-chain signal. A typical application of this would be to send the different channels of one of Reason's drum machines out to different DAW tracks.

Within the rest of the Rack you can patch CV and audio signals between devices, just like in Reason. As well as instruments and effects devices, all the classic utility devices are present: mixers, CV splitters and mergers and, best of all, the Matrix step sequencer and Arpeggiator.

You also get Players. If it's been a while since you checked out Reason, these are among the most significant developments in recent times. Players are, in essence, MIDI effects that snap onto an instrument and process or

»

Suite!

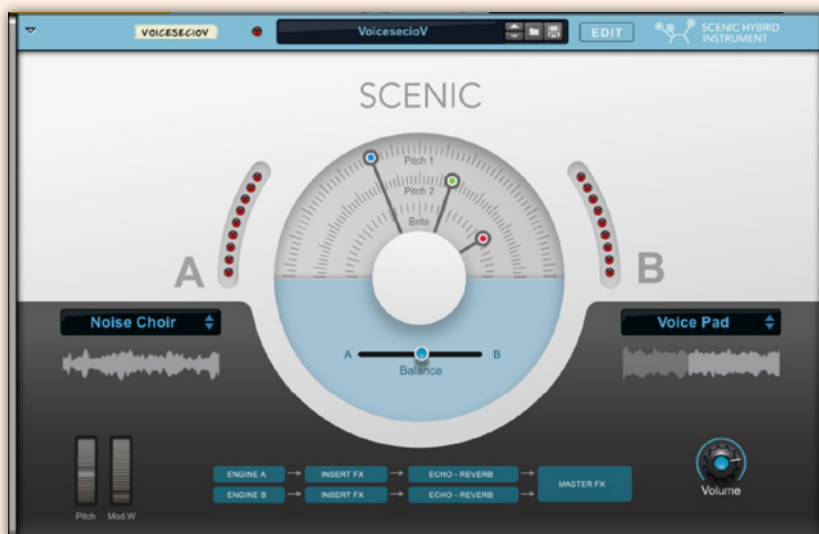
Reason 11 is launched as a new three-tiered product offering that mirrors Ableton Live's Intro, Standard and Suite naming structure. Suite is new, and for a touch under \$600, gets you pretty much all the premium add-ons Reason Studios offer for Reason. This includes the Complex-1 modular synth, which is the best thing in Reason since Grain. You also get the Parsec spectral synth, which is a big favourite of mine.

The big new synth that comes with Suite is Scenic (\$99 separately). This is a hybrid sampler and granular synth along the lines of

Omnisphere. It's designed for easy creation of cinematic soundscapes and features a simple front panel with three macros and a blend control.

For me, these three alone would justify the price, but Suite has other heavy-hitting essentials like Reason Electric Bass and Reason Drum Kits, and the Polar Dual Pitch Shifter. Two of the newest and most powerful Player devices (MIDI effects), PolyStep Sequencer and Drum Sequencer are also included.

The new Scenic instrument is included in Reason Suite.

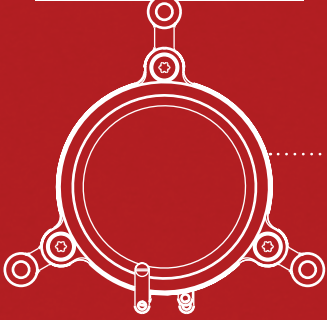


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All the new devices in Reason Standard. There are two brand-new modulation effects, and rackmount versions of Reason's mixer processors.

» generate notes. They can be used for scales and chords, arpeggios and note echoes, but there are also Players that are full-blown drum sequencers or polyphonic note sequencers.

Losing Control

There is one significant difference between the way the Reason plug-in and the stand-alone Reason program connect to the outside world: the plug-in is very limited in terms of MIDI. Each instance of the plug-in has a single Omni-channel MIDI input, and no MIDI outputs.

I discovered quickly that I couldn't recreate how I've worked in the past with Rewire, ie. to use just one Reason Rack containing multiple instruments, triggered from different MIDI tracks in Live. Whereas most multitimbral instruments or modular synth plug-ins let you route different MIDI channels to different sound sources, the Reason plug-in ignores

channel information, so all instruments in the Rack will respond to incoming MIDI. Reason Studios say that when researching the plug-in, they found most users opted for separate instances on each track, with one Rack per sound source. This totally makes sense, but it does lose some of the charm of setting up a unified performance Rack with all your devices set out in front of you, able to interact with each other.

The Rack plug-in does not have Reason's Remote technology, the system that allows manufacturers of hardware controllers to script intelligent integrations with Reason's devices. Nor does the plug-in have a MIDI learn function for quickly taking control of parameters using MIDI CCs. However, parameters on devices will work with your DAW's automation system. In Live this means adding them as you need them in Live's device view, where they can then be mapped to a controller.

Finally, lack of MIDI Out is a shame. The Rack's External Instrument device, used to connect note and gate sources

from Reason to the outside world, is missing from the plug-in. It would be great to be able to use Reason's various sequencing devices and the awesome Players to trigger other tracks in your DAW.

Re-Rewire

My initial thought after testing the Rack plug-in was that I'd probably still use Rewire sometimes. I still have unfinished projects set up like that, and there are times when I want to work in Pro Tools. More importantly, I often start a song in Reason stand-alone, then want to hook it up to Live or Pro Tools later. On those occasions I'm using the Reason Sequencer, so appreciate Rewire's ability to sync my transports as well as route audio and MIDI.

But I'd missed the key fact that Rewire is gone from Reason 11. So that's that. I get that it's complicated to use, and is a complex bit of code to maintain through OS changes, and I'm aware that I'm probably in the minority, but this is a disappointing development for me.

Looking forward, then, I now need to use Reason 10 (which is not overwritten by the 11 installation) to finish those old songs, then figure out a new way of working. Ironically, it's probably going to mean I spend less time in Reason stand-alone, as it will be difficult to progress songs started in the Reason Sequencer in my other DAWs, short of exporting audio files.

Sequencer Tweaks

For the stand-alone Reason user, the Sequencer has had a number of welcome improvements. Grid snapping can now be absolute as well as relative, so you can move things directly to the grid instead of keeping the original offset. This is a mode change; a momentary modifier key would be nice.

Audio clips finally get crossfading. There are some peculiarities to the implementation: you need to enable crossfading on a per-clip basis, after which a fade will appear when you drag that clip across a preceding one. If you want to create a centred crossfade between two clips without moving them, you need to enable the crossfade (on the second clip), then trim out both clip boundaries individually until you get the desired fade. Once a fade exists, though, it's pretty easy to trim it and adjust its centre point.

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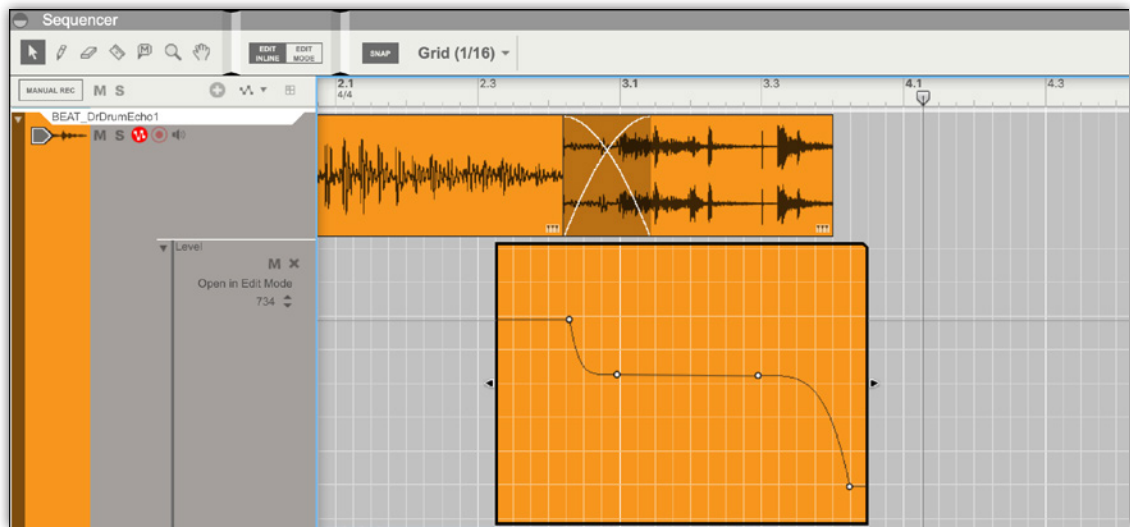
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» you can now draw curves into automaton and MIDI CC graphs. Hover the selector over the line between two points and it becomes a curve tool that will bend the line. Additionally, if you Shift-click with the Pencil tool you can add a breakpoint on the existing automation line, so you can create an edit point without disturbing the surrounding data.



The Sequencer gets some new features in 11, such as curved automation and crossfades.

Another Pencil Tool improvement is drawing in multiple MIDI notes. If you hold Alt/Option while drawing you'll add a string of notes on the current grid. Individual MIDI notes can now be muted within clips. Finally — and this might sound sad — my favourite new feature is being able to zoom track heights independently. Sometimes it's the little things!

What's Missing

However much work a developer puts into an update, there will inevitably be things on your personal wishlist that don't get

done. For me, the most significant thing that's not in Reason 11 is any update to its graphics. The Reason Rack still doesn't support 'retina' graphics, or resolution independence. It's beginning to look a bit small and fuzzy around the edges on today's high-density displays. We know they can do it: Reason's flagship synth, Europa, is available as a separate VST plug-in and has a glorious, resizable high-resolution panel.

As a stand-alone user, I'd love to see the Sequencer's much under-used Blocks feature developed into a real-time scene system, which would be a real boost to both live performance and song arrangement. It's great to see that

there were at least a few Sequencer improvements in this version, sending a message that Reason Studios aren't giving up on stand-alone at the expense of the plug-in.

Go Up To Eleven?

Of all the changes that Reason has gone through, turning the Reason Rack into a plug-in is probably the most important. It will be handy for current Reason users, but more importantly, it opens the shop to the many users of other DAWs. In this context, Reason 11 is a monster modular instrument and effects collection with huge depth and creative potential. It's also a lot more accessible than the various virtual modular synths on the scene, as many of its modules are complete working devices rather than component-level blocks.

If you're a pure Reason user who doesn't need the plug-in then the update currently looks a bit thin, but do investigate the significantly discounted upgrade to the new Suite package if you're craving a fix of new Rack toys. Either way, Reason Studios have a history of dot releases that deliver meaty features between paid updates. On that note, while the plug-in has a few gaps (and no Rewire to fall back on) this is, after all, the first version and will continue to develop. Overall this is a really solid first iteration of a new and exciting direction. **■■■**

New Devices

No Reason upgrade would be complete without some new devices, and Reason 11 has two new premium effects. Sweeper is a modulation effect with three discrete modes: Phaser, Flanger and Filter. The Phaser and Flanger are really sweet and sound great on most material. The Filter, which has multiple modes and a Drive stage, is a bonus that could have been justified as a separate device.

The defining feature of Sweeper is the modulation section. There's a conventional LFO, but more interesting is the envelope section, which can be switched between a manual custom shape and an audio follower. The envelope designer is the same one we've seen before on a number of Reason instruments, but what makes Sweeper particularly fun and useful is the Audio Trig feature. This lets you set a threshold for the input signal to self-trigger the envelope.

Quartet is a more compact device, as it doesn't have the envelope section. It's a chorus and ensemble effect with four different modes, providing a lush palette of glossy spatial

and thickening effects. The BBD mode is considerably more dramatic than the regular chorus, returning instant Banshees goth as the default. Dialled down it provides a lovely warm space that for some reason made me think "classy '80s", if that's not oxymoronic. FFT mode provides a band-limited chorus, and the Grain mode uses a grain cloud instead of delays to create the ensemble effect. All in all, a really nice couple of additions to the Rack.

There are in fact another three devices added to Reason in version 11, although they're not technically as new as Sweeper and Quartet. The Master Bus Compressor, Channel Dynamics and Channel EQ are rackmounted versions of the signal processing in the Reason mixer. The primary reason these modules have been extracted from the main mixer is to make them available in the plug-in version of Reason. The bus compressor is a particularly nice effect to have access to in other DAWs, and even in Reason stand-alone, these devices could come in useful in Combinators and other group patches.

\$ Reason 11 Suite \$599; Reason 11 \$399; Reason 11 Intro \$99.

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JON BURTON

I'm known as a fan of analogue desks, but needs must, and when a tour came in where the only practical choice was digital, I looked around for what to use. There is a fair selection of choice in the upper end of the market and as the shows were headline festival slots, I had the luxury of being able to choose from any of the more commonly available desks.

Having recently been out on a small theatre and club tour, picking up a different

desk every night, I had already ruled out some models. I won't name names, but some I didn't like the sound of, while some I found just lacked the flexibility and power I needed. In the end it came down to just two or three choices. And when a colleague phoned me up having just got back from a tour raving about the Allen & Heath dLive, it tipped the scales.

I have to say that I do have a relationship with Allen & Heath. I have lent them some of my effects units and mic preamps to be modelled. In return, they loaned me a mixer

Allen & Heath dLive

PROS

- A great feature set.
- Flexible and easy to use with a very shallow learning curve.
- A great all-rounder.

CONS

- After several weeks I could find no real downside to this console.

SUMMARY

Significantly cheaper than some top-end desks, the dLive punches well above its weight. Adaptable and well featured, the dLive is equally at home at FOH or on monitor duties, as well as on theatre shows.

for a tour to mix effects returns. It was the entry-level Qu-Pac, and it was so simple to use. Even better was that everyone said that the dLive shared its sibling's simple user-friendly interface — see the 'Perfect Harmony' box. I had never mixed a show on an Allen & Heath digital console, however, so it was a leap of faith, but one I was

willing to make, with some reservations.

As an engineer who rarely gets any pre-production time, I needed a console that I could get to grips with quickly. I was going straight into a festival run with Scottish rock giants Biffy Clyro, whom I had not worked with before, so I had to be confident I could achieve a mix quickly, and one that matched the band's and my expectations. So I headed down to Allen & Heath's HQ for a day with the dLive, a hard disk of live tracks in my bag.

Patch Work

A&H are based in Penryn, Cornwall, where they have a nice modern facility that houses the offices, R&D, and some manufacturing. The main manufacturing is based in China, but tightly controlled from the UK. I was



greeted by Lead DSP Engineer Anthony Evans, who ushered me into the mix room, a curtained-off office deep in the building, with a small PA system and a dLive S5000 for me to try. After a bit of almost inevitable messing around with networking we had connected my MacBook Pro to the desk and set it up in 'virtual soundcheck' mode. For those of you unfamiliar with the concept, a virtual soundcheck allows you to record the desk channels via a network such as Dante or, in this case, Waves SoundGrid, onto a laptop. This recording can then be played back through the same input channels, allowing you to seamlessly switch between recorded and real-time inputs. This is invaluable when you don't get enough time with the actual instruments, such as in my case where there would be little or no time for soundchecks.

Using a show recorded the year before by the band's previous engineer, I was able to patch the channels to the desk as I would when doing it live. Patching the dLive is easy. A dedicated I/O button on the surface opens the patching window. This tells you which stagebox is connected — in this case a DM64 stage rack, which gives you 64 mic/line inputs and 32 outputs. The DM64 also has three card slots, one



Photo: Jan Burton

The dLive S5000 getting a workout with Biffy Clyro at Indie19 festival, Ireland.

of which was taken up with the Waves V3 SoundGrid network card that we were using to connect my laptop. Patching takes place in a standard grid-style page, but that page is easily zoomed in on if, like me, your eyesight is not too good, or you have sausage fingers! This is all done through the right-hand touchscreen.

It is best practice to name channels and outputs before patching and this can be done either on the I/O page or via the channels. I had actually prepared the patch and named the channels using the free offline editing and control software Director at home to save time. My basic show loaded, I was ready to start listening to audio.

At Your Surface

The dLive's strength, for me, is the ease of setting up. Everything is pretty much where you think it will be, the surface is sparse but functional, and there are enough shortcut buttons to get to where you need reasonably quickly. All models in the dLive series share much the same surface controls, but with the larger models giving you more faders, screens and a bit more direct control. I had opted for the second-largest frame size, the S5000, which gives you 28 faders and dual 12-inch capacitive touchscreens. I wanted a reasonable number of faders and this seemed a good compromise between size, weight and features.

The desk comes with various templates that give you a starting point for how to

set up the desk. "Templates for what?" you may ask. Well, the surface has 28 faders and 26 soft keys that are fully assignable, and you need to tell them what they are, either by loading one of the ready-made templates or building up from scratch. I opted for a FOH template that gave me a good starting point, with roles assigned to all the faders, and the faders laid out in banks. Under the left screen are 12 fader strips, and each of those has four soft keys, a single rotary, an LED meter and a mini display screen providing channel/fader information. The next bank has just eight faders, laid out in a similar fashion, and to the right is another bank of eight. Between the banks are more soft keys that give you direct access to different layers. The concept of layers is a common one but in case you haven't used a digital desk before, it's like having several mixing desks piled on top of one another. Each layer allows you to control a different set of 12 or 8 channels, depending on which bank you are on. This increases the flexibility as it allows you, with simple switching on the surface, to control up to 168 assignable faders. Each fader can be assigned to a variety of uses. Here starts the fun.

Fader Assignments

I am a great believer in the idea of setting out one's stall, and by this I mean arranging everything in a clear and logical way. The phrase comes from market traders, where a well-presented stall would be more appealing to customers as they could see >>

» what was on sale easily and choice was laid out in a logical way: veg on one side, fruit on the other. I always strive to lay my desk out in a similar way, so that it made sense to me. I have always preferred inputs on my left, outputs to my right and any important channels to fall close to my right hand. Most analogue consoles were laid out this way, and so to a certain extent I have chosen to follow that layout, often grouping things so they were closer to hand. With a desk like the dLive any fader can control any channel, so any fader can govern any output, any subgroup, any DCA (Digital Control Amplifier or control group) or effect return. With all this choice it was good to start with a basic template or I would never make a decision!

Having assigned all the faders how I wanted, I immediately changed my mind. For example, instead of having a single fader control the left and right output bus, I decided to switch to two separate faders. There are many reasons to do this, especially if you are providing lots of outputs for subs, side hangs, front fills and so on, as you can control and mute each individually. I had decided to use the digital equivalent of VCA groups so I could group various channels together and control them with one fader. This is a great way of mixing, especially with a large channel list — and


Guitar Rig

My main headache on this gig was the large guitar rig of the main man, Simon Neil. I had inherited the previous engineer's channel list and wouldn't have time to make any changes as these could impact negatively on the monitor engineer. I decided to go with the existing list, adding just one channel. The guitar setup was not overly complicated but featured four amplifiers. The guitar started with a clean sound from a Fender Twin, which was miked and DI'ed; this ran continuously, but for heavier parts a Marshall would be switched in, also with a mic and a DI. Occasionally the sound would be bolstered by one of two Kemper amplifiers, which provided a direct feed. This meant a total of six channels of guitars stage-right. And just to make it more interesting, the guitar rig was duplicated! Yes, there was a second Marshall, Fender and pair of Kempers off stage should anything happen to the main rig. This was also miked and DI'ed up, should any problems occur. This is where the dLive really came into its own...

Laying out the fader banks, I assigned a layer for the main guitar and on the layer below I duplicated the positions with the backup guitar rig. Inside the channel preferences I linked the Fender Twin mic

with the mic on the spare. Linking gives you options and I decided to link every option apart from routing and mutes. Whenever I made a change on the main rig (EQ, compression, level and so on) it would be replicated on the spare rig. I then routed all the channels of the main rig to a master subgroup and the spare to a second subgroup. If I lost the main rig, I would just have to unmute the spare rig and it would all be set as the main, in theory!

Using amplifiers like Kempers alongside traditional amplifiers such as Marshalls and Fenders is problematical. The Kempers, being digital, do have latency. This is not noticeable when they are used stand-alone, but if the output is mixed with the output of the Marshall, comb-filtering occurs. This can make the sound thin at some frequencies, accentuated at others, and is generally undesirable. The dLive has a handy page that gives you access to all the inputs and allows you to delay any channel by samples, metres, feet or milliseconds. By delaying the channels of the Marshall and Twin by about 5ms I was able to compensate for the latency of the Kempers. This was all easy to achieve and within 20 minutes I had overcome the biggest headache I had anticipated.



The dLive's capacitive touchscreens respond to all the familiar phone and tablet gestures.

on this gig I needed 58, which is a lot, and another reason for my choice of digital.

One of the great things I found about the dLive was how easy it was to adapt the surface to my needs. I decided to use the left-hand bank of 12 faders exclusively for the input channels, and this I did in blocks. The first bank comprised the drum channels, which on my channel list was channels 6-19, as the first five channels were used for trigger lines: contact mics on the drums that could be used to open the noise gates on the associated drum. As I could set and leave the trigger channels, I missed them out, so my surface would start with bank 1, fader 1, kick drum. A nice traditional start! The next 11 channels were taken up with drums and cymbals and a nice woodblock. The next layer (B) had some electronic drums and the overheads as well as the bass channels. Layer C was dedicated to guitar channels (see box), including the main guitar rig and the channels for the second guitarist. Layer D handled backing

tracks and backing vocals. On layer E I placed the spare guitar channels and my iPod. The last layer I used as a dumping ground for rarely accessed sources, or things I wouldn't need to adjust, such as the drum triggers.

DCA Groups

Having now assigned all the inputs to a fader, and the faders to banks in a way that made sense to me and that I could find quickly, I turned my attention to the middle bank of faders. This would be my control area: faders I would normally assign as control groups. I started off by making a DCA group for all the channels in a way that made sense musically and ergonomically. I like to group rhythm parts such as kick and snare and bass so I can push up the beat with one fader. I made a series of assignments and filled up the bank. After an hour or so of mixing it occurred to me that I was using one group just for the lead vocal. Why not just swap out that group with the vocal channel? This is easy to do and something I would almost feel confident doing on the fly during a gig. Decisions about the surface have a dedicated button and that brings up an option screen. Choose Strip Assign and it brings up a little graphic of the layout. You can then drag and drop strips onto the bank of your choice. It's easy!

With my seven DCAs and one vocal channel set up in the middle of the desk, I started to get adventurous. My effects DCA I swapped out for my main reverb return. I routed input channels to audio groups, and placed the groups on the surface instead of the inputs. Why? Well, a DCA is just a controller and doesn't display the audio content in any way. By using a subgroup, I could get a meter

Alternatives

The dLive console without the input rack retails at around the same price as the **Midas Pro2**, **Soundcraft Vi4** and **DiGiCo SD12** consoles. If you include a mix rack it brings it in line with the **Yamaha CL** range. The dLive is a very keenly priced product, and in terms of channel count, control and facilities, it competes with some consoles that are almost twice its price.

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Clockwise from top-left: the EQ screen, the compressor/gate screen, the 16VU comp/limiter, and the Hypabass subharmonic generator.

» appearing beside the channel showing any activity on that group, which would be useful in working out whether stuff was happening on the electronic kit or backing tracks.

By now I had a fairly heavily customised layout. I had buried all the various outputs on their own layer, which included discrete outputs for recording, for PA left and right, for subs, for in-fills, and even a mix for the lighting designer. On the top layer I still had control of the left and right master faders that were feeding all these mixes via an easily assignable matrix.

On The Road

By now you will have got the idea that this is an extremely customisable console, which it is, but what does it sound like? At this point I'm going to jump forward two weeks until after the line check at the first festival. The band were headlining the Isle Of Wight Festival, which was being filmed and recorded by Shapland Mobile. To assist on what was going to be a hectic hands-on mix, my system tech Richie had been volunteered to sit in the truck and contribute.

"I'd better see what you have done on the desk," he announced. We then spent 10 minutes going over the channels looking at the EQ, the compression, the gating and so on. Bass drum, a bit of low-mids taken out; snare top, high-pass filter set at 80Hz, no EQ, no gate; snare bottom, filter set but in bypass (my mistake). Looking through most of the channels, not a lot was going on with the equalisation. Most channels had no inserts. For the lead vocal and the bass I chose one of the onboard insertable compressors from the library, otherwise I had just used the channel compressor, if anything at all.

I think this lack of equalisation and plug-ins tells you a lot about the desk. It just

sounds pretty good! You stick a microphone in and it sounds OK. You choose the right mic, put it in the right place and you don't need to do anything! I was pretty impressed with my lack of adjustments.

Where I had dived into a channel and used one of the library plug-in-style devices on this console, they had all worked just as I expected. There is a choice of a dbx compressor, a good OptTronik compressor, an excellent Mighty compressor, even a retro-looking one with a VU meter (I'll let you guess the inspiration). I found patching easy and the sounds matched my expectations. I have a huge collection of analogue outboard and I know what these units sound like, so it was refreshing to find plug-ins that emulated the originals in a convincing way. There are also a few specialist units which need to be inserted rather than just recalling from the channel strip. Amongst these are Dyn8, a powerful multiband compressor/dynamic equaliser. This took a bit of getting used to but repaid the effort — the Dyn8 matched the power and flexibility of some of my favourite plug-ins.

Screen Show

Equalisation on the desk, as with most of the channel options, takes place on the left-hand screen, which has a series of dedicated controls around it. The most important are probably the 12 dedicated knobs split into four sections: bass, low-mid, high-mid, high. The lowest row, illuminated with a red LED in the cap, controls cut and

boost. Above and to the left, with a yellow LED, is the Q or bandwidth control, and to its right, with a green LED, is the frequency selector. Immediately above these knobs, a section of the screen is dedicated to displaying the equalisation either as a single curve showing the summed action of the four sections, or as four individual sections showing the action of each band, or as a visual display of the knob settings in terms of frequency and gain in dB (the least useful, I found). These display modes can be switched using a button to the right. The left side of the screen has four more knobs beside it, whose purpose varies depending on the screen setting. In most modes these control gain, trim and the high- and low-pass filter frequencies. Once again, dedicated knobs control crucial functions.

In between the two screens are another six knobs. These control the functions of the right-hand screen, which can be set to display a number of options. I found the gate/comp option to be most useful. This view provides a cut-down look at the gate and compression section of the selected channel, with the top half showing the gate and two of the knobs controlling the threshold and depth. (It would have been nice to see the third control utilised for decay). Below the compressor section you have control over ratio, threshold and gain as well as a handy display in both sections showing you what the signal was doing (à la Waves).

Other options include auxiliary sends,



effects controls and more comprehensive compressor controls, and you can assign shortcuts to three views of your choice onto the soft keys.

Despite screen space being allocated to these other tasks there is still sufficient space for the main equaliser window. The centre touchscreen gives you all the information you could need, as well as space to manipulate the sound with even the clumsiest digits. The equaliser provides lots of options, with fully parametric mid-range bands, and the lows and highs being switchable between parametric, shelving and high-/low-pass options. This last option did cause me problems, as the options cycle through with the filter coming second. This can cause you to lose all the top end, as I did on the main mix, before you cycle through to shelving, the setting I was looking for. This, I believe, is being looked at in a later revision of the software.

The channel view provides you with not just a view of these screens but tabs to view the primary high- and low-pass filters, which can be set to various slope types such as Butterworth and Bessel. The gate

and compressor sections can be seen in detail here too, with all the options such as side-chain filtering and key options. These are easy to route internally from drop-down menus. I used the key option to trigger the gate opening on the various drums from my drum contact mic channels. The side-chain input I used to help attenuate the Fender Twin channels when the Marshall came in to help balance the level between the two. Internal routing and control were easy, straightforward and well laid out.

Outputs

Outputs are as easy to use as inputs. From subgroups to matrix mixes, auxiliaries and outputs, the controls are again very accessible. The fact that they all use the left screen helps build muscle memory as you always tend to go to the same place to achieve similar tasks. I like this way of working and I found it very intuitive. The second screen to the right of the console tends to cover the 'house-keeping' options such as patching, metering, scene control, ganging, grouping and utility options. During a show I tended to keep this on the

Hire Ground

Why review a desk that most people can't afford? I have been asked this question a few times, and the answer is twofold. There's the aspirational side, of course — people like to read reviews of top-end gear even though most could not afford to buy it unless they won the lottery — but the more important factor is that this is not a desk that is usually purchased by engineers, but rather by hire companies. Hire businesses stock the items that are most requested, and that they know they can recoup their investment on through rentals. From speaking to a few major companies this summer Allen & Heath are establishing themselves as a solid stock item, so even if you'll never be able to own one yourself, there's a good chance you may end up working behind one!

meter page. This gave me an overview of all the inputs and outputs of the desk, but the screen also has a dedicated area that always shows a meter strip that can be scrolled left or right to see a section of the inputs and outputs, even when you're using the main screen for something else. »

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» The right-hand screen has three shortcut buttons that bring up a choice of screens that have dedicated knobs. I found this balance of dedicated and multifunction knobs very user friendly. In cases where a knob is not needed, such as on the OptTronik compressor, which only has two controls, the LED in the cap turns off, indicating it has no function. This does save you the embarrassment of twiddling to no effect, and also provides monitor engineers with a handy DFA control!

The desk can also be used to record and play direct from a USB drive, although I did find the desk a bit picky about which drives it would talk to. Having said that, the ability to stick a playlist on a drive and play walk-in music without having to leave a laptop or iPod out I found very appealing. I recorded all the shows using an optional MAD1 card, as the band's high channel count exceeded the 32-channel limit for recording to USB. This worked pretty well and meant that I could do a virtual soundcheck of the previous night's show at the next gig.

Effects

I have a personal passion for effects, which is one of the reasons I had first struck up a conversation with the amiable Robin Clark, now MD at A&H. We had got chatting at a trade show many years ago about effects units, and what we did and didn't like. This led to a few more conversations and Rob going away and modelling a few of my favourites. It was nice to have a manufacturer show such passion and this is reflected in the built-in effects on offer in the dLive.

I was very impressed with their version of my go-to reverb, the EMT 250 plate. I have several emulations and this stands up well against the best. It was very usable and sat in the mix well. There are almost 50 presets across several different reverbs, and these are a combination of hardware emulations and 'real' spaces. I managed to find something for all my needs on this show. There are numerous echo machines as well as more specialist effects such as chorus and ADT, and a really good sub-harmonic generator which got some use!

I felt no need to look outside of the desk for any extra sounds. Normally I would tour my favourite effects but I had no need — even my beloved Dimension D was nicely emulated on board.

Final Thoughts

There are so many features of the desk that I don't have time or space to go into, but

Perfect Harmony

Harmony is the name Allen & Heath give to their graphical user interface. It features across their digital product range and unifies the user experience. It is based on a conscious decision by A&H to adopt the technologies we use every day with our smartphones and tablets, and to that end, the capacitive touchscreens on their desks respond to all the usual pinch, swipe and drag-and-drop gestures that we have all become accustomed to.

The Harmony interface shares a common look across the whole product range, so going, as I did, from a desk with primarily app control (the Qu-Pac) to the large multi-screened dLive was easy.

All the digital products have the ability to be controlled by apps allowing you wireless remote control, as well as access to most of the

features on the desk in compact and clear, well laid-out remote screens. The desk screens and any apps share a common format, so things are in the same place, meaning very little re-learning as I progressed from an iPad app to the larger more complex desk. This doesn't mean that you can do everything on a Qu-Pac that you can on a dLive, but that most tasks have a seamless commonality.

The combination of a well-presented and sensibly thought-out GUI along with dedicated hardware control sped up my workflow to a level that was comparable ergonomically to the analogue flow I am accustomed to. I believe that even a Luddite like myself would find my productivity increasing exponentially as I got more familiar with this console.

I wanted to give a summary of how I found it in practice. I used the desk on six shows, I only had one soundcheck, and they were all high-pressure gigs. I can honestly say I felt confident at every show. By the first half of the first show I had relaxed into using the desk and started enjoying it. By the last show I think I had pretty much mastered it enough to say I was a confident user. My only blips were down to a dodgy multicore (it wouldn't let the desk talk to the mix rack), and my own incompetence (I hit my poorly assigned master mute whilst distractedly thinking I was adjusting the echo tempo). I received a reasonable number of compliments, and the desk definitely sparked interest in the FOH tower.

I was not alone on my festival run, however: on several occasions I came across bands further down the bill also using a dLive, but mostly the C1500. These diminutive desks have only 12 faders and a single screen, but with similar power to their larger brethren, and come in a package you can carry single-handedly through a field and check in on a flight! If push came to shove it would be an option I would happily consider.

I stood in on sound duties on a few dates with Bring Me The Horizon. Their monitor engineer, Jared Daly, is a confirmed advocate of dLive. He was touring with a large-format version but also carried a Peli hard case with his backup desk. This was a regular laptop running the A&H Director software and an IP8 dLive remote controller. This is a small box with eight faders and a few softkeys. If his console did fail for any reason, he would still be able to control his mix rack with this powerful setup. This seemed very appealing, especially when a cheap

'get out of jail' option is preferable to touring with a spare desk (yes, it happens!). As a monitor desk Daly was more than convinced by the power of the dLive and was using it to the max, with a huge channel count and numerous mixes.

In conclusion, I found the desk a doddle to use. I also enjoyed using it. I was happy to develop my mix and had no qualms moving channels around, adjusting the layout, and tinkering with things without the safety blanket of a soundcheck. I think this is very telling; there are not many desks that I feel I can approach with this level of confidence. It is all too easy on some desks to tinker and find that you have inadvertently de-routed a channel or an output, but with the dLive I felt I was in control, and I don't consider myself an experienced digital desk operator! I also spent the summer proudly showing off my somewhat diminutive FOH setup. I had no external rack beyond a set of drawers and a pair of laptops (one for recording, the second for playback). My footprint was smaller than most other headliners, for which the local techs were grateful, and it all worked each day. In short, I would highly recommend the dLive to anyone, and in any of the formats. ■■■

Thanks to Skan Hire for their support, and Biffy Clyro for providing such great-sounding inputs!

\$ dLive system street prices from \$15,000 to \$42,000.

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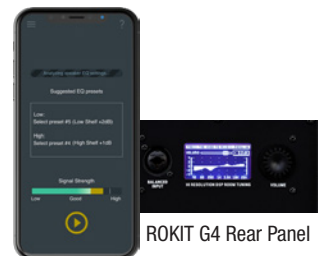
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Neural DSP Archetype: Nolly

Guitar Amp Modelling Software

Neural DSP have bottled the guitar-recording wizardry of one of the hottest rock and metal producers around.

JOHN WALDEN

If you inhabit the world of modern rock, metal and prog, then Adam 'Nolly' Getgood may be a familiar name. Over recent years, he has built himself an impressive reputation as a songwriter, guitarist, bassist and, in particular, as a producer, through his work with his own band Periphery and other modern rock/progressive metal bands such as Haken, Architects and Animals As Leaders. He is also a complete technology obsessive and, seemingly, a thoroughly nice bloke!

Nolly is no stranger to music technology development. *SOS* have already reviewed the excellent *Modern & Massive* drum sample library (see the February 2019 issue) released under his own GetGoodDrums brand, but his latest endeavour sees him teaming up with developers Neural DSP to create a compact virtual guitar amp and effects setup.

4x4

In contrast to open-ended plug-ins like NI's Guitar Rig and Line 6's Helix Native, Archetype: Nolly features a fairly compact equipment list. It's based around four



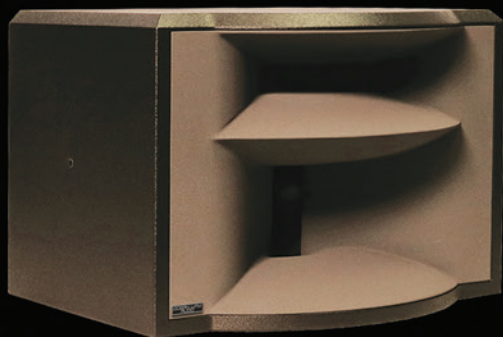
amps and four cabs taken directly from Nolly's own collection, plus a modest, but carefully chosen, set of preamplifier stompboxes comprising two overdrives, a compressor and a delay. A nine-band EQ sits between amp and cab, and there are also post-cabinet delay and reverb effects. A very flexible dual microphone configuration simulates mic choice and placement. Some of the original amps have been modded by Nolly, and the speaker/cab selections are based upon over 700 of his own impulse responses — this is a 'signature' plug-in with the artist's signature embedded deep within.

The actual amps modelled for the plug-in are not specified but the rumour

mill would suggest that they are a Bognor Shiva, a modded Marshall, a Peavey 5150 and a Victory Kraken. It's a small collection but, in combination with the included cabs and the two distinctly different overdrives, the plug-in can easily cover the diverse array of pristine cleans, crunch and super-saturated (but controlled) distortion that is characteristic of modern rock bands. This is ably demonstrated by the excellent presets, which include some fabulous ambient clean tones that make good use of the post-cab delay and reverb; some even bypass the cab modelling for a super-articulate sound. Whatever level of crunch and overdrive you might like, you can get there, and the >>

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» nine-band graphic is particularly effective, shaping the tone so that even at higher gain settings the sound can still be controlled and remains articulate.

Smooth Surface

Archetype: Nolly can run in both standard native plug-in formats and as a stand-alone application. The UI is clean and crisp, with five icons located top centre to navigate between the pre-effects, amp head, graphic EQ, cab/mic and post-effects control sets. Other core controls are laid out at the top of the screen including the input/output level knobs, the preset system, mono/stereo switch, the oversampling switch (which trades off quality against CPU load) and a noise gate. The latter is particularly useful at higher gain settings, but it is either on or off around the threshold; it might have been nice to have a little more control. The smaller amp head and cab icons at the very bottom of the display allow you to mix and match between the four amp and four cab models.

All the controls are very easy to use and, in the main, the visuals reflect the familiar real-world hardware. Aside from the four different cabs, you also get to choose between a one or two-mic setup,



The built-in cabinet and microphone modelling is a highlight of the Nolly sound.

with four modelled mic choices. Mic position can be adjusted and level, pan and phase controls are also provided. The cab/mic modelling is exceptionally well done, but it's also possible to use third-party IRs for added variety. It's also worth noting that the software is MIDI controllable, both in its stand-alone incarnation and using your usual DAW MIDI automation features.

Better Than Good

While some potential users might have wished for a few extra stompbox options, particularly in the modulation sphere, I doubt many people will complain about the sound. I've already commented on the diversity, but what really stands out is the quality of the modelling. I'm fortunate enough to have used most of the obvious software competition at some point, and have access to a Line 6 Helix as well as a couple of nice valve amps for my own studio guitar work. Sonically, Archetype: Nolly is up there with the very best.

I'll add a personal caveat that I think perhaps applies to every way of recording guitar parts. For Nolly to 'work', I think you need to let the guitar player monitor at a level that is capable of getting them excited, and where the guitar itself interacts with the air moved by the speakers. If you are lucky enough to have a recording space where some volume is possible, then I think Archetype: Nolly both sounds — and, importantly for the player, feels — very close indeed to a real amp. The various amp controls behave in a very realistic fashion, and the interaction

between the preamp gain and power-amp volume controls feels very amp-like. Hats off to Neural DSP and Nolly: this is some of the best amp/cab modelling I've ever experienced.

Just as musician fans of Nolly have snapped up GetGoodDrums because of his prowess at getting great rock drum sounds, I suspect there will be a very receptive fanbase for Archetype: Nolly. There is more to getting a good rock or metal guitar tone than winding up the gain, and the presets include some examples straight from the Periphery locker. That said, while Archetype: Nolly can certainly cover a lot of sonic ground, it's genre-specific to an extent, so if you are looking for a more generic, all-in-one guitar rig modelling solution, there are perhaps more obvious choices.

I'm impressed by what Neural DSP and Nolly have created here. On the surface, it may appear to be a niche plug-in, but it's actually a surprisingly versatile virtual guitar rig built from a modest, but well chosen, selection of original components. Neural obviously have some clever boffins capable of capitalising on Nolly's own gear geekery, and if you are a fan of modern rock, metal and prog guitar tones, this is pretty much a no-brainer, especially given the relatively modest price. **///**

Neural DSP Archetype: Nolly €135

PROS

- Sophisticated modelling that can produce very realistic results.
- Great range of clean and overdriven tones for modern rock and metal players.
- Competitively priced.

CONS

- Limited range of effects might deter some.

SUMMARY

Neural DSP and Adam 'Nolly' Getgood have created something really rather good that deserves to find an audience beyond the obvious rock and metal genres. Archetype: Nolly is surprisingly flexible and sounds fabulous.

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Tascam DR-07X

TOM FLINT

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They replace the DR-07MkII and DR-05 in Tascam's range of portable recorders, and the idea is that everything a user needs to make a decent-quality basic audio recording is already built into these compact and lightweight devices. To make this possible in such a small package, neither recorder offers XLR inputs or 48V phantom power, and the onboard mics are permanently attached to the recorders. Although that's not always as convenient as using mics

Stereo Digital Recorder

This updated entry-level pocket recorder doubles up as a USB audio interface for computers and iOS devices.

that can be independently positioned close to a sound source, the position of the DR-07X's mics can be adjusted to change the stereo image. And should you not need even that functionality, the less expensive DR-05X's mics are fixed in place. Indeed, the mics and their impact

on the form factor of the two devices are the only things that differentiate the two models. (Here I'll focus mainly on the DR-07X, and explain the differences between the two in more detail in the separate box.) And despite the lack of XLR inputs, both models retain their predecessors' stereo 3.5-mm mini-jack socket. Intended primarily for connecting lavalier mics, there are nonetheless other external mics that can be connected in this way. (I should also mention that if you do have need of a more feature-rich device, Tascam's range includes several more sophisticated models, with features ranging from XLR mic preamps with phantom power to multitrack recording.)

Tascam haven't attempted to reduce the size of the DR-07X compared with its predecessor; it remains a bit fatter, longer and wider than my 10 year-old Olympus LS-5. But then it doesn't really need to be smaller; the DR-07X is still nice to hold and remains shorter and narrower than a typical smartphone. Importantly, Tascam's design is at the same time spacious and quite minimal — it should appeal to people who favour simplicity and clarity, but it also has one main benefit of larger devices, in that there's plenty of space for the controls and buttons.

In The Round

The controls are mostly very intuitive, though I had to check the manual a couple of times during the review. It's nice that a physical user manual is included, though note that this gives only 14 pages to each language so some functions are only partially explained — if you need more detail, you'll want to refer to the 62-page English-language PDF Reference Manual, which can be downloaded from Tascam's website. This reference manual describes a number



Tascam DR-07X \$120

PROS

- Capable of A-B and X-Y microphone configurations.
- Simple, streamlined interface and controls.
- Works as a stereo audio interface via USB.
- Rotating the LCD screen in tuner mode is a great idea.

CONS

- While you can't expect every accessory at this price, an SD card or onboard storage would have been nice.

SUMMARY

The DR-07X is a simple-to-use stereo recorder with the ability to act as a computer interface with PC, Mac and iOS devices. It should appeal to people who want to record audio in a variety of situations, as well as solo instrumentalists hoping to capture a focussed recording of their performance.

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The USB interfacing and the ability to rotate the onboard directional mics to form an X-Y pair, and change the display orientation, could be useful for self-recording musicians.

» of potentially useful functions we've described before when reviewing the previous incarnations of this recorder, so I won't detail them here — features such as, for example, level-dependent automatic recording, and recording to multiple incremental files of a specified size.

Most operations on the DR-07X are enacted via a circular panel of buttons on the upper surface, which behaves a bit like a gaming controller. Tascam's implementation of this idea (which has also been used by other manufacturers) employs nine buttons. The Up, Down, Left and Right ones navigate the menus, control headphone and record volumes, help the user flick through audio files, and act as fast-forward and reverse buttons. In the centre is the Play button, which doubles up as a selector for menu items. And arranged in between all of these are four fixed-purpose buttons.

I can't say I've ever seriously used the tools offered by the PB Cont and Mark buttons on a Tascam portable recorder outside of a review, but should you need such facilities these allow you to set up loops and markers, and alter the playback speed of a file. It's also possible to select a section of audio and 'punch in' to record over it. A further button, Menu, makes it possible to access all the record, play, I/O and system settings. Finally, Tascam have minimised the pain and time involved in navigating sub-menus by including what they call a 'Quick' button. This brings up menu items that are related to the current operating mode — and this can be a real time-saver.

Overall, this circle of buttons is very neat and effective, and it's hard to make an argument for having any further dedicated buttons for specific functions, even though there's potentially the panel space necessary to host some.

If you've used any of Tascam's other portable recorders in recent years, the DR-07X's screen layout will seem very familiar. It's not the biggest display and there's a lot of information crammed into it, but it is bright, sharp and accurate enough to make it easy to monitor important things like recording levels. At the top of the screen is one LED that warns when signals peak too high and another that shows the record status. Recording is primed and started using one of the few dedicated buttons, found at the foot of the screen. Adjacent to that is a final control that acts as On/Off (when held down for a time, so that you don't switch the recorder off accidentally).

Ins & Outs

On the right-hand side of the recorder, a slot accepts micro SD/SDHC/SDXC cards of up to 128GB. In fact, you'll need a card to start recording — and you'll have to buy this separately if you don't already have one, since Tascam don't include a card as standard and there's no internal memory either. This seems a bit stingy to me, since previous recorders I've used have either been fitted with some internal memory or have shipped with a small (eg. 2GB) card as standard, so that you have everything in the box that's required to start recording. Apart from the hard-copy user manual and

two batteries, there are no accessories included at all — there's no memory card, no wrist strap, no wind shield and no protective bag or carry case.

Such things are available separately, though, including a mains power adapter, an external battery pack and a lavalier mic, and there seem to be several resellers who offer packages comprising the recorder and various different accessory bundles, including an SD card and both Tascam's and third-party accessories. In the interests of balance and fairness, I should point out that it's arguable that Tascam's minimalist approach to the bundled accessories allows the price to be kept down for those who already have the accessories they need — but I'd still have preferred to see some storage included!

Next to the card slot is the USB connector, via which the DR-07X can act as an audio interface and capture the sound from the onboard mics directly to your DAW. The USB audio interfacing is one of this recorder's most attractive functions in my view, and makes it a significant improvement on the earlier DR-07MkII model. (But again, you'll need to provide your own USB cable!). On the opposite edge is a 3.5-mm mini-jack socket for headphone monitoring, and this can monitor playback from the recorder or from your DAW via USB.

Underneath the recorder, a threaded socket allows for mounting on a stand/tripod, and there's also a battery compartment that takes two AA cells, plus a single speaker that can be used for playback monitoring of recordings when

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» headphones are not in use.

The only hardware features left unmentioned so far are the two unidirectional microphones. These can be

DR-05X

The only difference between the lower-priced DR-05X model and the DR-07X is the design of the onboard electret condenser microphones. The DR-07X sports a pair of movable directional mics, whereas those on the DR-05X are omnidirectional and positioned in a fixed array, facing away from one another at a 120-degree mutual angle. This arrangement is suitable for recording in situations where the wanted sound sources are spread around the room. Because there's no need for a hinge to swivel the mics, the DR-05X is also narrower and shorter than its stablemate — making it a nicer object to carry in the pocket. The sound of the 05X is of a similar quality to the 07X when the latter is in its A-B configuration. However, I suspect most recording musicians would find the X-Y option of the 07X more attractive.



rotated to face outwards at a 90-degree mutual angle (i.e. each 45 degrees away from centre), or folded in so that they cross over in an X-Y configuration, in which case the left/right channels are obviously swapped. Helpfully, as soon as a mic is moved, the display offers the option of swapping the channels, making it impossible to get things mixed up accidentally.

Solo musicians recording themselves playing instruments such as acoustic guitar will obviously be viewing the screen from the wrong side when the mics are pointing at them, so Tascam have made it possible to rotate the display's orientation 180 degrees when using the tuner. I think this is a great idea. (I looked hard, but couldn't find a way to rotate the metering/monitoring page, which would have been better still!)

Recording Tests

To test the DR-07X's audio-recording abilities, I ran it alongside the DR-05X and my Olympus LS-5 (which I've used as a reference for previous portable stereo recorder reviews). First, I set the DR-07X's mics into an A-B configuration to match that of the other two, and I found that there was little to choose between them in terms of noise levels, tonal balance and overall quality. When in the X-Y position, though, the DR-07X sounded noticeably tighter, more focussed and intimate than the other two, and I definitely preferred this on the acoustic guitar I was using. I then checked the X-Y position alongside my Yamaha H6 fitted with its (XYH6) X-Y capsule, and matched the levels and setups as closely as possible. I recalled that the H6 has tended to feel a little more 'solid' in the low end than Tascam's recorders (possibly because it's a bit flatter around 1kHz) and, indeed, the DR-07X did seem just a touch brighter — but even after lots of switching from one recording to the other, I couldn't say that there was really much in it.

Using the recorder as an interface over USB was fairly straightforward. A couple of aging DAWs I still insist on using couldn't detect it, but I had no problem when I tested it with a range of current software. Making a connection is simply a matter of selecting 'Audio I/F' and then telling the device to connect to either iOS or PC/Mac. After that, you decide whether you wish to use the recorder in Direct mode — in which the input signal is monitored over headphones — or in PC/

Alternatives

There are a few small, fixed mic-position recorders on the market at the moment, such as the **Zoom H1N** and **Olympus LS-P4**, but I can't think of any similarly priced recorders that offer both dual mic positions and USB interfacing.

Mac mode, in which the recorder monitors the playback from the DAW.

The interface sample-rate options are 44.1 and 48 kHz, yet when recording to the internal card it's also possible to select 96kHz if you feel the need. There are a range of MP3 options up to 320kbps, plus uncompressed 16- and 24-bit WAV formats.

Summing Up

On the face of it, Tascam's range of portable recorders haven't changed massively over the last decade. But that's no bad thing — it's mainly down to the fact that the original hardware and OS were extremely well designed in the first place. There's certainly no logic in changing things just for the sake of it, so when Tascam have made tweaks it has typically been to accommodate new technologies, such as higher-capacity memory cards, better mic preamps and sharper LCD screens. Along the way, though, Tascam have made the occasional refinement to their recorders' menus and buttons to improve usability, sometimes adding controls and features, but at other times removing ones that weren't really needed and merely added to the overall clutter.

Consequently, the DR-07X really does feel like the culmination of years of refinement. Its USB interfacing is a particularly attractive new feature and will be of interest to podcasters and musicians wanting to record direct to portable laptop-based setups. There's no doubt that Tascam's DR-40X, DR-44WL and DR-100MkIII are more 'professional' tools, chiefly because of their XLR inputs and phantom-power provision, but they're also more costly and bulkier, making them less convenient in some respects. For many users — those who are happy to make the most of the onboard mics — the DR-07X is all that's required. In short, it's neat, flexible, affordable, it sounds good and is easy to use. What more could you want? **■■■**

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Universal Audio Apollo x4 & Twin X

Thunderbolt 3 Audio Interfaces UA's new desktop interfaces give their rackmount siblings a run for their money.

SAM INGLIS

When you buy an expensive item of studio hardware, you usually expect it to last. Mics and outboard processors half a century old remain the touchstones by which modern gear is often judged. In the world of audio interfaces, however, change is relentless, and yesterday's cutting-edge tools now gather dust in machine rooms. Of course, it's in the interests of manufacturers to have us replace our audio interfaces as often as possible, but it's also a process that reflects real technological development. Year on year, we expect better audio specifications,

more DSP power and higher channel counts, not to mention compatibility with whatever connector Apple decide to put on their laptops this month.

Audio interface manufacturers thus refresh their product lines every few years to keep them in line with buyers' expectations, and 2018 saw Universal Audio update their rackmount Apollo range. Reviewed in November 2018's *SOS*, the Apollo X interfaces feature state-of-the-art conversion and clocking, six DSP cores, built-in talkback, surround monitor control, Thunderbolt 3 connectors and friendlier I/O arrangements that reflect real-world use cases. Individually, there was perhaps no single headline

improvement that changed the lives of Apollo users overnight, but together, these changes made for a pretty substantial upgrade. UA also introduced a new interface, the Apollo x6, which filled the gap between the desktop Apollo Twin MkII and the much more costly Apollo x8 and x8p.

Super Twin

Being only 18 months old at the time, the Twin MkII itself escaped the 2018 round of upgrades, but its turn has now come. The MkII was available in Solo, Duo or Quad configurations, with one, two or four SHARC DSP chips respectively. Apparently the Solo was the least popular of these, despite being the most affordable, and so the new Twin X is thus available only in Duo or Quad formats. The two Twin X variants are identical in every respect apart from DSP grunt, and offer exactly



the same I/O as was available on the Twin MkII (which already had a talkback mic, for example). There wasn't space in the Twin chassis to include the dual-crystal clock circuitry found in the rackmount Apollo X interfaces, but the benefits would have been less apparent in the Twin in any case, as its smaller dimensions and consequent shorter signal paths leave less scope for jitter to creep in. However, the Twin X does feature the updated converters found in the other X-series units, and now sports the Type-C connector associated with Thunderbolt 3 rather than the older Mini DisplayPort socket.

Bigger news is the introduction of another new interface in the X series. The Apollo x4 represents a second intermediate stage between the Twin X and the flagship x8 and x8p, but this time in desktop format. About 50 percent wider than the Twin and

x4 Or x6?

When the Apollo Twin was first launched, there was plenty of clear water both price-wise and feature-wise between it and the rackmount Apollos. The advent of the x6 and now the x4 has narrowed that gap, but with both of these interfaces offering similar specs at around the same price, it might not be obvious which is the better option for you. The main differences are outlined below.

Compared with the x4, the x6 has an extra pair of line inputs and outputs, coaxial S/PDIF

and word-clock I/O, a second Thunderbolt port and a second pair of optical sockets to support eight-channel ADAT recording at high sample rates. Its line-level I/O can be aligned for +24dBu operation, and its line and mic inputs have separate connectors. Its 'ladder' meters are also more detailed.

Lastly, the x4 has four Unison-enabled mic preamps to the x6's two, and the top-panel layout makes it a much more convenient monitor controller.



The Apollo x4's back panel squeezes in a generous selection of I/O. Headphone ports and high-impedance sockets are found at the front of the unit.

sharing the same specs and control features, it's only available in Quad DSP configuration and, compared with the Twin, provides significantly more I/O. The x4 has four mic preamps to the Twin's pair, all implementing UA's Unison preamp-modelling architecture.

High-impedance inputs for connecting guitars are available on the first two of these, and the Twin's single headphone output has been duplicated. On the rear panel, there are now two pairs of line outputs in addition to the main monitor outputs — meaning that, among other things, the x4 can be used to switch between three pairs of stereo speakers, though unlike the rackmount Apollos, it doesn't support 5.1 surround monitoring.

Out Of The Box

UA haven't felt the need to refresh the look of the Twin, and the new models are outwardly identical to the MkII in most respects; driver and latency performance is also unchanged. Removing the new interfaces from their packaging,

however, reveals a mundane but very welcome improvement. The Twin MkII used a wall-wart PSU with a cable that was annoyingly short, and unless you happened to have wall a socket in exactly the right place, actually getting your shiny new desktop interface on to your desktop

»



Physically, the Apollo Twin X is exactly the same as the Twin MkII...



... The only difference is the updated Type-C Thunderbolt port.

Universal Audio Apollo x4 & Twin X

PROS

- With two headphone outputs, six line outs and four mic preamps, the new x4 is a desktop interface that can tackle serious recording tasks.
- Twin X upgrades bring it in line with the Apollo X rackmount interfaces.
- New power supplies are much more convenient.
- Channel DSP Pairing in UAD 9.10 enables more complex plug-in chains in the Console utility.

CONS

- No Thunderbolt cable supplied.

SUMMARY

Universal Audio's new portable interfaces have specs to match their rackmount Apollo X units, and the x4 packs the I/O to handle a wide variety of tracking situations. The desktop interface is no longer the poor relation!

» could be frustrating. Mercifully, this has now been replaced by a two-part line-lump affair more akin to a laptop power supply. It's at least twice as long as the old cable and if you wanted it to reach even further, all you'd need to do is replace the figure-8 mains cable with a longer one. I'd love to report that UA have also decided to supply Thunderbolt cables with their Thunderbolt interfaces too, but, well, they haven't.

Being more than 23cm long and weighing over 1.5kg, the x4 isn't something you can pop in your laptop bag, but it does cater to a number of recording scenarios that weren't possible with a Twin. As an example, many of today's stars bring an engineer on the road with them so that they can write and record in hotel rooms. The x4's additional headphone output will make it perfect for this role, allowing both performer and engineer to hear the takes as they go down. Alternatively, two guitarists could jam simultaneously through complex chains of UA plug-ins (see box), while four mic inputs are sufficient to enable basic drum recording without an ADAT expander. With audio specs that match those of the rackmount Apollo Xs, there's no sense that the convenience of the desktop format comes with any performance penalty, and

Pair & Pair Alike

Even those who don't update to the latest UA hardware benefit from the software improvements that are regularly rolled out. These always tempt us with more plug-in recreations of iconic hardware we can't afford, but the recent UAD 9.10 software also introduced some interesting new functionality.

One of the key differences between native and DSP-based systems is that the processing resources available in the latter are fixed. A DSP chip has a certain amount of memory and can perform a certain number of calculations in a given time frame, and each plug-in you load grabs a fixed share of these resources when you load it. This arrangement has the advantage of being inherently stable: because these DSP resources aren't shared with non-music tasks, the system can tell in advance whether there's enough juice left to run the plug-in you want to run, and it will continue to run reliably even as the DSP load approaches 100 percent. On the down side, however, it's also inherently inflexible. Not only is it not possible for a single plug-in to span multiple DSP chips, but in previous versions of UA's Console utility, this was also true of plug-in chains. So, to use a series of plug-ins on the same channel, they all had to run on the same DSP chip, and you could only have one chip's worth of processing per channel.

As UAD plug-ins have become more sophisticated, and especially now that there are more Unison plug-ins designed expressly for tracking purposes, this has become

a problem in some contexts. For instance, some vocalists like to have their signal compressed and pitch-corrected in headphones, with a splash of reverb. If you were already using a DSP-intensive Unison preamp emulation such as the Neve 1073, and then you added Auto-Tune and a couple of other processors within the Console channel strip, you could easily exceed the resources of a single SHARC chip — and even if the other three chips were idle, they couldn't be brought in to help out.

UAD 9.10 software introduces a workaround for this problem called Channel DSP Pairing. In essence, a DSP Pair bridges two of the SHARC chips in an Apollo interface, allowing the creation of a single Console plug-in chain with twice the normal DSP resources available to it. However, individual plug-ins must still fit on to a single DSP, and DSP Pairs are not available on Aux or Talkback channels. A more significant restriction is that DSP Pairing effectively works by repurposing the Console mixer's Virtual channels, so for each DSP Pair you create, you lose two Virtual channels.

The Console still doesn't have DAW return faders, but another change in UAD 9.10 helps counter the all-too-common but annoying situation where the level of an input source is being drowned out by that of the DAW return: the Console faders now have 12dB gain above unity. This makes it easier to leave the appropriate headroom while recording, yet still achieve a workable balance of the live signal against previously recorded material.

for many, I think this will actually be the preferred option over the x6.

Other than the power supply, there are few surprises on offer, but that's not really

“There is clearly a growing market for premium interfaces that can handle critical recording tasks, and if you have that sort of money to spend, the Apollo x4 is a heavyweight contender.”

the point. There's obvious benefit both to UA and to new buyers in having the Twin be closely aligned with the X-series interfaces, even if there aren't any major new features that would justify upgrading from the MkII. The new x4, meanwhile, is an interesting and, I think, a very shrewd move on UA's part. When desktop audio interfaces first appeared, they were usually budget affairs, but as more and more serious mixing and production is tackled 'in the box', we've seen a market open

up for professional interfaces that diverge from the traditional rackmount form factor. There's nothing amateurish about the Prism Sound Lyra, the RME Babyface Pro, the Merging Anubis or the JoeCo Cello, but the most obvious rival for the x4's crown is Antelope Audio's Zen Tour. Similarly priced, this has broadly comparable I/O and audio specs that look very impressive, and can host its own low-latency

effects and processors courtesy of Antelope's FPGA-based AFX system. There is clearly a growing market for premium interfaces that can handle critical recording tasks, and if you have that sort of money to spend, the Apollo x4 is a heavyweight contender. ■■■

\$ Apollo Twin X Duo Core \$899, Twin X Quad Core \$1399, Apollo x4 \$1799.
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UJAM's Virtual Drummers may not come with their own van and annoying personal habits, but they're quick, reliable and always on time.

JOHN WALDEN

With a team including some of the original designers of Steinberg's Virtual Guitarist, over the last few years UJAM have been bringing that concept up to date in the form of virtual guitarists, bassists and drummers. The latest releases bring significant updates to the three virtual drummers within their software-based session musician catalogue; Solid, Phat and Heavy. As the titles suggest, each specialises in a somewhat different range of musical styles, but they are built around the same front-end and underlying

engine. So, if you are auditioning for a (virtual) session drummer, do UJAM have a suitable candidate?

Agents Of Groove?

While Steinberg's own Groove Agent has subsequently gone in a very different direction, in Solid, Phat and Heavy, UJAM are essentially reviving and updating the original GA concept, while also adding new features. That original concept was based around three key features; a solid set of sample-based drum sounds, a collection of style-based preset patterns that could easily be triggered to create a full drum

performance, and a UI that emphasised efficient workflow rather than more forensic editing of the drum sounds and performances. UJAM have retained those design intentions here.

In terms of the first of these three elements, Solid, Phat and Heavy each have their own collection of five sample-based drum kits. Within a specific kit, you also get the option to swap between multiple snares, kicks, etc, allowing you to customise your drum selection. The second major element of each package is a sizeable collection of style-based pattern presets. Each of these presets contains a series of 23 related drum patterns, providing intros, patterns for verse or chorus sections, fills, breakdowns, endings, a few specials (extra groove variations) and a 'stop'.

The third element is the UI. This is straightforward, with the bulk of the controls displayed in a single screen. The top is dominated by a keyboard graphic

that specifies how the sounds can be triggered. For the patterns, MIDI keys spanning C3-B4 are used. All the patterns associated with the black keys (for example, the fills), when triggered, play back once and, if you have Latch enabled, the engine then returns to the previous main pattern. While patterns are in playback, the Mod Wheel can be used in real time to control the performance intensity. This essentially emulates a drummer playing softer or harder and, under the hood, uses MIDI velocity to trigger different velocity layers; it's a very effective tool for adding some life to the performance.

Opening the Micro Timing panel allows you to switch between normal, half-speed and double-speed tempo, adjust the push/pull feel, apply some swing and adjust the tightness (or otherwise) of the playing. This is all pretty standard stuff for a virtual drummer instrument but does, of course, both increase your performance options and allow you to dial in just how 'human' you want your virtual drummer to sound.

Keys C1-B3 allow you to trigger the currently selected drum sounds manually,

whether from a MIDI keyboard, electric kit, or your sequencer. This is great if you want to add further variations to a pattern, create your own custom fills, or simply create a performance entirely of your own. This section of the MIDI mapping also provides a very neat 'mute' option for use during pattern playback. You can, for example, use this to drop the snare or hi-hat out of a pattern for added variety in a breakdown.

In terms of user adjustments, once a kit is loaded, at a macro level, you also get six Mix Presets, each of which has its own sonic character. For example, in Solid, these are Smooth, Edge, Retro, Big, Power and Crush. Around the central logo, you also get the Amount and Slam controls. The former allows you to blend in just how much the Mix Preset settings alter the sound of the underlying kit, while Slam applies some overall compression. Things can get pretty squashed at extreme settings (if you like that kind of thing) but it's very usable at lower values.

The update to v2 provides some very useful further mix options, and I'll come

back to these in a moment, but the overall control set remains compact and well focused. If you just want to get a solid drum part together with a minimum of fuss, the workflow of Solid, Phat and Heavy will have some obvious appeal.

New For Two

So, what's new in v2? Well, perhaps the first thing to note for existing owners is that the new versions are fully backwards compatible with the originals; projects created with the originals should therefore work fine with these v2 updates. There are, however, four significant new features that will appeal to both first-time purchasers and updaters.

Top of that list is expanded collections of pattern style sets and top-level presets. Each title now contains 60 (rather than 30) pattern style presets, effectively doubling the number of included patterns. The top-level presets within each title — which pair a kit configuration with one of the 60 pattern style sets — have also been doubled from 150 to 300. For non-drummers, the pattern content of any virtual drummer is just as important as

»



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The lower panel provides an expanded set of controls for customising the sound of your drums but without any danger of option paralysis.

» the drum sounds themselves; more pattern content to work with is therefore a very welcome addition.

Second, UJAM have added further options for tweaking the drum mix. The changes are all both useful and easy to use. The bottom-most section of the UI offers eight mixer channels, six for the instruments (drum types) — kick, snare, tom, hi-hat, ride, crash — and two ambience channels for overheads and room. As well as being able to adjust their relative balance, by selecting one of these channels you can access further controls for that channel at the very base of the display. For the instrument channels, this includes Type (choose between multiple kicks, snares, etc), Decay (for example, to tighten up the ring-out on the toms), Tune (adjust the pitch) and a Reverb send level. Solo and Mute buttons are also included. For the Ambience channels, you get a one-knob compressor and low-/high-cut controls to customise the overhead/room ambience sound. The contribution of overheads and room mics is often the gel required to glue an acoustic drum kit together so these controls are very welcome additions.

The Master section provides a master output level control, a drop-down Mix Preset menu (offering a small selection of different reverb styles), and a Reverb level knob. The Maximize knob provides another dollop of dynamics control, while the Saturate control does exactly what you would expect and actually sounds rather good.

The third new feature is provided by the Individual button. You can activate this for each channel (including the overhead/room channels) and, providing you also activate the appropriate extra outputs within your DAW host, you can then route individual drums to either the master stereo out or to their own stereo output. These individual channels bypass the Master section, but it does allow you to apply the full arsenal of your DAW's processing options to your drum mix should you so wish.

Fourth, and also very welcome given my comment above about the importance of the pattern collection in any virtual drummer, all three titles now support MIDI pattern drag and drop to your DAW. You



can, therefore, drag any of the patterns available in Solid, Phat or Heavy to a MIDI track and edit them as with any other MIDI clip. You can then replay that MIDI either with any suitable sample-based drum instrument, including any of the other UJAM drummers.

The upper, dotted, sections of the pattern MIDI keys are where you can drag from and, rather wonderfully, the current Mod Wheel setting as you drag is used to scale the MIDI velocity data. You can therefore easily create multiple copies of a pattern with different MIDI velocities to build a more dynamic performance; very neat. Not unsurprisingly — but just to be clear — you cannot drag MIDI patterns back on to one of the pattern slots within Solid, Phat or Heavy; drag and drop only works in one direction.

Solid, Phat & Heavy?

While they obviously share a UI/engine, the three titles are different in terms of both sounds and pattern styles. As a starting point for solid, studio-friendly drum sounds, Solid would be the obvious choice. In fact, given the five distinct kits, and the six Mix Preset starting points, sonically, Solid can easily offer you something for singer-songwriter, straight-ahead pop, funk, R&B, blues, rock and, with a suitable bit of added grit, into heavy rock. The pattern styles span various flavours of ballads, pop, reggae, funk, soul, rock with the occasional less mainstream options (for example, Soca, Train Shuffle or Country Waltz 3/4) also included.

While still based around acoustic drums, Phat is obviously targeted more at hip-hop, funk and R&B styles. As a result, the kicks are a little bigger and have more low-end boom, while the snares offer punch rather than crack. The pattern styles also reflect the greater dance/groove focus with lots of

soul, funk, R&B and house-based options plus some classic rhythms within presets such as Bolero, ChaChaCha and Salsa. There are also a few presets based upon 3/4 and 5/4 meters. As a slight aside, if hip-hop, R&B and EDM are particularly your thing it's also worth noting that UJAM do have a family of three Beatmaker virtual drummers — Dope, Eden and Hustle — based upon drum machine-style sounds.

There are no prizes for guessing that Heavy has a distinctly 'rawk' flavour to both its sounds and pattern styles. I'm perhaps less convinced of it as a source for that super-tight, modern metal or progressive rock sound straight out the box, although you can get in the ballpark by making use of the new mixing/sound-tweaking features described above. In terms of style

UJAM Virtual Drummer 2 Solid, Phat & Heavy \$99 Each

PROS

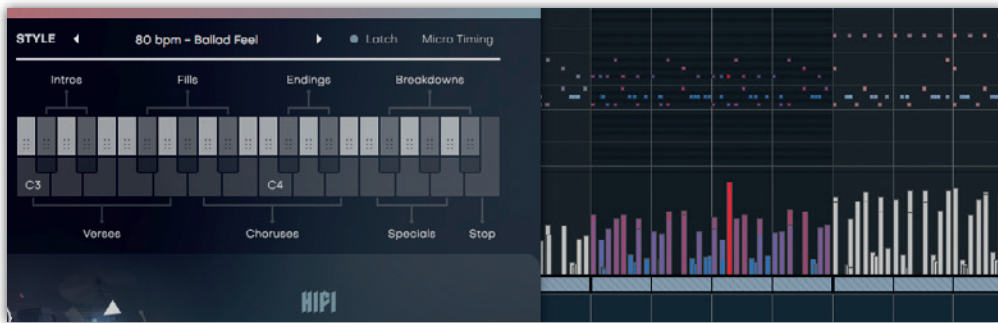
- Remains super-easy to use even with the expanded feature set.
- Good range of underlying drum sounds and expanded pattern collections.
- New mixing, MIDI export and multiple output features very well implemented.

CONS

- Degree of sonic control may not suit the pickier of drum producers.
- Heavy covers rock styles, but perhaps is not so strong for metal.
- At this price there's absolutely nothing else to criticise.

SUMMARY

UJAM are perhaps targeting some specific types of music producer with Solid, Phat and Heavy. They may not be for everyone, but UJAM have hit their 'keep it simple' design approach to drum track production right on the head.



Patterns can be dragged and dropped from the upper (dotted) section of the key display to your host and, usefully, retain the MIDI velocity scaling that is controlled by the Mod Wheel.

will appreciate that simplicity and media composers facing deadlines will appreciate the speed. In that latter context, I'd happily use all three of these

patterns, Heavy covers various types of rock including classic, blues, indie, grunge, stoner and garage.

Pass The Audition?

Could the UJAM drummers pass the audition for a role as your virtual session drummer? This is undoubtedly a case of horses for courses. If you are super-picky about your drum sounds, and the degree of control your virtual drum instrument offers, then I suspect products such as Superior Drummer 3 or BFD2 are going to be ultimately more satisfying, albeit at a higher price. Equally, if modern metal is your thing,

products such as GetGoodDrums Modern & Massive might give you something more finely tuned to your sonic needs at a similar price to UJAM's Heavy.

However, that still leaves a potentially large audience of music producers who just want some solid drum sounds and performances in their projects with a minimum of fuss and a modest outlay. In that context, UJAM's Solid, Phat and Heavy are excellent contenders. The workflow is super-simple and, for non-drummers in particular, each of these products will let you create polished drum parts in less time than it takes to mic a snare. Songwriters

instruments in my own commercial work. My only other question is whether UJAM have a percussion-based title in development using the same engine? That would be a very useful addition to the series.

For my money, Solid is perhaps the best, and most versatile, of the bunch but, as UJAM offer a free 30-day trial of any of the titles via their website, you can easily try before you buy and see what might suit you best. **///**

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Mike Crofts - REVIEW Sound on Sound Sept 2019

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IK Multimedia UNO Drum Drum Machine

IK follow up their popular UNO Synth with a drum machine cast from the same mould...

SIMON SHERBOURNE

IK Multimedia's follow-up to the UNO Synth is, sensibly enough, the UNO Drum: a hybrid analogue and sample-based drum machine. It shares the UNO Synth's portable form factor, with a touch-based control panel and a row of knobs on a small, lightweight wedge. It is defiantly old school in sonic character, but has mod cons like patch memory, USB and plug-in control.

Taking the UNO Drum out of its box I was struck by how light and compact it is. I said the UNO Synth was the most backpackable of synths; well you'd barely notice if you stuck one of these in with it. The box also has MIDI jack to DIN adaptors, a micro-USB cable for power and connectivity, and, rather generously, a four-pack of AAs if you want to go cable-free.

IK Multimedia UNO Drum \$249

PROS

- Battery or USB powered.
- Simple and immediate.
- Real-time performance effects.
- 12 voices.

CONS

- Noisy.
- Touch pads often miss manual triggers.
- Limited palette.
- No choke.

SUMMARY

UNO Drum allies some of the analogue and digital character of early '80s drum machines with modern sequencing and effects in a very portable format.



A single mini-jack output carries the mono mix, and there's an input for merging another device like, say, an UNO Synth. Plugging in my headphones I was met with a steady hiss that seems to be an UNO family trait. This doesn't track the volume setting, so the noise floor is lower if you can drive the outputs hard into a mixer, but it's pronounced and distracting even at fairly loud headphone levels. The USB interference that plagued the UNO Synth didn't seem to be as bad here.

Sounds

The UNO Drum has 12 drum channels, split across two rows of six touch pads. Each of the first (bottom) six channels can be switched between a fully analogue drum synth voice and PCM sample playback. The remaining six always play samples. The channels are named and dedicated to specific drum types, and have appropriate synth types and samples. There are five sound presets per channel, with slot 1 being the analogue voice on the hybrid tracks.

Most sounds have just three controls (level, tune and decay), but the analogue

kicks and snares have some extra editable parameters. There are two Kick models. One is in the classic analogue 808 mould, the other has FM with variable tuning and sweep which can emulate 909-type sounds or go into more interesting territory. The Snare is pretty much classic analogue fare again, with noise and body components shaped by a Snap control and a low-pass filter.

The Hats are pleasing enough fizzy dings. Other than the decay you can choose between four tunings. This seems to change the relationship between two sources, and the phase shift between them changes the tone noticeably from one setting to the next. The hats are independent voices, and there doesn't seem to be any provision for channel choking on the unit. Finally the analogue Clap has only a Decay setting.

All other sounds are pre-baked samples: there's no way to load your own. There's a consistent character to the sound selection: old school, lo-fi and crunchy. The 1 and 2 slots are populated with 808 and 909 sets, although sounding as though they've



been sampled into an EMU SP1200.

All the other samples have an early '80s 12-bit (at most) vibe as well: think Drumulator, Oberheim DX, Yamaha RX5 maybe, complete with authentic quantisation noise and alias tones.

A snapshot of the 12 channels can be saved as a Kit into one of 100 slots. This stores each channel's sound type and parameter values.

The Big Crunch

The UNO Drum has two master effects: Comp and Drive, both of which serve to hot up the sound and add impact. When cranked up the Drive rewards you with some nice warm and fuzzy saturation, especially on the kicks.

This does tend to push the noise floor up a storey or two, and exacerbate some sonic oddities that I'd already noticed. Parts of sounds tend to overstay their welcome. Kick 1's FM operator and the analogue snare's body component both leave their VCAs open for a set time, regardless of the Decay. This results in a stray tone from the former and a hummy noise from the latter.

Then there's something going on with the analogue hats. After either is triggered all other sounds seem to open up the VCA for some part of the hat tone, and it provides a very noticeable and unwanted accompaniment.

Patterns

The UNO Drum can be played live via the touch pads, each of which is split into two zones for two different velocities (though this can be disabled). Generally I moan about lack of velocity-sensitive inputs, but two levels is consistent with classic drum

machines. What's harder to swallow is that the pads occasionally fail to respond, especially when playing multiple fast triggers. Trying to play 16ths or rolls with alternate fingers always results in many missed triggers.

Patterns can be recorded in real time (but are always quantised), or entered as steps using the 16 pads along the bottom. Patterns can be up to 64 steps long, and there's a page button for banking bar by bar. All tracks in a Pattern have the same length. Any of the channel sound parameters can be automated in real time, or you can print per-step ('parameter-lock') settings by holding a step and adjusting a value. The automation doesn't support sound mode changes.

There are 100 slots for storing Patterns, and these are saved independently of Kits (again, like an old-school drum machine). Only the Swing amount is saved with a Pattern, so there's no way to save a complete 'session' and recall a Pattern, Kit and effects settings. Likewise, Tempo and the Humanize feature are global settings that will be inherited by any Pattern you load. Often it's useful to be able to juggle patterns without losing your current sound set and vice-versa to try out different kits with a pattern. A link option would be nice, though.

Song mode gives you a handy way to play a sequence of patterns. With Song mode active, the step buttons represent steps in the pattern chain, and you can simply hold a step and dial up a pattern.

Performance Tricks

OK, time for some positivity. Where the UNO Drum is successful is performing, jamming and improvising. This is aided by a few features, not least of which are the Stutter and Roll effects. Stutter temporarily messes with your pattern to create various loops, fills and pitching effects. There are 10 different styles to choose from with an Amount factor that adjusts how many drum channels are included. It's fun and effective and way faster than manually programming variations, fills and builds.

Roll is a more traditional note repeat feature. It affects whichever channel is selected. While held you can adjust the repeat rate from eighths to 32nds, including triplet rolls. As well as these regular rolls, there are four 'fill' repeats that produce more irregular patterns and include some pitch modulation.

Complementing the simple Swing option is Humanize. This adds random

Alternatives

Arturia's **DrumBrute Impact** is almost as portable and certainly as playable as the UNO. It has more sophisticated analogue synthesis and sequencing, but doesn't have PCM sounds or kit saving. Staying in this price range you could also get a pair of **Volcas**, say, the **Beats** and **Sample**. If analogue is less of a key requirement compared to overall power, consider stretching an extra \$100 for the **Elektron Model Samples**.

variance to timing and velocity in a non-repeating pattern.

Beyond these special features, other aspects of the design help with live tweaking and experimentation. For a start, the whole panel stays active during Song mode playback, letting you mess with sound parameters, change kits and trigger Stutter and Roll while a large arrangement plays back.

It's also easy to revert to the saved version of a kit or pattern, so you can go crazy then restore normality. This is great when coupled with the Elektron speciality: a Select All button. This applies all changes you make to all 12 sounds, again great for making a dramatic change during a breakdown before a drop.

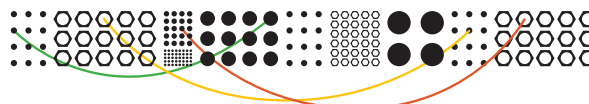
Overall

This is a tricky one. The UNO Drum does have some compelling features, particularly its portability and how easily it can help you generate interesting drum tracks. The idea of an affordable hybrid analogue and PCM beatbox with both kit and pattern memory is really good.

IK Multimedia describe the UNO Drum as "the ultimate beat creation station for anyone and everyone to create warm, punchy, high-quality and inspiring grooves", letting you perform "using the widest sonic palette". This is a bit of a stretch and seems to be missing the real character of the UNO Drum. In fact, sonically it is an instrument with quite a focused style and palette: mashing a little bit of Roland X0X with a large portion of low bit-rate Emu. Which is fine, although some of the noise and weird artifacts seem more like bugs than features.

It is what it is. If you fancy a taste of some classic drum machines in a handy modern package which, alongside the UNO Synth, can easily knock out gritty hip-hop or synthwave, then it could be just the thing. **///**

\$ \$249.99
W www.ikmultimedia.com



MODULAR

Supercritical Synthesizers Demon Core Oscillator & Expander

Eurorack Modules

Supercritical Synthesizers are a new company from Finland and their first product is the wicked-sounding Demon Core Oscillator. It's an analogue oscillator with a remarkable 16 voices, which can be spread and detuned, but perhaps more importantly, and with the help of the Demon Core Expander module, played polyphonically. So, a cloud oscillator with benefits.

The base module contains the oscillator itself. This will work without the Expander and will provide you with 16 sawtooth or pulse waveforms which can be detuned and spread with unison and octave stacking. The oscillator can be tuned using coarse and fine controls, and there is a handy LED indicator to show when you hit a C note (it also doubles up to show MIDI activity when the Expander is connected). The Spread control adjusts the detuning, allowing anything from subtle unison detune to complex equally spaced microtuning. Another control labelled Core Stability will introduce pitch drift per voice, which can sound like anything from a gentle chorus effect to a swarm of angry bees.

In terms of connectivity, there is of course a 1V/oct input, which is joined by FM, Spread and Stability CV inputs. The Spread input can also double up as a PWM control when the oscillator is set to pulse wave. An input labelled Sync Trig will, when triggered, re-sync all the oscillator voices. This is useful for sounds where you might want a very consistent attack transient (feed the same trigger you use to trigger your Amp envelope). Typically this is used on bass sounds, but you can get some interesting rhythmic effects by sequencing it too. The last connection socket is the mono audio output.

The number of voices used by the oscillator can be adjusted with a single button and a star of 16 LEDs shows you just how many voices you're currently using. Successive presses

of the Voices button rotate through 1-2-3-4-5-6-8-10-12-16 voices. As well as choosing the number of voices, you can also decide the octave stack value. For example, if you had eight voices and a stack value of two, you would have two sets of four voices an octave apart. Eight voices in a four-octave stack would give you four octaves of two voices each. You can even combine numbers which don't divide precisely, the oscillator will simply do the maths for you and assign the voices as best it can. It's a very visual, flexible and easy way to allocate the voices in different ways.

The Demon Core Expander connects to the main module via a supplied ribbon cable. Power is transferred from the main module, so no additional power cable is required, although will draw more power from your system. The main feature on the Expander is the addition of polyphony, by allowing a MIDI input on TRS mini-jack. This means you can play the 16 voices just like a normal synth. MIDI channel can be set from the front panel, as can other MIDI-related functions such as pitch-bend range, a hidden LFO you can use for vibrato via modulation

wheel, and even variable polyphonic portamento.

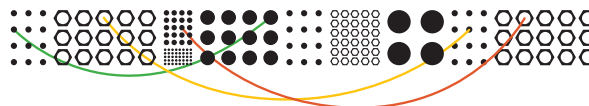
The Expander also gives you 16 VCAs with ADSR for each voice, effectively making it a complete synthesizer voice, minus the filter. The behaviour is altered using the Stream/Gate mode button, with Gate mode employing the ADSRs and VCAs and Stream mode acting more like a normal oscillator with each voice droning indefinitely. In this mode, you might utilise the Gate output, which will output a trigger or gate signal for each MIDI note, allowing you to use other VCAs and envelopes in your system to create a paraphonic synth voice. It's a neat solution which turns almost any existing modular into either a full polyphonic or paraphonic powerhouse. Furthermore, the voices can be divided up to form various combinations of polyphony and stacking. For example, you might have seven-note polyphony with two voices per note, using up 14 voices in total. Or you might choose four-note polyphony with four-voice stack. As long as the total doesn't exceed 16, you can come up with almost any combination. Not only that, but you can also add octave stacking to the mix. For example, four-note polyphony, two voices per note and a two-octave stack per note would use up all 16 voices.

The ADSR works with only two front-panel controls, Attack and Release. Pressing the Function button on the main oscillator module will turn these into Decay and Sustain controls. It's not quite as nice as having dedicated controls for all four stages, but it works well enough. One nice touch is that you can switch the VCAs to work in legato or re-trigger mode.

If you don't want to use MIDI, you can still get four-voice polyphony using the four FM inputs. This will control the first four voices. If are using MIDI, however, these inputs aren't wasted and can be used to control other parameters: Sync Time, Sync Random, Octave Stack Size and Octave Stack Interval. One neat feature of having both main



Supercritical Synthesizers Demon Core Oscillator
 Size: 12HP
 Current: +12V 120mA
 -12V 20mA



Supercritical Synthesizers Demon Core Expander

Size: 8HP
Current: +12V 130mA
-12V 30mA



module and Expander is that the MIDI input and main 1V/oct input work together. So with the Expander in Stream mode, use the MIDI input to play a chord, and then use the 1V/oct input to transpose that chord. Ta-da! Instant chord memory. Now patch in some control voltages to the four individual FM inputs on the Expander and

you can further transpose the individual notes of the chord. Perhaps you could add a sequencer which transposes each note up or down an octave.

The possibilities of the Demon Core Oscillator and its Expander module are many. I think it would make an excellent heart to a polyphonic or paraphonic modular setup. You could use the MIDI input on the Expander and the Gate mode envelopes to give you a traditional polysynth. Adding in an external filter would make it paraphonic. Alternatively, you could set the polyphony to four and use the four FM inputs on the Expander to control four voices via 1V/oct pitch CV giving you direct analogue control over four voices at once.

Do you need the Expander? Well, if you want to do polyphony, then yes. The main oscillator module is still an excellent cloud oscillator, but I feel that having the combination and being able to play 16 voices polyphonically will be the biggest selling point for most people. Occasionally it can feel complex, there is a reliance on button combos to enable modes. Frequent checking of the excellent manual is probably going to be unavoidable until it's committed

to muscle memory. Overall, though, the Demon Core Oscillator is an impressive debut from Supercritical Synthesizers.

It represents another step towards truly polyphonic modular systems and sounds really rather excellent. The build quality and styling are equally impressive. I look forward to more from Supercritical Synthesizers. *Rory Dow*

\$ \$510 & \$410

W www.supercriticalsynthesizers.com

DPW Design MÖG D-2

Eurorack Module

Described as a four-band preamp, overdrive and distortion module, the MÖG D-2 puts guitar pedal power at the heart of your Eurorack. Further crossover is evidenced by twin inputs; joining the expected mini-jack is a quarter-inch Hi-Z input, enabling you to process guitar, bass or line-level instruments (such as synths) within the modular environment.

Operation is simplicity itself: connect your source, flip the switch then set the levels for each of the bands, adjusting overall gain and output to taste. With CV control only of its on/off status, the D-2 is about playing, listening and tweaking those well-spaced knobs. As a useful reference, the Volume LED lights when the output breaches 6V, while a Gain LED ensures you don't overdrive the input and therefore lose the benefits that lie in store.

From the first moments, your ears tell you this is no ordinary distortion. The old-school analogue design does a credible impression of a quartet of tube amps in parallel, with Gain acting as the overall drive control. By dividing the distortion into Low, LoMid, HiMid and High circuits, the D-2 offers a rare level of fine-tuning, rendering other distortion effects crude and unsubtle in comparison.

The Low boost is a suitably monstrous tool for toughening up kicks or beefing up basses. Next,

the LoMid covers a warm, full distortion suitable for adding body to practically anything. In contrast, the HiMid adds zap or shape; I found small to moderate amounts of it ideal for giving analogue synths a little extra presence. Finally, the High range adds bite and clarity wherever it's needed — even to sources such as muddy kicks or dull basses.

In use, this module can supply a mildly dirty four-band EQ with a slightly compressed, fat quality. Simply set the Gain low then compensate with the Volume control. As you start to crank it up, you selectively unleash a distortion that is both musical and refined — not terms typically applied to distortion effects. Perhaps because of its guitar pedal inspiration, the D-2 is highly responsive to dynamics, providing boost without mud and mush. Lastly, I found it a great companion for other, more pristine modules, eg. it worked wonders in the send/return loop of a 4MS Dual Looping Delay.

As I mentioned earlier, the only CV control is an on/off switching CV input, which activates the effect on receipt of 1V or more. The input easily copes with rates between DC and audio, the results being an irregular but pleasing variety of glitching. By feeding a Yamaha Reface piano into the quarter-inch input and modulating the on/off status with a wide-ranging VCO, my piano began spitting out everything from edgy tremolo to pseudo-ringmod-type noises. Incidentally, use of the Hi-Z input doesn't disable the regular mini-jack, which is helpful.

While some users might wish for CV control of each band, there's something to be said for at least one module in your case that is simple, hands-on and just sounds wonderful. The D-2 has more sweet spots than a dalmatian dipped in sugar; once experienced, it will probably find its way into pretty much everything you do. Given it also serves as a preamp to bring your guitar, bass or synth up to Eurorack levels, I have no hesitation in giving it an unqualified thumbs up.

Paul Nagle

\$ £208

W www.dpw.se



DPW Design MÖG D-2

Size: 10HP
Current: +12V 25mA
-12V 15mA

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Steinberg Cubase Pro 10.5

Music Production Software

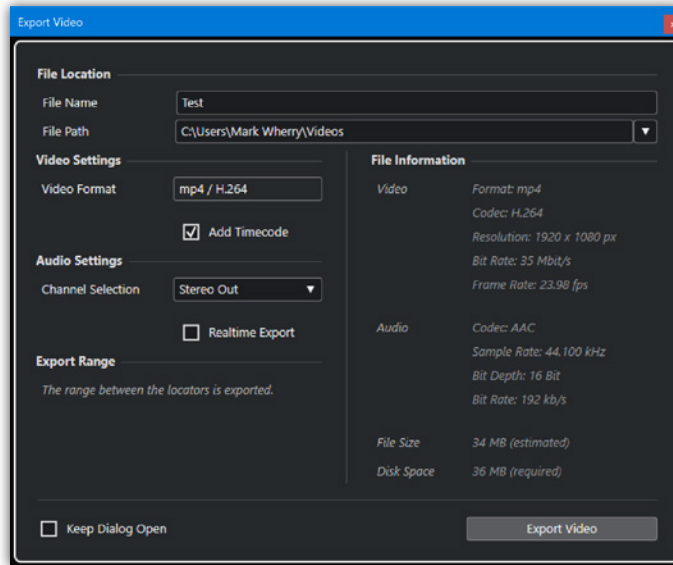
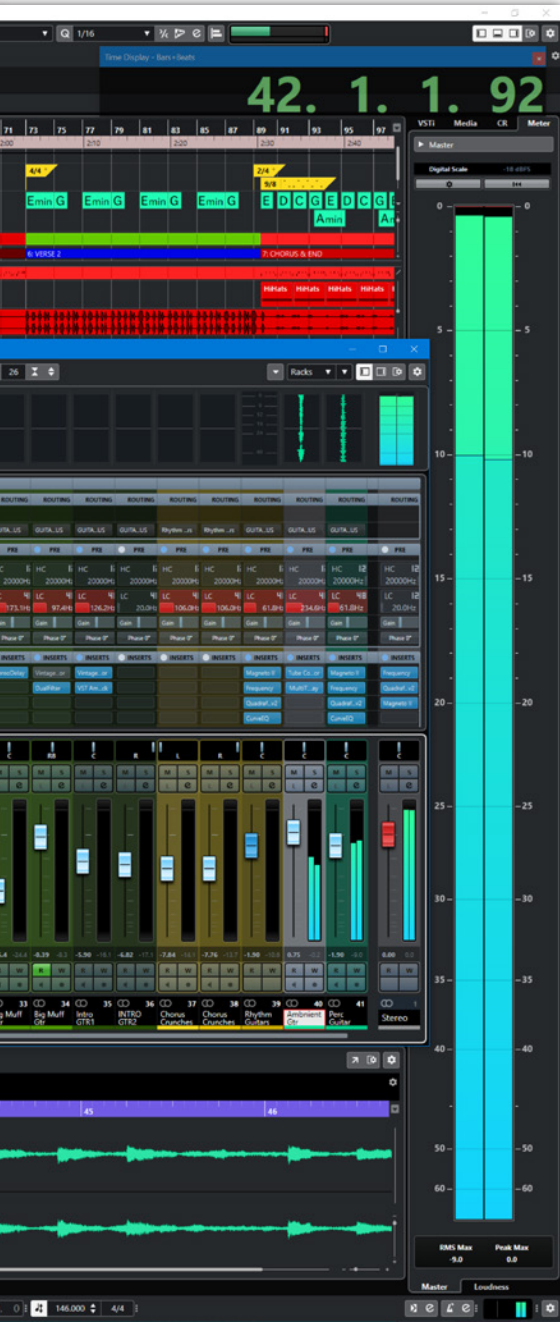
There's something for everyone in the latest version of Steinberg's flagship music production package.

Cubase Pro 10.5 in all its glory. Note the new MultiTap Delay effect included with the Pro and Artist editions, and the coloured channel strips in the mixer.



MARK WHERRY

Christmas was coming, the goose was becoming rather overweight due to a lack of dietary restrictions, and Steinberg released what has become the now-traditional end-of-year update to Cubase. Given that the last release of the company's Advanced Music Production System was version 10, launched on November 14th 2018, Steinberg's past roadmap strongly suggested 2019 would be an x.5 year. And indeed, the company didn't disappoint, unveiling Cubase 10.5 almost a year to the day on November



The new Export Video window makes it easy to create a new video file with audio from the active Project.

Unlike the original Replace Audio in Video command, which added audio to an existing video file, the new Export Video command offers much more flexibility by creating new video files based

on the position of the locators. To export a video, simply set the required range using the left and right locators and select Video from the File / Export submenu. In the Export Video window, set a name and path for the video file to be created, choose the stereo output to use for the audio to be included in the video (and whether that audio should be rendered in real time or not) and click the Export Video button. Et voilà!

13th with no shortage of new features and improvements. Available in the usual Pro, Artist, and Elements editions, this review focuses primarily on Cubase Pro 10.5 – the inclusion (or lack thereof) of certain new functionality in other editions will be mentioned as necessary.

Exporting Video, Ten-Point-Fivenally!

One of the most requested features in Cubase has been the ability to export a video file containing both the audio mixdown and accompanying video from a Project. Long-time Cubase users might remember the Replace Audio in Video command, which allowed you to replace the audio within an existing video file, and which was dropped when Steinberg introduced a brand-new video engine in Cubase 9.0.30.

Despite some consternation, this new video engine was necessary — and arguably overdue — because Steinberg's previous cross-platform video engine utilised Apple's QuickTime technology. The Windows version of this had been withering on the vine for years, and was finally put out of its misery in 2016. The new engine was a huge step forward in terms of supporting modern containers and codecs for playback, but the capability to export video was, to paraphrase a Steinberg support article, pushed back as a planned development for future updates. It was promised for Nuendo 10, released in April 2019, but missed the initial 10.0 release, and eventually showed up in Nuendo 10.2 around six months later. Finally, it's now available to Cubase users too.

A new video file will be created in an MP4 container, using the H.264 codec to render HD (1920x1080) video (upscaling or downscaling the resolution as necessary from the source resolution) at the same

Steinberg Cubase Pro 10.5 \$560

PROS

- The ability to export audio to video is restored.
- A much-improved way of importing tracks from other Projects.
- Everything you could ever want from MIDI Retrospective Recording.
- Improvements to Macro creation, selection tools, working with the built-in EQ and more.

CONS

- More advanced options when exporting video could be useful.

SUMMARY

It could be argued that the x.5 updates in Steinberg's roadmap are designed more to keep existing Cubase users satisfied than to tempt new users. However, Steinberg's Advanced Music Production System remains at the top of its game, facilitating many unique cross-platform workflows.

» frame rate as the imported video. For audio, the AAC codec is employed to compress audio at a 16-bit resolution using the Project's sample rate, although only 44.1 and 48 kHz sample rates are supported.

I tested the Video Export command by importing a MOV file into a Project. The video was encoded with Apple ProRes 422 SD (1280x720) at a 23.976 frame rate, and the feature worked as advertised, which was great. However, I couldn't help but think I'd like to see more options for choosing additional containers and both video and audio codecs, with support for 24-bit resolutions and other sample rates. The lack of options on offer at export stands in sharp contrast to the wide array of video formats supported for import and playback. Perhaps we'll see more choice in future versions, although maybe I'm being picky: even in professional contexts, sending a file created using the current Export Video offering gives you something that can be played back by almost any system and will be more than acceptable to almost any client.

One additional thing that can be optionally included in a video exported from Cubase is a timecode burn-in based on the current Project time. This could be useful for reference, or if the source video doesn't include such a burn-in already, although there aren't too many options to define how the timecode appears on the video. Based on the Video Player settings in the Studio Setup window, you can set how the timecode is horizontally aligned, resulting in white numbers in a black box being displayed. Again, more options would be nice: since you can add burn-in to the video, why not offer size, font, colour, and vertical alignment options as well? Or take it a stage further and allow dates to be added, plus the source video file name and custom text if required?

For such a useful and advanced feature, it's perhaps surprising Steinberg decided to include Export Video not in just Cubase Pro, but Artist and Elements as well — although given the number of users who add audio to video clips these days, it probably makes sense, giving Cubase Elements even more allure for those looking for powerful features at a cheaper price point.

In Retrospect...

The seemingly innocuous Retrospective MIDI Record command has been both renamed and enhanced in Cubase 10.5. Previously, this relatively straightforward feature captured any incoming MIDI data

while the sequencer's transport was either stationary or playing back, maintaining the timing data in both states. If you happened to play something unbelievably brilliant, rather than kick yourself for not being in record, you could simply select Retrospective MIDI Record, and a new MIDI part containing the captured sequence of Events would be created on the selected

MIDI or instrument track. Easy. In fact, many people I know prefer to use this command rather than putting the sequencer into record, although I find myself being more old-fashioned in this respect.

MIDI Retrospective Record functionality is now accessed through several different commands, which are available in a few different places, starting with the Transport »

Import Tactics

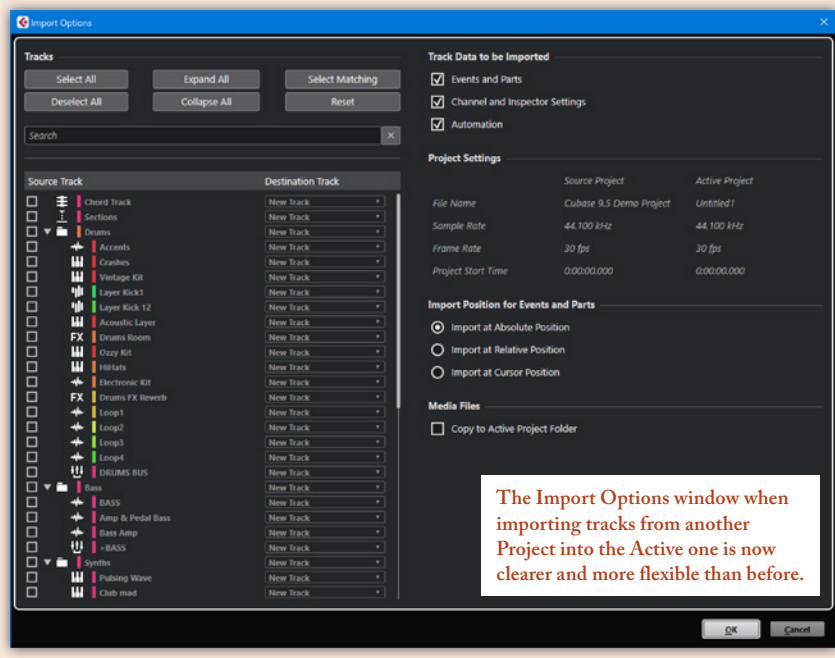
Cubase Pro's Import Tracks from Project feature has been dramatically improved in 10.5 compared with its previous incarnation in version 10.0.50 (which is the same one still featured in Nuendo 10.2). Firstly, its appearance is more elegant, with the list of source tracks in the Project to import now presented hierarchically based on the folder structure of the source Project. As before, you can select or deselect all of the tracks, but now you can click on a folder to select all tracks within that folder (before deselecting individually as required), and folders can be individually expanded or collapsed.

It's possible to filter the display of source tracks with a new search field, which is handy when importing from a large Project. But one of the neatest additions is a new way to manage how source tracks are assigned to destination tracks in the Active Project. Previously, you could select New Track (which is still the default), whereupon a new track is created for the imported track, or you could pick an existing track from a pop-up menu. However, this process can now be automated with the new Select Matching command. For example, if you have a source audio track named Accents, and an audio track in the Active Project also named Accents, you can

enable that track to be imported, click the Select Matching button and Cubase will automatically assign the source track to the destination of the same name. This works for all tracks enabled for import, although the track configurations need to match.

As before, Project Settings are displayed for the source and Active Projects, and you can select whether media files should be copied to the active Project folder (and sample-rate converted if necessary) to keep everything in one place. But Cubase Pro 10.5 now lets you choose what types of track data are imported, if any (as opposed to everything, as before), and these include Events and parts, channel and inspector settings, and automation. It's also possible to set the position at which Events and parts are imported, either as an absolute or relative value, or at the position of the Project Cursor.

You could argue that this command is still missing a few tricks, such as the ability to distinguish audio and video assets rather than copying both types of media files to the Active Project Folder. Still, to be fair, Cubase Pro 10.5's updated Import Tracks from Project feature represents, as I wrote in the first paragraph of this box, a dramatic improvement over what came before.





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Tap Dancing

What would an update to a music creation application be without a new plug-in or two? The Pro and Artist versions of Cubase 10.5 include a new multi-tap delay effect called, well, MultiTap Delay — Steinberg marketing have done it again! — which, despite the prosaic name, is a useful, fully featured, and decent-sounding addition to Cubase's included arsenal of plug-ins.

Each MultiTap Delay preset can be assigned one of five different delay characters: Digital Modern, Digital Vintage, Tape, Crazy and Custom, which is selected automatically if you adjust the parameters of another character. You can create and manipulate up to eight taps, and there are three separate effects sections — Loop, Tap and Post — each with slots for six effects modules. Loop effects feed the output back into the delay input, Tap effects allow you to apply difference parameters for each of the six chosen effects modules, while Post effects apply to the output of the plug-in.

Cubase Pro and Artist 10.5 also include

the new Padshop 2, which merges the original Padshop bundled in earlier versions of Cubase with the stand-alone Padshop Pro and adds a few new tricks. Padshop can create complex, evolving sounds that, as the name suggests, are particularly useful when performed as pads. Padshop 2 adds a second synth engine (a spectral oscillator) and over 100 new presets and samples, and you'll also get a new arpeggiator, filter, and other effects, enhanced modulation, the ability for Cubase users to now import custom samples, and the obligatory darker, 'refreshed' design, although it still looks pretty red to me.

I rather like Padshop and its sonic aptitude in facilitating the creation of performable, evocative sounds, despite the dangers of cranking up the reverb and venturing into the land where dairy produce is combined with rennet and a dash of bacteria. The interface could possibly use a less utilitarian approach, because while access to the controls is undoubtedly direct, the addition of a 'simple

mode' wouldn't go amiss if you're lazy or new to the instrument.

Like Padshop Pro, Padshop 2 is also available as a plug-in for any host supporting VST, AU and AAX instruments from the Steinberg online shop for £110 — updates from Padshop Pro are priced at £25. Therefore, it should be pointed out Steinberg state this update is incompatible with the version of Padshop previously bundled with earlier Cubase releases. So, for existing Cubase users looking to update to Padshop 2, the company recommend updating to Cubase Pro or Artist 10.5.

Cubase Elements users haven't been forgotten, though, as 10.5 of this iteration now includes three effects previously only available in versions higher up the product ladder: Stereo Delay, Roomworks (a reasonable algorithmic reverb with flexible controls, although Steinberg's more recent algorithmic reverb, REVelation, included in Artist and Pro, is considerably better in terms of quality), and DeEsser.

» menu. There's now a MIDI Retrospective Record submenu, and the closest command to the original is called Insert from All MIDI Inputs, which is assigned the same default key command (Shift and the * key on the numeric keyboard) as before. This results in the captured Events being inserted on the selected track; but, where previously the command could only be used once, this is no longer the case. Now you can select the command as many times as you like, inserting the Events on the same track or different tracks, with the buffer only being reset if Cubase either begins receiving new Events or you select Empty All Buffers from the submenu.

You'll notice the command is named Empty All Buffers and not Empty Buffer, and this is because — in addition to there being one overall buffer, as before — each MIDI and instrument track now has its own, independent buffer for retrospective recording. Using the new Insert as Linear

The new MIDI Retrospective Record commands can be accessed from a Transport submenu, or a new menu from the Basic Settings section in the inspector.

Recording command you can insert Events played on the track selected at the time of capture, which is similar to Insert from All MIDI Inputs except that Events can only be inserted on the track that was selected when they were captured: if a different track is selected, Insert as Linear Recording is unavailable.

To make this a bit more obvious without having to access the Transport menu, the Inspector's Basic Track Settings section now features a new Retrospective Recording pop-up menu at the bottom that provides access to the appropriate commands (sans Insert from All MIDI Inputs) for that track. This pop-up will be greyed out and inaccessible if there are no Events to specifically insert on that track. And, in addition to being able to insert a linear recording, if you were playing back a project in Cycle mode, MIDI Retrospective Recording can also insert the captured

Events as a cycle recording, which is rather neat.

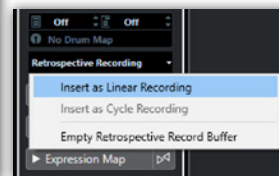
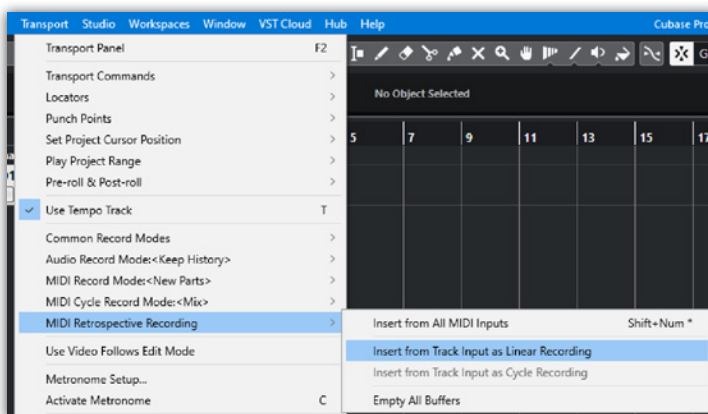
Overall, I think it's fair to say that if you use MIDI Retrospective Recording, Cubase 10.5 has got you covered — and, if you don't, this release might convert you.

She's Like A Rainbow

MixConsole windows are more colourful in Cubase 10.5: it's now possible to add a background colour to channel strips in the same way you've been able to with track headers on the Project window for some time. This invariably makes the channels look a bit like those on Pro Tools' mixer, which has enjoyed this ability for over a decade, but it is a welcome method of making MixConsole windows in Cubase a little easier to navigate, avoiding a sea of blue-grey on your display.

Speaking of colour, the default background colours for active send slots has changed. Previously, active post-fader sends had a light, cyan-green complexion, where active pre-fader sends adopted a darkish-blue hue. I didn't have a problem with this, to be honest, especially as you could modify these colours in the Preferences. But now, post- and pre-fade send slots are orange and light blue

respectively, and the default palette for similar controls has been made generally brighter. »





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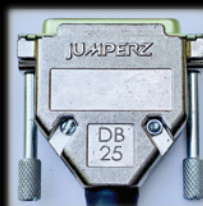
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UNVEILING

» Aside from colour, you'll discover some improvements in the way the built-in EQ is used and displayed in the Channel Settings window. Firstly, there's now a more-obvious settings button where the visibility of EQ controls can be switched separately to their appearance as sliders or knobs. More significantly, there are some enhancements concerning the useful FFT that can be shown behind the curve in the EQ Display area. This can now display either the signal before the EQ processing or a post-EQ, peak-hold curve, with variable transparency.

Cubase Pro 10.5 also provides a new a Comparison Curve that, when enabled, shows two spectral curves overlaid within a single EQ Display: the Reference Channel in blue and a Comparison Channel in orange. The Reference Channel is the selected track/channel (or the one from which the Channel Settings window was opened), while the Comparison Channel (including the master output) can be selected from a pop-up menu. The focus of the Reference and Comparison Channels can be toggled to determine the EQ curve being edited, and, usefully, these Select buttons also offer duplicate Solo buttons.

This feature is clearly designed to make it easier to EQ one track/channel against another, although no doubt some will find less-obvious use cases for employing it. Now, how about adding a secondary Comparison Channel?

Macro Bionic

As always, supplementing the headline features of a new release are a number of less glamorous-sounding improvements scattered throughout the update, and it's these that often turn out to be the most

The EQ display in Cubase Pro 10.5 can now superimpose the EQ curve from Reference and Secondary EQs in the Channel Settings window.



System Requirements & Catalina Compatibility

It's worth observing Steinberg's requirements when upgrading to or purchasing a new Cubase 10.5 licence. The company state that Windows users should be using 64-bit Windows 10 version 1903 or version 1809, while Mac OS users need to be running Mojave (10.14) or Catalina (10.15). Mac OS 10.15 is named after Santa Catalina Island, located about 30 miles off the coast of Long Beach in Southern California — and, arguably, Mac OS applications are becoming more like islands, due to increased security measures such as what Apple refer to as the Hardened Runtime.

In order for an application not distributed via the App Store to be executed on a Mac running Catalina, it needs to have the Hardened Runtime enabled and be submitted to Apple for Notarization [sic] where it will be automatically vetted for malicious content. Once an application has passed this notary process, a developer will receive a notarisation ticket to be included with that application for distribution.

Notarisation is but one factor of the Hardened Runtime, which aims to protect both system and application integrity using many different techniques, such as restricting access to certain runtime capabilities and resources. The latter, for example, would include access to Location Services, Apple Events and a user's Address Book, Calendar and Photo Library — more pertinent to Cubase users would be access to Audio Input via Core Audio and the built-in camera. Should an application want access to these resources, they can be enabled as Entitlements within the settings for the Hardened Runtime.

That's why Mac users will be required to give permission for Cubase to access Audio Input the first time the application is launched on Catalina, or to use the camera in VST Connect.

As you might expect, the introduction of Hardened Runtime brought compatibility issues with music and audio applications, plug-ins and drivers, with most developers initially cautioning users on upgrading. However, this situation has greatly improved since Catalina's release last October and Cubase 10.5 is fully compatible. For more information, Steinberg offer a support page with a detailed breakdown on Catalina compatibility with the company's products.

As before, Cubase Pro and Artist also require a USB eLicenser and thus a spare USB port in which to insert it. Disappointingly, the design of the USB eLicenser hasn't really changed in the years since Steinberg acquired Syncrosoft, the company responsible for the copy protection technology in question. It's always been a bit flimsy for my liking compared to, say, the original iLok and the latest one-piece metal iLok 3. Strength aside, such devices still use USB-A connectors, and it seems about time Steinberg offered a USB-C eLicenser. I'm singling Steinberg out here only because this is a review of a Steinberg product, but it's a minor pain some users will have to factor in the use of USB-A to USB-C adaptors — remember to pack one in your laptop bag!

In terms of hardware, Steinberg recommend a minimum of 4GB memory (with 8GB or more being preferable), and 30GB of free disk space — the download itself requires nearly 22GB.

useful. In the case of Cubase 10.5, this includes enhanced graphical performance, and simple tweaks like the ability to name folder tracks when adding them from the Add Track windows.

There are also welcome and overdue improvements to the way Macros are created in the Key Commands window, in Cubase Pro and Artist 10.5. This window and its functionality haven't changed since Cubase SX was released back in 2002, and although the changes aren't perhaps earth-shattering, it's nice to see somebody at Steinberg still cares.

The Macros section of the Key Commands window used to be fixed at 10 rows of text, even though the

window itself was resizable. A new divider control between the Key Commands and Macros views now allows this to be varied. Better still, there are now additional buttons grouped between actions that apply to Macros and those that apply to Commands within Macros, replacing the three confusingly general-purpose buttons of the past. The Macros group consists of New and Delete buttons, along with a new and extremely handy Duplicate button. Previously, if you wanted to create a Macro based upon another, you had to start from scratch, creating a new Macro and adding the required Commands all over again. Not any more!

Test Spec

Steinberg Cubase Pro 10.5.0.
Microsoft Surface Studio with 2.7GHz i7 processor and 32GB memory, running Windows 10 Pro version 1903.
Tested with RME Babyface Pro.

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Working with Macros has been made easier with new, sensibly grouped buttons.

» The Commands group offers four buttons, the first two allowing you to Add and Remove Commands in a Macro and the second pair making it possible to move the position of a selected Command up and down in the list of Commands within a given Macro. Managing the order of Commands used to be a real pain, and you'd often have to delete Commands ahead of where you wanted to add the new one before adding them back again.

Multi-tasking

Another Pro- and Artist-only improvement is the ability to combine the Object and Range Selection tools in the Project window such that they're both active simultaneously, without having to switch between the two. This Combine Selection Tools mode is toggled and indicated via a new button added to the left of the Select tool button, and is usable so long as a track is set to at least a height of two rows. Now, when you hover the mouse in the upper half of a track, Part or Event in the Event Display, the Range Selection tool is employed, whereas if you click in the lower half the Object Selection tool is used instead.

The Glue tool is also now quicker to work with, featuring its own dedicated Edit command. Previously, if you had two

adjacent MIDI parts you wanted to merge into one, you'd have to select the Glue tool and click on the first part before reselecting the previous tool. Now, you can simply select the first part and choose the Glue command from the Edit menu (or, better still, assign a key command) and the two parts are merged without any need to switch tools. You can also, as would seem logical, select multiple consecutive parts and use the command to merge them all, just as you could and still can with the Glue tool.

When it comes to offline audio processing, Cubase Pro 10.5 borrows from Nuendo in that you can now set whether audio is normalised using Peak Normalization, where the level is set in

dB, or the new Loudness Normalization, where the level is set in LUFS (Loudness Units Full Scale). This complements the loudness master meter inherited from Nuendo 6 in Cubase 8, and is useful if you need to normalise a piece of audio to broadcast standards.

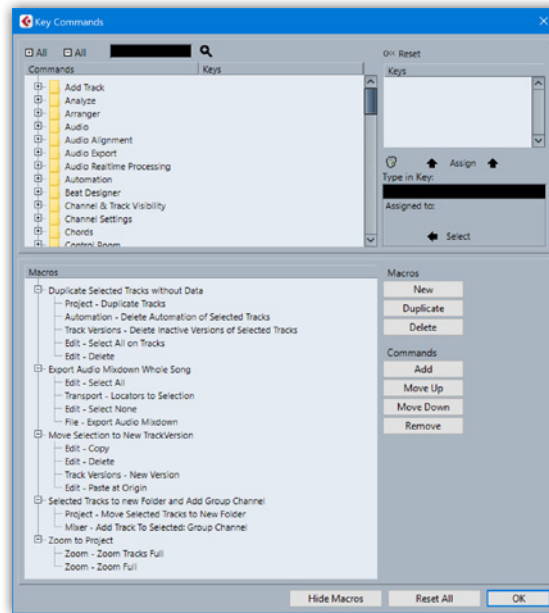
The Safe Start Mode option, accessed by holding down the Ctrl, Shift and Alt/Option keys just after starting Cubase, has been made easier to understand. For example, you now choose using radio buttons whether to start Cubase with your existing preferences, to temporarily disable them and run with the defaults, or to delete the current preferences and begin afresh, before clicking the OK (or Cancel) button to proceed, rather than these options being available as separate buttons. This is to accommodate a new feature allowing you to choose whether third-party plug-ins should be deactivated when starting Cubase, which could be helpful when troubleshooting (or if you forgot your iLok).

Finally, closing the Active Project will no longer automatically activate any other loaded (and thus deactivated) Project automatically. This is actually quite a big deal when you have multiple large Projects open at the same time, such as different versions of the same piece, for example. If they contained a complicated MixConsole configuration with a high track count and many effects and instruments, you used to spend minutes in order to wait for what was previously the wrong Project to activate before moving on.

It Ain't Half Cubase

It sounds like a cliché, but Cubase 10.5 really is one of those updates where there really is something for all users. While some of the new features — notably the revised Import Tracks from Project window, the ability to colour MixConsole strips, and perhaps the Combine Selection Tools mode — are, shall we say, 'inspired' by competing products like Pro Tools, this is no bad thing. Sometimes the wheel is worth rethinking, but equally there isn't always a need to reinvent it, and I think in this latest update Steinberg have achieved a good balance in observing the competition, meeting the needs of users, and keeping Cubase true to itself. ■■■

\$ Cubase 10.5 Pro \$559.99; Artist \$309.99; Elements \$99.99. Upgrades and crossgrades available.
W www.steinberg.net



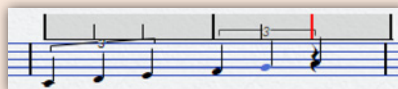
High Scoring

Steinberg continue to enhance Cubase Pro's Score Editor, and version 10.5 introduces some nifty new tricks that wouldn't go amiss in other score editors. For example, you can now transpose selected notes using the scroll wheel on a mouse if the appropriate preference is enabled. This feature also works with a trackpad's scroll gestures, although I found it to be considerably less precise and harder to control. It would be even nicer if you could move notes horizontally by holding down the Shift key whilst scrolling, similar to how the Event Display can be scrolled horizontally in many of Cubase's windows. But that's me being picky.

One improvement when moving notes or inserting a note in musical time, though, is the optional ruler that can now appear above the staff when performing these operations. Enable the Show Bars and Beats Positions

When Inserting Notes preference, and you'll see a ruler with the current destination position indicated by the appropriate division line coloured in red.

An Event Movement Restrictions button has been added to the Score Editor's toolbar, providing easier access through a pop-up menu to three options previously available as preferences: Keep Notes within Key, Snap Slurs to Notes and Keep Crescendo Symbols Horizontal. These are joined by new a Snap Rests and Repeats Vertically option, which pretty much does what it says on the tin.



The Score Editor can now optionally display a useful bars- and beats-based ruler when creating or moving notes.

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Capo Elevator *Guitar Accessory*

The Capo Elevator is the smallest product I've ever been sent for review, and also the product with the most specific function. The brainchild of Tim Loe, it's designed to counteract a fairly well-known issue that can arise when recording steel-strung acoustic guitars.

Most serious acoustic players, especially those who play in open tunings, make fairly extensive use of capos. However, clamping the strings to the neck with a capo doesn't just raise the pitch of the instrument: it also changes the sound of the 'open' strings, making them sound a little damped and less rich in overtones. In fact, for obvious reasons, it makes them sound less like open strings and more like fretted strings.

This isn't necessarily a bad thing. You don't always want a contrast in sound between open and fretted strings, and some guitarists actually prefer the less 'zingy' sound of capoed strings, to the point where they habitually tune down a semitone in order to use a capo at the first fret. Equally, however, there are times when any reduction in the brilliance of the open strings feels like a loss, and it's for these situations that Tim has invented the Capo Elevator.

Supplied in a neat metal box reminiscent of a tobacco tin, the Elevator is a precisely shaped brass rail that sits

on top of and behind the fret, raising it by perhaps half a millimetre or so. It's positioned using a small, detachable wooden handle and held in place by the pressure of the capo itself, which is not included. Most types of capo should work, and I had no problem using the Elevator both with a G7 capo and a more traditional screw-based model. Attaching the Elevator feels a little fiddly at first, but you soon get the hang of it. It is probably best considered a studio-only tool, though, as it adds to the time involved in moving a capo from fret to fret, and would be easy to lose or misalign on a dark stage.

The Capo Elevator is optimised for a particular fret size and radius, but I don't think this is something to worry about too much unless your guitar is a long way out of the ordinary. My Larrivée OM-3 has an unusually flat fretboard, and although the Elevator was visibly more curved than the frets, it proved flexible enough to sit comfortably in place once the capo was attached.

Does it make a difference? To the feel and intonation of the guitar, none whatsoever, which is as you'd hope. To the sound, yes, albeit a difference that can be subtle and sometimes elusive. I tend to find that there's some variability in the effect of the capo itself, because you



never quite attach it the same way twice, and it's quite hard to do reliable A/B tests when you have to remove and reattach it each time. Nevertheless, I consistently felt that the sound with the Elevator attached was livelier and less choked, and this was a difference that was audible both in the room and on recordings. It's not a radical, night-and-day transformation — in the scheme of things, perhaps on the same sort of level as swapping out one small-diaphragm mic for another — but it's one that you probably can't achieve any other way, and which has no negative aspects apart from the additional time and effort involved in attaching the Elevator. This is a highly affordable device that does what it sets out to do, and if the issue that it tackles is one that has ever frustrated you, the relatively small investment seems well worthwhile. *Sam Inglis*

\$ £17.99 (about \$23) plus shipping.

W <https://capoelevator.com/products>

OLLO Audio HPS S4 & S4R *Headphones*

Most new headphone manufacturers target the booming hi-fi and consumer markets, but OLLO Audio are an exception. They've made their entrance with two models that are explicitly aimed at audio engineers, and their major selling point is said to be a flat frequency response, achieved without the use of DSP correction. The OLLO headphones are also intended to be fully field-serviceable, with every major component available as a spare part that can be replaced by the user.

Both the S4, aimed at mix engineers, and the S4R, intended for tracking duties, are passive headphones that employ conventional moving-coil dynamic drivers. The key difference between the two models is that in the S4R, these drivers are housed within closed earcups, whereas the S4 is an open-back design. In

fact, apart from the panel that forms the outer side of the earcup, the two appear to be otherwise identical. Both are supplied in a simple cardboard box, with a soft bag but no rigid carry case.

The documentation includes both individual test plots for each unit, as supplied with good mics, and some curious warnings about not plugging the phones in with the earcups face down on a flat surface.

Unusually, the OLLO headphones are not 'handed': the earcups are perfectly circular, and the frame is symmetrical front-to-back. Which earcup is the left one depends solely on which way round you connect the detachable cable. This is a very nice braided, uncoiled affair that divides into two about three-quarters of the way along its length before attaching to the earcups using 2.5mm mini-jacks. Unfortunately, though, the letters L and R on these jacks are printed in a light grey which is almost indistinguishable from the silver body of the connector, making it needlessly difficult to determine which side is which.

The distinctive cable is just one part of



a strong visual aesthetic, which is carried further in features like the walnut earcup surrounds and artificial leather headband. The design and construction give the S-series phones a pleasingly solid and somewhat handmade feel, and a generally comfortable fit, which comes at the cost of one fairly significant annoyance. The frame on which the earcups and headband sit is formed of two rigid wire hoops, which are alarmingly resonant. Even a slight bump on the frame is loudly audible, and when I deliberately 'pinged' one of the hoops, it took the ringing more than 15 seconds to decay to the point where I couldn't hear it any more. There were a number of times during the review period where I felt the need to damp the hoops with my hand to check that they weren't contributing to some sort of low-mid-range problem.

The published impedance of both S-series models is 32Ω, suggesting that they should be easy to drive from a low-power source such as a laptop's built-in headphone output, and so it proved in practice.

When I received the S4s, I was about to start some revisions on a mix in response to feedback from the artist, and so put them through their paces straight away. The results were not quite as I'd expected, and when I auditioned the resulting mix on other systems, it sounded a bit odd. Some A/B comparisons with other phones confirmed my initial impression that the S4s have a noticeable suck-out in the mid-range. Above 2kHz or so they seem admirably restrained, while the bass and very low mid-range are more prominent than on most open-backed phones. In comparison, the closed-back S4Rs are more aggressive and coloured in the 1-2 kHz region, the bass being less prominent and also less deep. It's a tonality that is probably well tailored to their

intended role as tracking headphones.

We all want our monitoring systems to be 'neutral', but designing a pair of headphones that is subjectively neutral is a complicated business. If we take it to mean sounding like a flat-response loudspeaker in a good room, the S4s don't sound wholly neutral to me, and I wonder if a clue to that is to be found in the references to the ISO 226:2003 standard on OLLO's website. This standardises the 'equal loudness' curves originally developed by Fletcher and Munson, showing how the ear's frequency response varies with SPL. At all SPLs, the shape of the equal-loudness contour is broadly similar, with the ear being much more sensitive in the mid-range than at low frequencies. It could be that OLLO have tried to compensate for this contour; if so, I'm not sure how valuable this is for music production, since people who listen to your mixes won't be hearing them on compensated systems. I found the explanation on OLLO's website confusing, especially when the enlarged versions of their frequency-response graphs are visibly different from the thumbnails.

Having said all that, the soft mid-range and even high-frequency response means they're pleasant and non-fatiguing for long-term use, and neither do they deviate from the flat so much that you couldn't learn to compensate. Subjectively, I think there are more 'neutral' mixing headphones than the S4s, but their emphasis on the low end has obvious positives for some applications. In fact, learning to fill out the mid-range properly is a skill that many novice engineers struggle with. A monitoring tool that encourages people to push things a bit harder in this department might be no bad thing! *Sam Inglis*

\$ S4 & S4R \$320 each.

W www.olloaudio.com

Classic Keys: Keyboard Sounds That Launched Rock Music by Alan S. Lenhoff & David E. Robertson

Book Review

Electric and electronic keyboards played a massive role in shaping the development of rock and pop music throughout the 20th Century and beyond, from the early electro-mechanical organs and piano-like devices to more sophisticated and versatile electronic keyboards and synthesizers. Many are instantly recognisable and are rightly famed for introducing unique sound characters now associated with individual specific classic hit tracks and even for creating entirely new musical genres.

Alan Lenhoff and David Robertson are both devoted admirers (and experienced users) of classic keyboards and, together, they have created a marvellously impressive book (University of North Texas Press, ISBN: 9781574417760) detailing these instruments in great depth and breadth — and with gloriously detailed

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» photographs throughout. It's a very large (240 x 320 x 25 mm), heavy book, with over 400 pages of fascinating content divided into 17 chapters. Most chapters devote around 15 pages to a single instrument, describing the developmental history, construction and operating concepts, restoration issues and tips, associated musicians and recordings, and a wealth of other interesting and often little-known anecdotes and facts. The first three chapters really set the scene for the very earliest development of electric and electronic keyboard instruments. Thereafter, individual chapters are devoted to the Hammond B3, Wurlitzer piano, Hohner Pianet, Vox Continental, Farfisa Compact, Gibson G-101, Rhodes piano, Hohner Clavinet, Mellotron, RMI Electra-piano, Yamaha CP70B, and the Moog Minimooog.

There's also a chapter on the rise of the Japanese keyboard manufacturers like Roland and their forebear Acetone, as well as Yamaha, Korg, Kurzweil and others. Pretty much every significant keyboard and synth ever built and used professionally in the studio or on stage throughout the 20th Century, from the 1930s to the 1980s is at least mentioned somewhere in this enthralling and astonishingly comprehensive book. All that is, apart from the Yamaha CS80, a rather curious and disappointing omission (although its big brother, the GX-1, gets a brief mention). The final chapter concerns the collecting of vintage keyboard instruments, and some personal experiences and advice regarding their restoration.

I became completely engrossed in this book within seconds of opening the cover. It is very readable and generously illustrated throughout, celebrating the history, technology, innovation and application of so many classic instruments with an obvious love, respect and, at times, humour too. The depth of information and breadth of anecdotes and



fascinating insights into the mechanical and electrical workings of the instruments, some of which I previously knew very little about, is almost overwhelming.

As a lifelong fan of Hammond tonewheel organs, I like to think there isn't much I don't know about them, but I learned several really interesting new things from this book — and not just about Hammonds, either. I also found myself returning frequently, just to look at the pictures and discover more gems I'd perhaps overlooked or skipped past on my first read.

For anyone with more than a cursory interest in the classic keyboard instruments, the outstanding quality and breadth of content in this magnificent hardback book (an e-book version is also available) easily justifies its price and I really can't recommend it highly enough. It should be required reading for anyone considering a career in the audio industry, whether as a performer or an engineer/producer! *Hugh Robjohns*

\$ \$59.48.

W www.classickeysbook.com

Fishman TriplePlay Connect MIDI Guitar System For iPad

Fishman's wireless TriplePlay guitar-to-MIDI system has a strong reputation for fast and accurate tracking, and the company now offer these benefits in their TriplePlay Connect for iOS. This system combines a hex pickup and mounting kit with the TriplePlay Connect App (for iOS 9.3 or

later), and the latter is used both for setup and for sound generation. An included USB-to-Lightning cable connects the pickup to the iPad and also powers the pickup, which comes with a variety of spacers so it can be fixed to the guitar without modifying the instrument or leaving screw holes;

a simple slide-lock allows the pickup to be separated from the spacer when not in use. It is vital to install the pickup as close to the bridge as possible and to use the spacers and adjustments to get the pole pieces as close as possible to 1mm from the strings. (The pickup fits most guitars, including steel-strung acoustic models, though not Telecasters with traditional 'ash tray' bridge plates.) You can then download the app, which requires around 2.5GB of free memory, and start playing.

The app introduces itself with a guitar tuner and a page for balancing the string sensitivities. You're then taken to the main page, where you can select from a library of instrument types and sounds and explore the factory presets. While the sounds that come bundled with the app are fairly conventional with very limited editing available, they're of generally good quality and are arranged by instrument type, including drums, bass, keyboards, wind and so on. There are even guitars, both electric and acoustic. There are also numerous ready-made musical loops, plus the ability to record and store your own — the idea is that you can trigger these from specific notes on the guitar and then play along with them. The preset examples use the low strings above the 12th fret for starting and stopping loops, so you're unlikely to stray there by accident. Further clips can be bought as an 'in-app' purchase, but so far you can't add new sounds this way (something for the future?).

Many of the factory patches include a split to put bass sounds on the bottom two strings, but you can set up your own splits and layers by string and fret position, all shown on the fingerboard display running across the bottom of the window. You can also set up arpeggiators or program chords to be played from single notes. There's a lot to explore, including a range of onboard effects, but a series of video guides makes the process fairly painless.

Using a USB cable, also included, you can plug the pickup directly into a computer running a DAW and the pickup will show up as another MIDI input. There's no adjustment when working this way and with the current version, all the data runs on a single MIDI channel, which limits pitch-bends to single-note lines, but I found that it worked very reliably when teamed with Logic Pro X and is ideal for laying down pads or basic melodies. Having said that, Jam Origin's MIDI Guitar software also does a good job in this context, is cheaper and doesn't require a hex pickup. I mentioned this to Fishman and they revealed that



a multi-channel update would be available very soon, which should make the USB controller option far more appealing, as it would allow for string-independent pitch-bending, note splits and so on, all with the speed and accuracy for which Fishman are renowned.

As with earlier TriplePlay systems, the tracking latency really is minimal, making everything feel very immediate and if you play cleanly you shouldn't find yourself triggering rogue notes. The app is a lot of fun and puts in a very solid performance — and if you prefer to work only in your usual DAW, then the standard TriplePlay system may be more flexible, and costs only around 50-percent more. *Paul White*

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An Introduction To Writing Music For Television by Michael Kruk Book Review

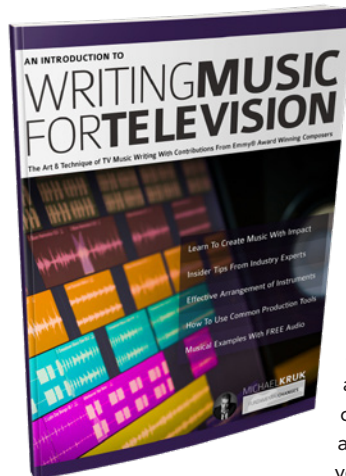
I recently reviewed Tristan Noon's excellent 'From DAW To Score' (see *SOS* October 2019) which provides budding composers with an excellent introduction to the process by which their freshly composed music can be turned into a score ready for the recording session. However, before you can attempt that, the music itself has to be written... If you're looking for some initial insights into the creative stage of writing music to picture, then Michael Kruk's new book (published by Fundamental Changes, ISBN: 9781789330557) might be just what you need. Michael is writing from a position of considerable experience as he has an impressive set of credits based in the documentary genre and, throughout the book, calls on the expertise of fellow composers Michael Price (*Sherlock*, *The Unforgotten*), Mac Quayle (*Mr Robot*, *Feud*) and Walter Murphy (*Family Guy*, *American Dad*) to chip in from their genre-specific backgrounds. If you're a guitar player, you may already be aware of the book's publishers, Fundamental Changes. Led by Joseph Alexander, they have an excellent reputation for music tuition books, so the team behind the project is a very good one.

Indeed, two commendable traits seen elsewhere in Fundamental Changes' extensive catalogue are evident here. First, this is a concise book (it runs to 112 pages in total), but it's right on point, and a great introduction to the topic. There is no waffle, just the good stuff. Second, to accompany examples of the practical compositional concepts introduced in the book (many illustrated by musical notation in the text), a collection of audio files can be downloaded from the Fundamental Changes website. Whether you read notation or not, the audio examples make the written concepts very 'real' and they're a valuable part of the

overall package.

This is very much a book focussed on how the compositional process differs for TV and film compared with that for other musical contexts, and the nine chapters cover a lot of ground. This coverage starts with some general principles of the composer's role within the overall project and, in chapter one, the importance of creating a sonic palette at the start of a project, so that your musical storytelling has a consistent 'voice' and you can work efficiently under the (often) compressed time-scales associated with composing for TV. There's solid, sensible, advice here. As elsewhere in the book, the examples used to illustrate different 'mood palettes' (action, tension, scary, etc.) are described in a clear, concise and helpful fashion.

Chapters two and three cover the topics of creating musical movement within a cue (or across related cues) and how chords/harmony can be used to create certain moods. In both topics, the different mechanisms available to a composer are discussed concisely and neatly illustrated, while the pros and cons of 'temp' music for the composer are also discussed. If you're approaching TV composition from a performance background (rather than music theory), chapter four is one of the best (and most easily understood) descriptions I've ever read on musical modes and their uses. It's light on theory but a great practical introduction. Chapters five and six focus on the use of melody, motifs, and how to handle intros, outros, builds and



reveals. Again, this is very much an introduction to all these topics, but the key concepts are clearly covered and the examples, in both notation and audio format, are always helpful.

The final three chapters cover the need for developing 'production' and 'orchestration' toolkits, as well as the pros/cons of developing both short-form and long-form versions of your cues to give the music

editor choice and flexibility. Given that so much TV music is produced entirely in the composer's computer, production skills can be as important as compositional skills. Equally, orchestral music, in pure or hybrid form, is a core part of TV soundtracks, and this section of the book outlines the key issues involved in making your sample-based orchestral sounds 'work'.

Of course, reading these 112 pages won't transform you into a fully fledged TV composer overnight. The content is more likely to appeal to the less experienced media composer, and you should definitely see it as a starting point for further study. But if this is a world you wish to inhabit, the book really does provide an excellent introduction to the creative (and some technical) parts of the process. Throughout the book, quotes from the experienced Michael Price, Mac Quayle and Walter Murphy are used well to clarify or add emphasis to the ideas being covered. If this is the sort of advice you need, the modest price would be money very well spent. *John Walden*

\$ Kindle Edition \$8.93. Paperback \$19.99.
W www.fundamental-changes.com

DDMF Metaplugin 3 Plug-in Chainer for Mac & Windows

I was surprised to find out just how many years ago it was that I first checked out DDMF's Metaplugin — my review appeared way back in *SOS* May 2011. Since then, this most handy of problem-solving plug-ins has matured considerably. Not only is the interface more user-friendly, but this VST/AU/AAX/RTAS plug-in (for Mac/Windows) is now capable of hosting VST2, VST3 and AU plug-ins, bit-bridging, and up to 4x oversampling for the whole plug-in chain being hosted. It can also deliver DAW automation data to any parameter of any of the hosted plug-ins, and supports DAW plug-in delay compensation. And what's more, Metaplugin itself isn't all you get, because bundled with it are three other utility plug-ins that can be used either in your DAW's insert slots or within Metaplugin: the M-S encoder/decoder and multiband splitter came with the original version, but there's also now

an audio routing plug-in called SendIt — and for users of some DAWs that last plug-in alone could prove worth the price of admission.

Metaplugin itself is essentially a plug-in wrapper that provides a blank canvas in which you can organise and route your own third-party plug-ins freely. By default, the GUI displays the audio and MIDI inputs to Metaplugin and outputs from it. It's your job to add plug-ins and link them together. You can load plug-ins directly, drag them from Finder (and presumably likewise from Explorer on Windows... I tested Metaplugin on a 2018 MacBook Pro running Mojave), or automatically scan chosen folders to populate a plug-in browser in Metaplugin itself. You can also right-click to insert a plug-in, duplicate plug-ins, and so forth. Plug-ins can be placed anywhere on the main section of the Metaplugin GUI, and the hosted

plug-in's inputs and outputs are made visible graphically. Creating a signal chain is a simple matter of dragging and dropping from the Metaplugin inputs to the inputs of your plug-ins, dragging from one plug-in's outputs to the inputs of the next plug-in in your chain, and from the last plug-in to Metaplugin's outputs. The links between plug-ins are indicated by graceful, curved lines — and when you hover over one of those lines, you'll find a control to boost or attenuate the signal level.

It's all very intuitive, and creating complex effects chains (eg. with feedback loops, parallel processing and more besides) is a doddle, and you can save these as patches in your DAW. Cubase/Nuendo users in particular might be interested to note that Metaplugin will reveal the external side-chain inputs of any VST2.4 plug-ins that have them. And coupled with SendIt (of which more below) that can grant you access to side-chaining with VST2.4 compressors and gates — Cubase normally requires VST3 plug-ins for side-chaining.

Slightly more complex is the parameter automation, and that's because the host DAW obviously can't directly 'see' the plug-ins hosted in Metaplugin. DDMF get around this by providing an assignable list of parameters — you map whichever ones you want to whichever parameter you wish to automate. This is obviously slightly more fiddly than accessing parameters directly in the DAW, but it's a sensible approach, and if you only want to automate, say, three parameters in a chain you won't have to wade through them all in your host DAW.

So what of SendIt? Well, this is reminiscent of the old freeware 32-bit Senderella plug-in that I used to use as a problem solver. You require two instances, one to act as a sender and one as a receiver. The sender instance can be placed anywhere where there's an insert slot — within Metaplugin, directly in your DAW, or even in another piece of software entirely. To return to that Cubase example, you could place the sender in your kick track's post-fader insert slot, and have the receiver feed the external side-chain input of a VST2.4 compressor inside Metaplugin to duck your bass part. When you stop to think about it, it could be useful in any DAW, though, as I can't think of a DAW that allows you to send from any point in the signal chain (such as from between two insert processors). You could, for example, use a send instance as a meter tap, moving the sender instance to get a reading on your meter from any point in your DAW.

I did make one suggestion to the developer that was warmly received, so I have high hopes that it will be implemented in the not-too-distant future: SendIt currently features a Thru tickbox that allows audio to pass through the plug-in as well as being sent elsewhere. My idea was to add a blend control, to balance the thru and sent signals — it would make it possible to create a 'BBC echo mixture'-style send (see here for details: <http://sosm.ag/send-and-blend>) very easily in any DAW.

Some of the other uses of Metaplugin are obvious. You can use it as a wrapper, for example to host VST plug-ins in Logic,



Here, Metaplugin has been used with its bundled multiband splitter to create a four-band saturator, with separate instances of Klanghelm's freeware IVG12 plug-in. When using the splitter, the signal can be recombined by dragging multiple outputs to a single input.

or VST/AU in Pro Tools. It can also act as a bit-bridger, breathing life into 32-bit plug-ins (if your OS allows them). But really, the most fun is to be had when you start exploring what can be done with the ridiculously flexible routing inside Metaplugin; your imagination really is the only limit. *Matt Houghton*

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Recording Status Quo:
 Guest engineer Andy Brook discusses his approach to recording the legendary British rockers' 33rd studio album.



ANDY BROOK

There can't be many acts who've released as many albums as Status Quo. I recently had the privilege of working on their 33rd studio album, *Backbone*. Shortly before embarking on the project, I'd taken over the reins in the Quo camp from Grammy award-winning engineer Gregg Jackman, and as I'd already worked on the Francis Rossi & Hannah Rickard release 'We Talk Too Much' I had some experience of working with Rossi. But *Backbone* was to be forged in very different circumstances to previous Quo albums.

This would be the first record of original material to be released since

long-time guitarist Rick Parfitt passed away in late 2016, and the first album to feature the newest band members, Leon Cave and Richie Malone. It was also the first Quo album that Rossi decided he'd produce solely by himself since 1991's *Rock 'til You Drop*, and from the outset he was determined to make it very differently from the Quo albums of the '90s and '00s. He wanted to make it on his own terms, rather than deliver something the band felt expected or obliged to — so when I say it was a privilege to be involved in this project, it's not just about the excitement of working with a big name; it was also a genuinely original, interesting and engaging project.

In this article, I'll take you through how

we approached the *Backbone* sessions, explaining some of the decisions we made and techniques that we used along the way, and hopefully I'll be able to cover a few things that might be of benefit to your own productions.

Building On The Demos

We recorded, mixed and mastered the entire project at Rossi's home studio. It's obviously a little grander than what most SOS readers would think of as a 'home studio' — it comprises a live room, a green room, a vocal booth and a comfortable control room, with a Harrison 48:32 console at its heart — but it offered us a similar balance of pros and cons. The band would, of course, be able to record the final album



Francis Rossi and Andrew Bown writing for *Backbone*, which you can hear at <https://statusquo.lnk.to/Backbone>.

in the studio, but they'd also have the luxury of time to work on developing the songwriting and making demos there too.

Over the years the studio has made the move from recording on ADAT machines to a Mackie HDR 24-track hard-disk recorder and then, eventually (on Jackman's arrival), to a computer-based setup based on Logic Pro X. I'm actually a dyed-in-the-wool Pro Tools user, but it was important that I use the in-house system for this production —

thankfully though, I've used Logic many times over the years, which meant the switch to Logic wasn't too daunting.

The band started the demo sessions around February 2018 with a view to recording the album in the Autumn. We kept everything that Rossi and long-term collaborators Andrew Bown, John 'Rhino' Edwards and Bob Young sketched out during songwriting sessions, and we also imported drafts from band members who'd recorded things remotely, on various systems: Boss hardware recorders, Logic sessions, and even some voice memos.

This material formed the core of the album and was kept for reference in the working project right up until the final mix — in fact, a few of the 'demo' parts we'd recorded even made it onto the finished record. The idea of retaining the demo material throughout was that we could make sure we never lost sight of the energy in those original ideas — and I think it's fair to say this approach worked well.

One upshot of working with the demo material in these sessions in this way was that the drums were not the first thing we laid down, as they typically are in rock productions. Instead, we built the album

recordings around the guides we'd created during the writing sessions, starting each song off by recording guide guitars from each writing partnership, and a guide vocal from Rossi.

Whether working on demos or the final product, I'm a firm believer in using studio time with the artist for recording; the engineer/producer should always be focused on capturing the energy of the performances, and things like editing, processing and comping can be done later at your own leisure. Rossi has a similar philosophy. Thankfully, it was also apparent from an early stage that *Backbone* was not going to be a project that would leave us with take upon take of material to sift through later— recording really was a case of getting down just one or two takes (three at most) that could be tidied up to appear as the final 'take' on the record.

Vocals

Although I've just called them 'guide' vocals (since that was the intention), we »

The Harrison console in Francis Rossi's home studio, with some of the other gear mentioned in this article in the rack to the right, including dbx 160 and Drawmer 1960 compressors.



» rarely needed to redo any of these initial lead vocal parts, other than to drop in a word or two to accommodate the occasional lyrical tweak. At some stage, the lead vocal would be double tracked, to achieve the definitive Status Quo vocal sound — sometimes immediately after the initial take, sometimes after everything else had been laid down. Incidentally, it's fascinating to see just how exacting an artist is about their double tracking when they've spent a lifetime doing it! I often reach for Synchro Arts' Vocalign or Revoice Pro when dealing with multiple vocal tracks, but great as those tools are, for this album I happily left them unused on the lead parts.

We used an AKG C414 and a Neumann U87 set up in tandem, each running through a dbx 160 VCA compressor. On many 'classic' Quo records the vocals were tracked and double-tracked with a C414, a setup that persisted until Jackman encouraged Rossi to adopt the U87 in the '90s. Eager to see if the mic choice could help recreate a vintage sound, we kept both channels running for most of the early vocal takes, allowing us to compare the results later. As it turned out, the C414 wasn't as impressive as we'd anticipated — it sounded somewhat brighter, but it didn't really lend itself to this vocal. The U87 wasn't entirely what we wanted straight off the bat either — it offered something the C414 didn't, yet still seemed sort of 'muted'. To arrive at the desired sound, we EQ'ed the U87 to emulate the frequency response of the C414, which yielded a nice, full-bodied tone, with neutral mids and airy highs.

A theme of huge 'block vocals' developed very early on into the recording process, and this became part of the signature sound for the album. It was a case of quadruple-tracking a background part and then adding a fifth and an octave above that (again, quad tracked). These had to be super tight — unnatural even — to achieve the desired effect, so I made liberal use of Revoice Pro for this. (Prime examples of this sound can be heard on the tracks 'See You're In Some Trouble', 'Backing Off' and 'Falling Off The World'.)

Electric Guitars

The move away from the wall-of-guitars sound of some recent Quo records was deliberate. The idea was to get the sound to 'breathe' more, and so our approach involved capturing sounds with



To capture Rossi's guitar sound, a Shure SM57 facing the centre of the cone of a Marshall JCM900 combo made the biggest contribution, but this was mixed with a more distant Neumann KM183 omni room mic set up at head height.

considerably less harmonic distortion, and again choosing not to track take upon take of the same part.

Many of the songs were constructed around riffs, but there's a signature sound to Quo which involves a guitar figure movement from playing a fifth and then a sixth and then back again — otherwise known as 'the finger'. It forms the rhythmic basis of the songs, so we chose to get these down at an early stage — other figures and parts could then be developed around them.

Rossi is an exceptional blues guitar player and rather than recording rehearsed ideas for the solos on this album, he preferred to jam them out to see what felt right. As with the vocals, it was a case of two to three takes before discussing whether he'd nailed the part and if it served the song. As a guitar player myself, I had to make a concerted effort to think as an engineer/producer, more in terms of the record and how the phrasing, articulation and tone of the solos sat in the song than I usually would as a player. Sometimes we'd fine-tune the parts (not as easy as you'd expect when it's all improvised like this!) or comp the solo out of the two or three takes we had on file.

We quickly got into a system of a three-channel recording setup for Rossi's parts, with a DI signal feeding a tailored AC30 emulation in Logic, and a dual-miked 60W Marshall JCM900

combo set up in the live room. A Shure SM57 was placed dead centre on one of the cones, and a Neumann KM183 omni small-diaphragm capacitor model placed six feet from the amp at head height, to give us a more neutral reading of the room — the omni pattern doesn't seem to exaggerate room modes as much cardioid mics tend to. The height was simply set to reflect what the listener would hear if stood in the room.

For the most part Rossi used his custom Status guitar. This is a solid alder body model, fitted with Hot Rail pickups and a bolt-on graphite neck, and featuring a variety of tonal controls. I confess that I had my doubts to begin with — but I'd soon find myself pleasantly surprised by this instrument's versatility. Depending on the material, I often like to use some form of physical noise control, such as a Gruv FretWrap (string damper) when recording guitars, and something like this is essential when dealing with carbon-fibre models, which always seem to exaggerate sympathetic resonances.

Richie Malone's demos had been recorded at home on his Boss BR-800, using some of that device's COSM modelling options. In comparison with the less saturated tracks Rossi had already laid down, the tone was heavily driven, and the juxtaposition worked really well. Again, in keeping with the demo-to-delivery approach, we embraced the Boss unit as part of Malone's sound

when he flew over to record his parts. We ended up running that in tandem with a Marshall JCM2000 TSL that I'd brought in, and another AC30 model in Logic.

The only part of the album that was recorded with more than one musician at the same time was when Andrew Bown came in to lay guitar parts for the tracks 'Running Out Of Time' and 'Backing Off'. He'd co-written these songs with Rossi, and we managed to capture a real vibe when both of them played together. This is not to say that recording this way will necessarily be better, but it's always worth experimenting. In this case it made a refreshing change to do things a little differently and, unarguably, you can hear the chemistry in the result.

Drums

I'd previously done some recording with drummer Leon Cave on other projects — we'd even played in the same band at one point — and I knew him to be a no-fuss player who'd nail each song in a couple of takes. So it was imperative that we capture everything successfully in the short time he'd be in the studio, and not

rely on pick-ups at a later date.

Leon's kick is fitted with a suspended Shure Beta 52 to allow for quick setups, and it sounds great but it's generally a good plan to aim for the right sound before reaching for EQ, so I wanted to audition some other options. I'm a big fan of the Audix D6, but we wanted something less 'snappy' here; warmer,

but still with some 3-5 kHz presence. An AKG D12 was certainly warmer but lacked top-end. We ended up preferring an AKG D112, placed just inside the port of the kick drum. I ran this through a Drawmer 1960 tube preamp/compressor, with a slow attack, release timed according to each song, and a ratio of around 3:1 — it wasn't what you'd call subtle, but it gives a nice sustained fat kick sound. I also added a Solomon LoFRoQ mic

(a variant on the sub-kick theme) just outside the shell to augment this with some smooth low end, and after a couple of placement alterations to improve the phase relationship with the D112 we had our hard-hitting kick sound.

The snare, a Noonan Deluxe 10-lug 14 x 5.5-inch brass drum, was miked with an SM57s on each head (one polarity

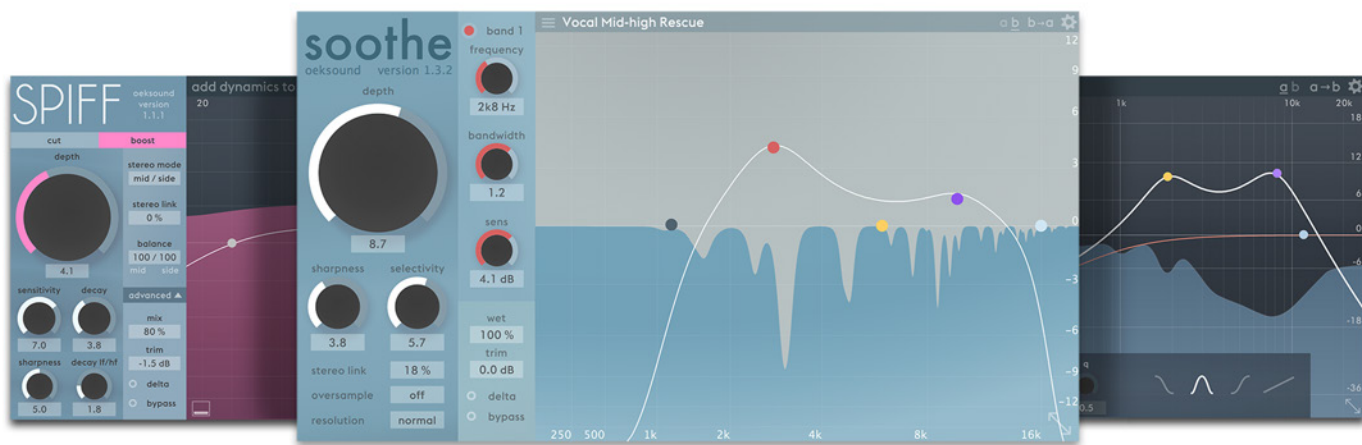
"The more parallel the mic is to the snare head, the more attack it will deliver; and the more perpendicular, the more 'body' it will capture."

inverted, of course), paying particular attention to the angle of the one covering the batter head. The more parallel the mic is to the snare head, the more attack it will deliver; and the more perpendicular, the more 'body' it will capture. We wanted snap, so we went with a low profile, and left enough distance to avoid the worst excesses of proximity-effect bass boost.

I approached the toms similarly, using Beyer M201s on the racks and Sennheiser >>

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» MD 421s on the two floor toms. (The M201s' tight hypercardioid pattern makes them great for isolation, but MD 421s deliver a beefier sound, despite their flatter low-end response.) These were all run through dbx 160 compressors, and some subtle EQ was applied via the Harrison console — this is a parametric EQ, making it easy to isolate and boost the fundamentals for each tom while scooping the mids and adding a touch more attack.

Rather than opt for an X-Y or A-B overhead configuration, I spot-miked the crashes, china and ride cymbals individually, with a few Neumann KM184s. This gives ultimate control over their balance in the mix and can deliver a brilliant tonal quality that exaggerates certain nuances often lost with more distant recording techniques. You must make sure these spot mics are not too close to the cymbal, or shear waves can cause the diaphragm of the mic to distort. A good rule of thumb is to have the mic at least the diameter of the cymbal away from the surface, and to offset any transverse wave distortion by angling the mic between 45 to 67.5 degrees. Perhaps counterintuitively, this spot-mic approach can help avoid phase issues — although there are more mics, there's no one overhead pair that all the others need to be phase-aligned to.

For the hats, I used a U87 set to figure-8, pointing horizontally towards the floor tom across the hi-hat, and with the side null aiming at the surrounding cymbals. This was extremely effective in rejecting other elements and created a tight isolated hat signal.

Without a typical overhead setup, the room mic could play an important role. I used another U87, this time set to omni and placed at the farthest point in the room from the kit. Being so close to a physical boundary gives it some semblance of a PZM, but with the addition of the low-end boost you'd expect from a U87. Again, with such a distant, ambient mic, phase relationships weren't such a critical issue.

Bass

John Edwards played through a Markbass LMK head, each channel of which offers an arsenal of parameters (you can blend between these channels too). The head powered a Markbass 4x10 cab, which I miked with a D112 and a Solomon LoFRQ. I took two DIs (one from the



There was a conscious decision to spot-mic the cymbals and glue the drum kit sound together with a room mic rather than traditional overheads. And despite some pricier and vintage options, an inexpensive AKG D112 was preferred for the main kick mic.

instrument, and one via the head) for possible re-amping. While there were no phase issues between the mics, there was work to do to align those signals with the DIs. (It's not uncommon to find phase-alignment issues between electrical and acoustic signals captured from the same source, so it's always worth checking — the solution usually lies in time-alignment or using an all-pass filter, sometimes called a 'phase rotator')

Acoustic Guitar

Acoustic guitar only appears on the introduction to 'I See You're In Some Trouble'. We recorded 16 separate takes of this intro, with a U87 set to omni, a KM183 room mic and a DI fed from the guitar's piezo pickup. As the texture swelled, it became apparent that the



The SM57 snare mic's body was low, and almost parallel with the snare's head, helping to capture plenty of each hit's attack.

DI tracks were inconsequential and the room mics needed to be used minimally. Once I'd tidied the tracks up, panned and levelled them to taste, they were bounced as a stem and all the original tracks discarded from the Logic session. I'm all for keeping alternate takes and versions 'just in case', but when you have already have the sound you're after it makes no sense to clog up the GUI, and sometimes it's good just to commit to a decision.

Keys

Rossi had recently picked up a Yamaha Clavinova CSP-150 on tour, on account of its impressive on-board sound library, and this sat in the corner of the studio with a pair of Roland DS-90A monitors, creating a nice 'keys' station that gave Andrew Bown a degree of privacy to



For the hi-hat, a U87 was set to fig-8 mode, and the side nulls were aimed at other cymbals for maximum rejection.

work in — Rossi felt this did something for Bown's comfort, compared with sitting him in the middle of the room. He was right: Bown really shone when working in his corner!

Most of the keys parts were recorded as audio, with only a few using MIDI to access some sounds from Logic's EXS24 sampler or EastWest's Composer Cloud after the event; with a player of Bown's calibre, there's no need to keep the MIDI purely to tweak the performance. There's a great deal of tonewheel organ on the record, and to begin with we fed a Roland RD-2000 stage piano patch through a Leslie cabinet, with a three-mic setup. But while Bown has always been something of a purist when it comes to recording Hammond, he was open to broadening his approach, so we tried triggering Logic's Vintage B3 virtual instrument using the RD-2000. With a couple of MIDI-learned control changes and a drum-speed switch operated by a volume pedal, we were away! The sonic difference was small but, importantly, there were now no squeaks, mechanical

noises or voltage drops to worry about.

We also made good use of Roland's pocket-sized VP-03 vocoder in the second verse of 'Liberty Lane', although it proved tricky to capture the sound we wanted as we found the VP-03's input seemed prone to overloading with loud vocals. After experimentation with different mics, we found this device worked best when singing in the register of the keyboard part... and even then it needs be more of a whisper than a firm vocal to really get the effect to shine.

Mixing

This article was intended to be more about recording than mixing, so I'll only pick out a few mixing highlights — but I'll emphasise that this wasn't a linear project, where we'd finish tracking everything and then start mixing. In fact, there wasn't any separation between these stages, and that was deliberate. If there was an idea for the mix it would be executed there and then. We wouldn't have to revisit ideas later, and that helped to keep everything fresh.

An interesting aspect of these mixes was that I'd set up a side-chaining template at a very early stage, aiming to address a lot of mix issues almost automatically. The bass was ducked by the kick and the floor toms to let those events punch through; the hats by the snare, to keep the snare attack nice and clear; the rhythm guitars by the lead guitar figures; and the entire track by the main vocal. These interactions were subtle, with only 1-2 dB of gain reduction at most, but it really helped to glue things together without having to engage automation at every turn. That said, we eventually abandoned the idea of using the vocal to duck the mix; when it comes to where a vocal sits, there's a clear distinction between pop and rock tracks, and the side-chaining gave the vocal too much prominence.

The bottom end of the snare was 'shaved' to remove some honk/boxiness. People often opt to push a snare in the 150-250 Hz area to fatten it, but I find this can be counterproductive in the context of the mix. Wanting to push the snare

»





The Markbass cabinet was miked up with an AKG D112 and a Subkick-style Solomon mic, but two DI signals were also captured for maximum flexibility — one direct from the instrument, and one post the Markbass head's preamp stage.

» further sonically, we instead opted to use gated reverb to fashion the drum's decay. (Decay times were always song-specific, designed so they'd fall in line with the tempo of the track.)

We also took a lot of low end out of the vocals, particularly the BVs, to address what Rossi would describe as 'woofiness' or 'wooliness' — that distinctive build up of 'mud' when multiple layers of vocals are stacked, even when more conservative high-pass filters (around 100-120 Hz) are used. We pushed the filter for most vocals to around 300Hz, and even 500Hz in a couple of cases. It's amazing just how much of a sound we can still perceive when its fundamental frequency is missing!

Mastering

The album that the band, the record label, the management and the agent had heard in the studio prior to mastering was dynamic, earthy and organic. We needed to maintain this in the finished record, so the final master needed to be supremely sympathetic to the mix.

After having a few test masters back, a decision was made that the album would be mastered 'in-house' — in other words, by me! The main argument in favour of this approach was that everyone who'd heard it to date was very excited with how it sounded, so it didn't make sense to outsource the project and have that sonic content change. Now, I do enjoy the mastering process but I'm also of the mind that if you've been involved

in the recording and mixing, it's essential to get some distance from the project before returning to work on it at this stage. So I made a point of stepping back for a while, and spent some time listening to current releases to get a better idea of contemporary trends.

Thankfully, we seem to be reaching a time where we aren't necessarily expected to produce unnecessarily loud or over-compressed masters. Having approached the mixing understanding that someone else would master the record, I'd paid very close attention to the levels throughout, making sure there was ample headroom on the master bus, and avoiding clipping at plug-in, track, group or aux stages. That was harder than it sounds, due to the sense of excitement in the room; throwing track after track at a song eats headroom quickly, and it kills creativity if you keep stopping to rebalance the mix. You can't just bring all the faders (or fader groups) down when automation is already in force, though — in those cases, creating VCA groups allowed me to claw back headroom quickly, without disturbing the mix balance.

I mastered 'in the box' using iZotope Ozone 5 (not a current version — I prefer the interface and I use some tools that aren't available in later revisions!). There's no point applying processes for the sake of it — mastering is all about listening and making judgments. But for this project, I typically used M-S multiband compression to tame the bottom end and expand the high-mids and highs without making things too wide. I also applied subtle low-end boost for an added sense

of weight, and an air boost to brighten the high-end a touch. A tiny amount of filtered reverb (high-passed at 500Hz and low-passed at 2kHz) helped gel things together — the filtering prevented this tactic making the end result sound too boomy or bright.

The other issue one has to contemplate when mastering, of course, is loudness. I really like the interface of Waves' WLM loudness meter, and used that here — I don't aim at a particular LUFS figure, but it's handy to keep an eye on the meter while making decisions. Rossi and I agreed that we didn't want to push the levels too hard, and we found that if we tended to stay on the safer side of -10 LUFS it generally worked well: it didn't compromise the dynamics or sonics of the music, and the result was a polished, punchy, dynamic master that satisfied everyone involved and that any discerning listener could enjoy.

Here's what Francis Rossi had to say on the mastering: "We realised the way we mastered was different compared to many modern [*records*]... I don't see where the musicality is in making it louder. Surely, musicality comes with maintaining the

About The Author

Our special guest author for this month's Session Notes feature is Andy Brook (pictured), who works freelance as a producer, mix engineer, recording engineer and musician, and has produced, engineered and toured with the likes of Status Quo, Bonnie Tyler, Ginger Wildheart, Uriah Heep, Francis Rossi, Mel Gaynor, Greg Jackman, Del Amitri, Travis, Tiger Lillies, Wonk Unit, and Rich Ragany & the Digressions.

W www.andybrook.com



Photo: Tom Sheehan



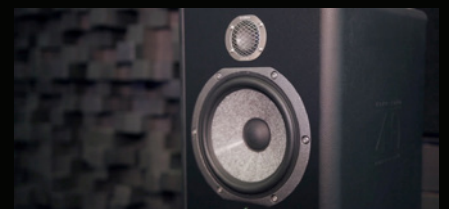
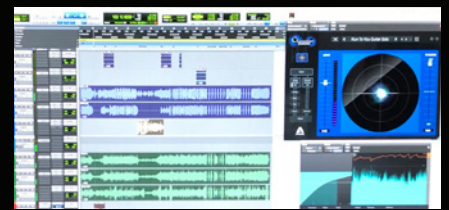
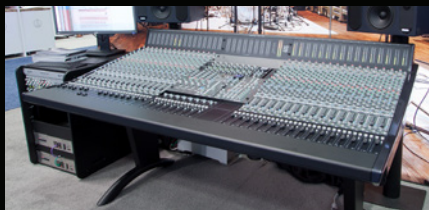
Having initially played an organ part through a triple-miked Leslie cabinet, for this record the band eventually settled on Logic Pro's Vintage B3 virtual instrument.

dynamics. There might've been listeners that found the album is not as loud as some current releases. However, we do have amplifiers and a volume control — turn the fucking thing up!

"At one point we'd got so far with the mastering, and I remember thinking it just sounds like every other Quo thing we've

done. We went back a notch or two from there, and it sounded like it does now. You have to let the music breathe a bit. It

breathes when it's live. It breathes when it's being put down... We ended up with a record that does just that." **///**



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MIXING VIRTUAL DRUMS

MIKE SENIOR

Software instruments that can emulate live drumming are 10 a penny these days, and many come with libraries of MIDI parts that make programming a convincing performance a breeze. Most such instruments also provide multi-channel outputs, a bit like the multiple mic channels of a real drum kit recording, but in practice you'll rarely get the best results at mixdown if you try to treat a virtual drummer just like a real one.

Start With The Source

The first thing to realise is that, with any software drummer, the mixing process should always start within the instrument interface itself — if any individual kit component doesn't sound quite the way you want it to, then reach for the software's own instrument controls before you start getting any other mix processing involved. Your snare, for example, will sound a lot more natural if you increase its sustain using the software's built-in envelope controls, rather than trying to compress the snout out of it with down-stream plug-ins. Similarly, adjusting the design or pitch of a virtual drum makes a tonal difference that's impossible to replicate with EQ — and if you tune the

drum to the key of the track it'll often blend better straight away, without the need for artificial reverb.

Beyond changing the sounds themselves, you'll usually get some flexibility to rebalance those sounds within each of the software instrument's audio output channels. In the simplest case, there'll be a selection of output channels for individual kit components (kick, snare, rack tom, floor tom, hi-hat, ride, crash and so forth), each providing a 'complete' dry mixed sound for that instrument, and those will be accompanied by some kind of 'ambience' or 'room' track you can use to give the mix a roomier sound if you

like. In this kind of situation, your highest priority should be to decide how loud each kit component is within the simulated room, because the louder a given kit component is in the room sound, the roomier it'll seem in the mix relative to the other components. However, it's not uncommon to find that the software instrument's simulated room sound is more appealing on some kit components than others within your specific mix context, in which case you can make your life a lot easier by muting certain kit components in the software instrument's room channels, and then adding effects to the individual component channels within your

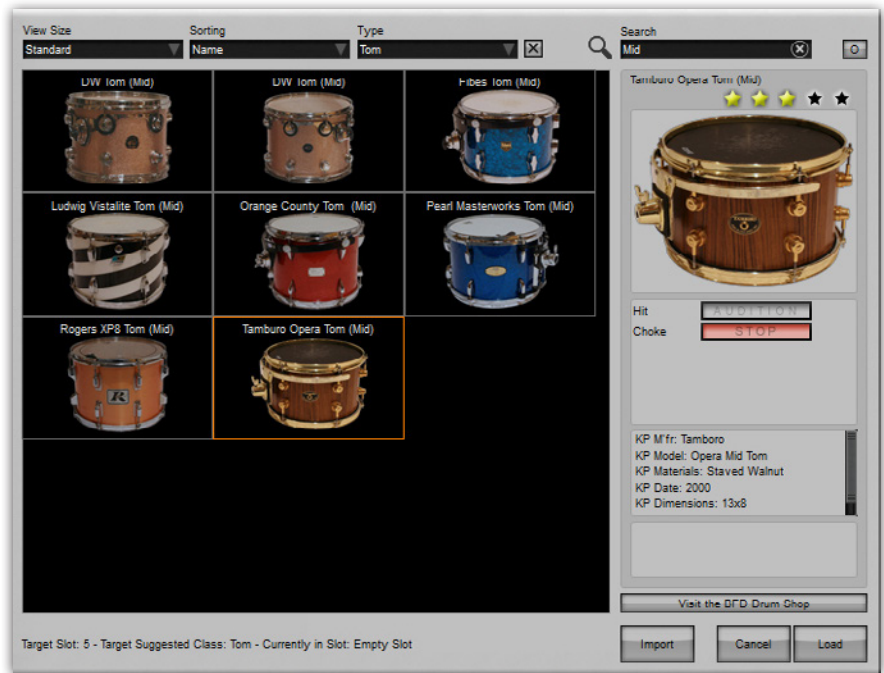
Simply swapping out a kit component within the instrument can solve mix problems that EQ is powerless to remedy.

DAW instead.

The more sophisticated virtual drummer instruments, however, don't output each kit component 'ready mixed', but rather emulate the signals you'd expect to get directly from a miked-up acoustic kit: in other words, multiple virtual close-mic signals per kit piece, together with overhead mics and room mics that pick up the whole kit. While this does afford you much more scope for sonic adjustment, for many readers that often translates in practice as 'enough rope to hang yourself with' — especially for those lacking enough experience mixing real acoustic kit recordings.

Mixing Inside The Instrument

My first shortcut to decent results in this case is to ignore all the close mics at the outset, and just listen to the virtual overheads. Ideally, these should sound as close as possible to the kit sound you have in mind. I can't tell you how



often I fade up the overheads of a virtual drum instrument and it sounds nothing like you'd expect a natural kit to sound

— perhaps the snare's very low in level, or the hi-hat's overwhelmingly loud, or one of the toms is entirely missing. Unless >>

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Many modern music styles feature an unnaturally dry kick, so if you're pursuing that kind of sound then you may want to turn off the kick drum's feed to the software instrument's internal overhead, room or ambience channels.

» your overheads sound fairly natural, you'll be facing an uphill struggle trying to get the full drum kit mix to sound any good. If the snare is too low in level, it'll lack width and size; if the hi-hat's too loud, you'll probably balance the overheads too low, so everything else won't really glue together; and if you're only hearing the tom through its close mic, it'll always sound a bit 'stuck on' rather than blending with the kit as a whole. There is one common exception to this rule of thumb, though: it's very common for kick drums to be presented unnaturally dry in modern rock mixes, so you may want to remove that drum entirely from the overheads mix. (That said, legendary rock mixer Andy Wallace once told me that he actually likes the sound of kick-drum ambience, so this

is something you might want to consider while listening to some of your own favourite productions.)

My second piece of advice is to switch off any simulated 'spill' on the virtual close mics. You know, where you can decide to hear a bit of the cymbals and toms on the snare close mic, for instance. The reasoning behind this facility seems sensible on the face of it: real kits have spill on their close mics, so why not emulate that? But almost invariably I find that this simulated spill simply doesn't respond in the same way that real acoustic spill does. You see, with a real drum recording you can make spill work in your favour, manipulating the phase relationships between the different mic signals so that the spill supplements and enhances the sound of each kit component. But whenever I try to do the same thing with the virtual spill signals in software instruments, the result always ends up sounding a bit hollow and phasey, and generally less satisfying than simply switching off all that spill and working with the artificially spill-free close-mic signals instead!

While you're in the software instrument's internal mixer, I'd also consider switching off any compression effects that are built into the instrument patch, because they can be a bit overblown — presumably to impress prospective customers who are likely surfing presets

in isolation! The problem is that many styles that rely on live drums also benefit from assertive master-bus compression for cohesion and a sense of excitement, but if you've already squashed the drums flat within your software instrument, you won't have enough drum peaks left for the master-bus compressor to react to. I've also discovered a tendency for people to over-compress software instrument close mics in their DAW's mixer, forgetting that it's both easier and more effective to adjust the dynamics of the performance without any compression side-effects, simply by editing the Velocity values in the MIDI trigger data.

Another internal feature I'm normally inclined to switch off in software drum instruments is any built-in reverb processing. Again, it's not that reverb isn't frequently useful for mixing drum kits; it's just that you'll get better cohesion in your mix if you use send effects within your DAW which can be applied to other instruments in addition to the drum kit. That way, lots of instruments will share some acoustic commonality with the drums, and you'll get more of a sense that everything was performed at once in the same room.

Mixing Outside The Instrument

If you've done your work with the software instrument properly, mixing the virtual drum kit channels really shouldn't be too difficult. Indeed, if it is difficult, that's a good indication that you need to tweak the instrument's internal parameters further! Because spill isn't really a concern, you obviate the need for most of the gating and filtering you might require for a real acoustic recording. Similarly, you can avoid dynamics troubleshooting by refining your MIDI trigger data, and there should be much less need for in-depth reverb treatments once you've balanced your overhead and room signals sensibly — just a touch of global reverb to bind the kit together with the rest of the band, perhaps. As such, it should probably go without saying that mix templates designed for live drums aren't likely to be much use for virtual drum instruments.

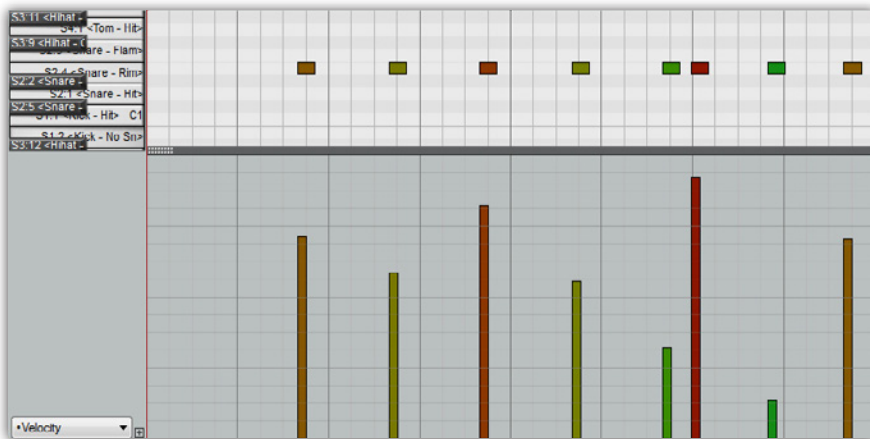
One thing that is always worth confirming, though, is that the phase-relationships between the different close-mic signals and the overhead/room mics are optimised. You'd hope that library developers would check for this kind of thing, but I have come

Sub Optimal

Back in the '80s, people often wired up a Yamaha NS10 woofer to an XLR and used it to pick up the low-end of the kick drum, and Yamaha later used a similar principle to develop a product called the SubKick. Because the speaker cone is so heavy, it hardly responds at all to high frequencies, while the low frequencies tend to ring on longer than you'd expect on account of the resonance characteristics of the driver itself. And, of course, if you mix the resultant low-end booming with your other kick-drum mics, it can make the low end sound huge. I seldom use this kind of signal much in the mix. This is partly because it leaves no headroom in the lower octaves of the spectrum for a full-sounding bass (something that's usually more important for underpinning the harmonies in mainstream song-led productions). But in addition, if you add that kind of low end to



up-tempo tracks, you'll just make the kick-drum patterns feel sluggish and lacking in punch. In my experience, more traditional kick-drum mics will almost always provide more useful and controlled low-end power.



There's no point in leaving loads of Velocity variability in your drum part's MIDI data if you're then going to smash it with heavy compression in your DAW mixer. You'll retain better transient definition if you make the MIDI data more consistent instead and then compress more moderately.

across situations where reversing the polarity of a mic or two has made a useful improvement to the combined sound. And don't be afraid to cut away some of the drum frequencies once you start trying to incorporate the kit sound into your mix. Remember that instrument presets are often designed to impress in isolation, which means they're unlikely to leave enough mix real-estate for the rest of your band.

For me, the bulk of the work when mixing virtual instruments usually involves trying to add two things: cohesion and interaction. The first of these is the sense that the ensemble belongs together and that the instruments don't sound like a bunch of disconnected overdubs. As I've already mentioned, a touch of global reverb can help here, and this doesn't need to have a long tail to work. I usually keep that kind of reverb quite short, in fact, so that it's not really detectable at all — until you switch it off and the mix doesn't blend any more! Subtle master-bus saturation or tape emulation can help here too (particularly if that adds a subliminal background noise layer to the whole production), but do keep a careful ear open for unwanted side-effects, such as any loss of low-end solidity on the kick, or any softening of snare drum transients.

Compression can also help improve the sense of cohesion if applied to the drum subgroup/bus, but if you apply it to your master bus, then it can introduce a sense of interaction too; when the kick-drum hits, the levels of the other instruments in the production are momentarily ducked, for instance. But the problem with master-bus compression is that all it sees is the signal level, not the musical

value of each part. So don't forget to take advantage of fader automation — because it uses a human gain control element (ie. you), it can introduce interaction between the drums and the other parts in a more musical way, fading up interesting hi-hat fills, for instance, or emphasising the kick drums and cymbal hits at the starts of important sections.

Soft Landing

I hope this article has given you some pointers to improve your mixing of software drums. Although they sometimes don't respond in exactly the same way as real drum recordings, they really can sound fabulous once you learn to manage their eccentricities. ■■■

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CRAIG ANDERTON

VCA (voltage-controlled amplifier) faders don't pass audio. Rather, they act as 'remote controls' for other faders: assign a track fader to a VCA fader, and now the VCA can control the track's gain, without moving the associated track fader. Assign multiple track faders to the VCA, and you can bring all their levels up and down simultaneously. You can also adjust track faders independently of a group controlled by a VCA fader, without causing the other faders to follow along.

Cakewalk doesn't have VCA faders, and it's understandable if your response is, "What's the big deal? Can't you just send the channels to a bus, and use the bus fader to control the level?" The answer is yes, and if that's all you need to do, groups are fine. Even with a channel in a group, you can still adjust a fader independently of the other tracks in it by holding down the Ctrl key while moving the fader. When you release the fader, the control returns to being part of its original group.

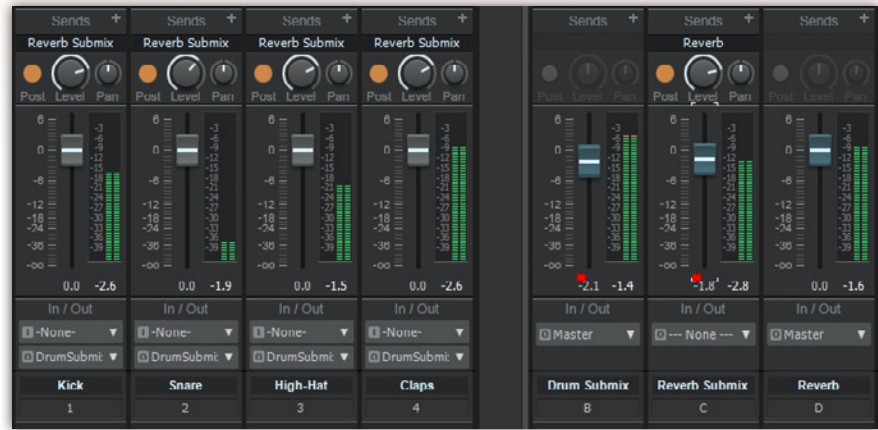
However, there are other VCA fader functions that Cakewalk can't do, like having the same fader belong to different groups. This can be helpful for projects with a high track count. For example, you could have fader groups for violins, violas, and cellos — nothing unusual there — but you could also have a violins + violas fader group, and yet another fader group with all the strings.

One of the classic scenarios where VCA faders win out is when several tracks (say, individual drums) go to a submix fader, and the tracks also have post-fader send controls going to an effect such as reverb. With a conventional submix bus, as you pull down the bus fader, the faders for the individual tracks haven't changed — so the post-fader sends from those tracks will still be sending signal to the reverb bus. As a result, it's not doing what a post-fader send is supposed to do, which is reduce the send amount as you pull down the channel fader. Even with the bus fader pulled all the way down, you'll still hear the reverb.

A VCA channel solves this because it controls the gain of the individual channels. Less gain means less signal going into the channel's send control, so with the VCA fader all the way down, there's also no signal going to the reverb. However, even though Cakewalk doesn't

High Voltage

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Screen 1: This setup pulls back on the drum track effects send levels when you reduce the drum submix level.

have VCA faders, you can use traditional bus-based routing, or grouping, to accomplish the same result.

Taking The Bus

Let's reprise our drum submix scenario, where you want to route drums to a reverb bus, which also provides reverb for some of the project's other tracks as well, and that you want the reverb level on the drums to drop when you reduce the drum submix level. To do this:

1. In addition to your drum bus and reverb bus, create a reverb submix bus.
2. Assign the reverb sends from the drum channels to the reverb submix bus.
3. Add a send to the reverb submix bus that goes to the reverb bus.
4. Group the drum reverb submix controls together.

Now when you bring down the drum submix level, you'll also bring down the level going to the reverb, which accomplishes the same as using a VCA channel to do this. As expected, this won't affect any other non-drum channels going to the reverb.

On the other hand, you may not always want the input level to a processor to decrease when you pull down a bus. For example, with saturation or compression, altering the input will change how the processor responds. To preserve the input level to the effect, you don't need an additional submix bus. Just use pre-fader

sends to the bus containing the effect, then simply group that bus' fader with the submix bus' fader. Bringing down the level of one brings down the other, but the amount of audio going to the effect remains constant.

Group Activity

This is an even simpler approach, which uses grouping rather than dedicated busses. Assuming you have a drum bus, reverb bus, and reverb sends on all the drum tracks where you want reverb, all you need to do is the following.

1. Select all the drum tracks.
2. Right-click on one of the tracks' send controls, and assign it to a group.
3. Right-click on the drum submix output fader, and assign it to the same group.

Now when you pull down the drum submix, you'll be pulling down the reverb sends as well. Remember that you don't have to remove a send from its group to change its setting — just hold down the Ctrl key while you adjust it. (With the bus-based method described earlier, you needn't remove a send temporarily from a group, because it's not part of a group. Just adjust the send control.)

The Offset Rule

Another use for VCA channels is to control the gain of a channel without affecting existing automation. If the automation changes are exactly as desired, but

Screen 2: Offset mode makes it easy to change an automation envelope's overall level.



the overall level needs to increase or decrease, simply offset the gain so that the automation curve remains intact, and then adjust the overall level with the VCA fader. This is often simpler than trying to raise or lower an entire automation curve.

Cakewalk has always been able to do this, thanks to the Envelope/Offset Mode option. To enable this, click on the Envelope/Offset mode icon in the Control Bar's Mix module, as in Screen 2. (In addition to level faders, this mode also works with audio track automation for pan, bus send level, bus send pan, bus return level, bus return balance, main out volume, and main out balance, as well as MIDI track automation for volume, pan, chorus and reverb.)

In Offset mode, any parameter that can be offset has a '+' sign appended to the parameter value, eg. 'Pan 11% L+' instead of 'Pan 11% L', or, for a level fader, '-2.3+' instead of '-2.3'.

Say a fader reads -4dB, you switch to offset mode, and then move the fader to +2.1+ — you've offset the net value by +2.1dB. The fader's effective value is now -1.9dB. This is ideal when all your automation moves are just right, but the overall level needs to be trimmed up or down.

Sonar used to default to using O as a keyboard shortcut to select Offset mode. However, it was removed as a default because people often hit O

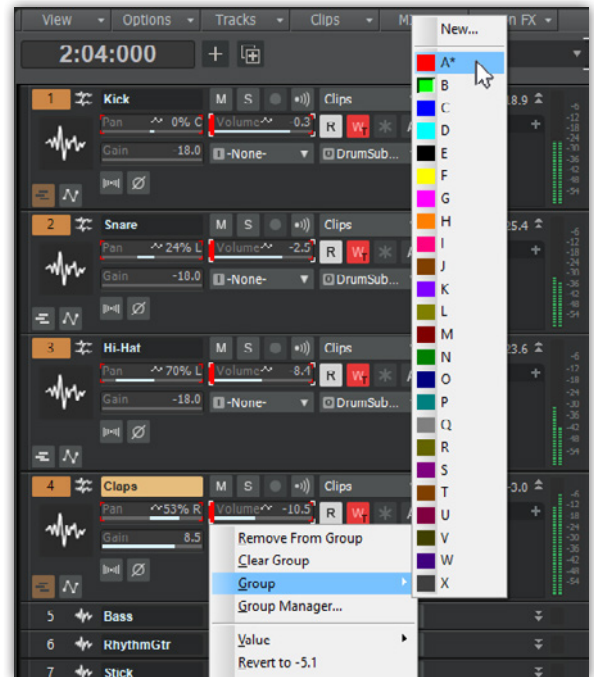
accidentally, didn't know why all their faders had reset to 0dB, and thought there was a bug. However, this is an extremely convenient shortcut, so go to Edit / Preferences / Keyboard Shortcuts, type Offset in the search box, bind it to O, then click on Apply and OK (see Screen 3). You'll thank me for this.

The Write Stuff

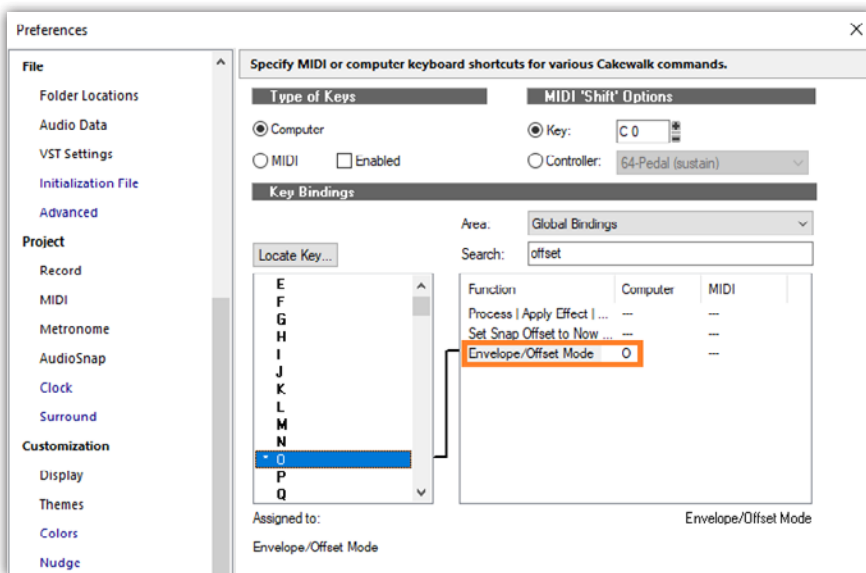
Another advantage of VCA channels is that when you're controlling the gain of multiple channels, you can write automation to multiple channels simultaneously just by varying one fader. Although using a VCA channel to do so is convenient, it takes only a few more clicks to do this in Cakewalk.

Suppose you want to automate volume for several channels. Group the channel faders together for which you want to write automation. Write-enable all those channels, and then moving any one fader will move all of them, and write the automation to all the tracks. To 'solo' automation for one channel, hold Ctrl while moving that channel's fader.

However, note that you can't do this for more than one fader at a time. For example, suppose you have two violin and two cello tracks and are writing the same automation for all of them. If you then want to continue writing automation for one of the violin tracks while the other tracks remain static, you can do that, but you can't continue writing automation for two of the tracks while the others remain static.



Screen 4: Right-clicking on a fader lets you assign it to a group. Note that multiple tracks are selected, so assigning one control to a group will assign the same control in all selected tracks to the same group.



Screen 3: Offset mode toggling can be assigned its own keyboard shortcut.

An alternative method involves quick grouping. Select the channels for which you want to write automation by Ctrl-clicking on the channel numbers at the bottom of the Console. Again, select write for all the channels. Hold the Ctrl key so that moving one fader moves the faders in all selected channels, and write your automation. Even while the transport is moving, you can Ctrl-click on one of the selected channel numbers to deselect it, then hold Ctrl and continue writing automation for the channels that remain selected by clicking on a selected channel's fader.

The bottom line is that while VCA faders become more relevant as a project's complexity increases, for many applications, Cakewalk already has the tools you need to perform many VCA fader-like functions. ■■■

Follow The Leader

LEN SASSO

We show you how to work with irregular tempos and rhythms in Live.

This month we'll see how to use Live's sample warping to align tempo-based material such as step-sequenced clips with freely played audio clips. I'll start with a brief look at how audio clip warping works in Live. You'll find more details as well as some creative uses for warping in the March 2010 and August 2012 Live columns. For what follows, it's best to turn off 'Auto-Warp Long Samples' and to set 'Loop/Warp Short Samples' to 'Unwarped One Shot' in Live's Record-Warp-Launch

1. Use Live's Clip view to warp and edit clip playback settings.



preferences. (The best long-term settings depend on the kind of work you do with Live.) If you're already comfortable with warping, you might want to skip to the next section.

Warping

Start with a clip from your audio samples library that has not already been warped in Live and drag it to the beginning of an audio track in Live's Arrangement view. As shown in screen 1, Live's Beat-Time ruler along the top of Arrangement view shows the length in bars and beats, whereas the Time ruler along the bottom shows the clip's playback time in minutes and seconds. Beat-time is a function of Live's tempo and time-signature settings; change either and the clips Beat-Time duration will change while the playback time remains the same. Aside from shortening the clip from either end, you cannot change the playback time without activating Live's warping, which you do by clicking the Warp button in Clip view. Do that, and a new world opens up.

When you turn warping on, Live inserts an orange Warp marker at the beginning of the clip and assigns the song's tempo to the clip in the clip's 'Seq. BPM' box. The clip will then span its natural playback time as shown on the Time ruler. You can change the clip's playback time either by changing

the song's tempo or changing the clip's Seq. BPM setting. (If the Seq. BPM numerical won't change, select the warp marker.) Once warping is enabled you can add and move Warp markers to force different parts

that the Seq. BPM setting is the tempo between the selected warp marker and the one to its left. The first warp marker shows the same tempo as the second warp marker if there is one; otherwise it shows the clip's global tempo. Below Seq. BPM you'll find buttons to double and halve the tempo, and below those, a menu to select the Warp mode, which is how Live handles differences in the playback rate caused by moving warp markers. For these examples, you'll want to choose the Complex or Complex Pro warp mode.

The Piano Leads

Screen 1 shows an eight-bar, freely played piano clip, most of whose notes do not fall on standard grid divisions. Adding parts such as bass and drums by playing, using a step sequencer or manually entering notes, might seem like a heavy lift, but in Live's Arrangement view, it's quite easy.

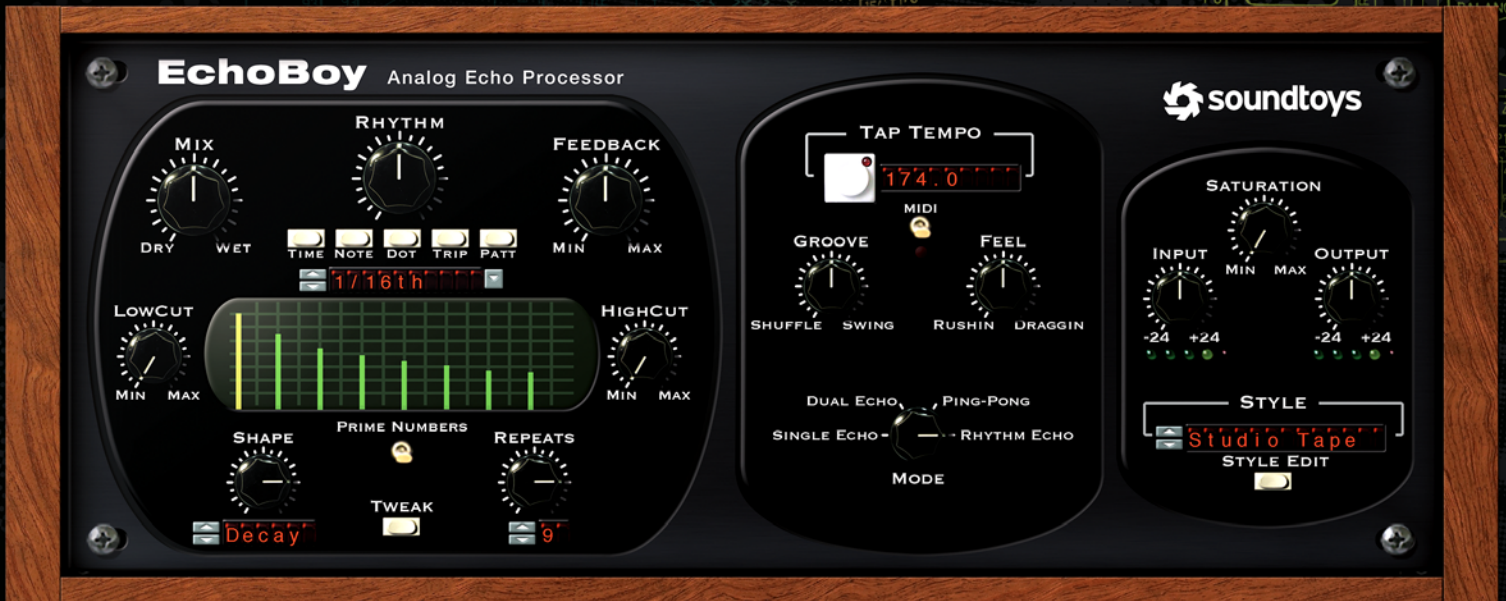
of the clip to play at different speeds. This happens because the number of samples between adjacent warp markers remains the same while moving one of the warp markers changes the time between of them. Notice



2. Played bass and step-sequenced percussion parts are added to the piano part from screen 1.

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» The first step is to create warp markers at the notes in the piano clip to which you want to sync and to then drag those warp markers to lines in the grid you want to use when warping other clips. You don't need to worry about how it sounds, because at the end of this process the piano clip will play with its original timing. This is accomplished by clicking the button labelled 'Follower' below the Warp button, which will then change to orange and be renamed 'Leader'. This inserts tempo automation on Arrangement view's Master track to play the Leader clip at its unwarped rhythm — moving warp markers closer together decreases the tempo and moving them further apart increases it. The resulting tempo changes will, of course, also apply to any other MIDI or warped-audio Arrangement view clips playing at the same time. When you're adding parts by playing, you might prefer to play against the warp-quantized version of the piano with its consistent tempo and rhythm. To do that, leave the piano clip in Follower mode to record the part and then switch it to Leader mode to have both parts conform to the piano's original rhythm.

In screen 2, six new warp markers have been added to the piano clip and then dragged to the desired gridlines. Moving the warp markers has increased the Beat-Time clip length to 12 bars, but that makes no difference because the piano clip is the Leader and the playback time between individual warp markers (and for the whole clip) is unchanged. The MIDI bass part (green) has been played at a steady tempo with the piano part in Follower mode. Then with the piano part in Leader mode, the MIDI percussion part has been step sequenced.

Notice that when any clip is in Leader mode, the Master track's tempo automation is greyed out and cannot be edited. If you turn Leader mode off, the automation disappears, but you can get around that by right-clicking in the automation lane and choosing 'Unfollow Tempo Automation.' That leaves the automation in place so that

you can tweak the tempo and have the adjustments apply to all tracks.

Up to this point we have relied on tempo wiggling to align MIDI and audio clips with a Leader clip. You can eliminate the need for tempo automation by rendering the tracks as unwarped audio files, and you can do that for the mix or for the individual clips. To render individual clips, select the desired Beat-Time range and then select each of the tracks you want to render (in this case, the Piano, Upright Bass and Percussion tracks in the first 12 bars). Next, invoke Live's Render function (Shift+Cmd+R/Shift+Ctrl+R). Now choose 'Selected Tracks Only' from the Rendered Track drop-down at the top of the Render dialogue, ensure the Render Start and Render Length settings are correct and click the Export button at the bottom. Live will produce a separate audio clip for each track. Played together without warping, they will replicate the warped, tempo-automated mix.

When Conflicts Arise

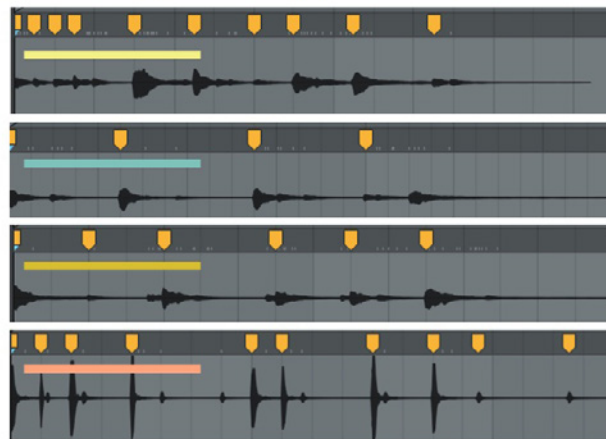
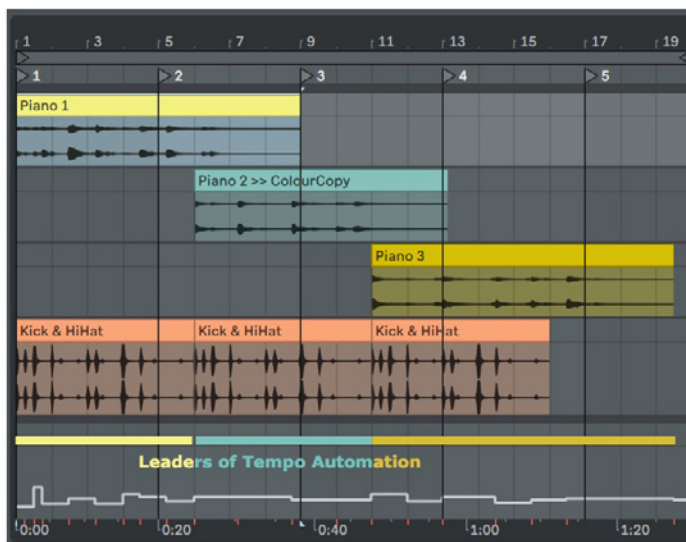
In more complicated setups you can have Leader clips on several Arrangement view

tracks. When you do that, the Leader clip on the lowest track rules. In screen 3, I've placed different freely played piano clips on separate audio tracks. They overlap slightly and have different pan positions and audio effects processing. Each has been warped and made the Leader in order to control the timing of the five-bar kick and hi-hat loop in the bottom track. Notice how the Master track's tempo automation pattern changes as it follows the three different leaders. To belabour the obvious: tempo automation ensures that events in the lowest Leader clip occur at their natural time and that events at the same beat-time position in the other clips are warped to that time.

Leader warping is also good for aligning tightly quantised material, step-sequenced, mouse-entered notes and for small changes in the groove. Live's Groove Pool (covered in the November 2013 Live column) is the easiest and most flexible way to match a short repeating groove, but for capturing the groove of longer clips, 'follow the leader' is the way to go.

Finally, you can use Leader warping to save and restore existing tempo automation. It requires a few steps, but it's a lot easier than redrawing the automation in the Master track's Song Tempo envelope editor. Once you have the tempo automation you want to capture, create a MIDI track and insert an instrument on it (an empty Live Instrument rack will do). Create an empty MIDI clip on that track the length of the tempo automation to be captured. Freeze and Flatten the MIDI track. You'll get a silent audio clip with a warp marker corresponding to each tempo change. Save that clip in your library or project folder. To use the tempo automation at any position in any Live song, simply add the saved audio clip, activate its Leader button (ensuring it is below any other Leader clips) and then right-click in the Tempo Automation lane and choose 'Unfollow Tempo Automation.' You can then delete the clip or the entire track. ■■■

3. Three freely played piano parts share warping leadership of a looping kick and hi-hat track. Warping for all tracks is shown at the bottom.



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Master Blaster

Get the most out of DP's MasterWorks plug-in suite.

MIKE LEVINE

D P's MasterWorks (MW) plug-in collection is comprised of five dynamics processors and an equaliser. All are deep, powerful and flexible, and provide you with high-quality tools for mixing and mastering. In part one of this two-part series, we'll look at three of the processors: MW FET-76, MW Leveler and MW Compressor.

Saturated FETs

The original UREI 1176 is one of the most celebrated hardware compressors of all time. The MW FET-76 plug-in provides an accurate emulation of the 1176N, which was probably the most acclaimed iteration of the hardware unit.

From a control standpoint, the MW FET-76 is almost identical to the original. Both have a fixed

The MW FET-76 emulates the performance and controls of the hardware 1176.

threshold, and you use the input gain control to raise the incoming signal and trigger the compression. The higher the input signal, the more compression.

Like the original, the MW FET-76 has four Ratio buttons — 4:1, 8:1, 12:1 and 20:1 — which can be selected one at a time or in combination. The FET-76 differs slightly from the original unit in the method for combining the ratio buttons. It features an extra button called Compression Combination, which needs to be activated before you can engage multiple ratios.

Just like on the original hardware, the attack and release controls operate in reverse compared to most other compressors, so the settings get faster as

you turn the knobs counterclockwise. If you adjust the attack knob on the FET-76 fully counterclockwise to its off position, it turns the compression off completely, but your signal still passes through the modelled circuitry and benefits from some analogue-style coloration. (This can also be achieved by unpressing all the ratio buttons.)

The FET-76 sounds and performs quite similarly to the original. Its response is extremely fast and therefore useful for controlling transients. Like the hardware unit, it adds some saturation to the signal, and it imparts its tonal signature on everything it's used on.



One source it's particularly flattering to is drums, whether on a bus or individual track such as a snare. To dial in a 'crushed' drum sound, use the Compression Combination button to turn on all the ratio buttons (a setting known as 'all buttons in' on the original). Adjust the attack and release knobs to about half way, and experiment with the input and output controls.

The FET-76 is also quite effective on guitars, basses, vocals and almost any source where a little added character is desired along with dynamics control.

The Great Leveler

The MasterWorks Leveler is also an emulation of a vintage compressor. It

emulates the famed Teletronix LA-2A. The LA-2A (LA stands for Leveling Amplifier) was the quintessential optical compressor and had a tube output stage, which further warmed up its sound. The combination of its tone and its prodigious ability to even out dynamics has earned the LA-2A classic status. The MasterWorks Leveler models the circuitry of the original to produce similar sound and characteristics.

An optical compressor takes a copy of the incoming signal, turns it into light and sends it through a photoresistor that triggers the compression. The louder the input signal, the brighter the light and the more compression. Optical compressors have a slower response than VCA or FET compressors, so aren't as useful for constraining transients. That said, they can still sound great on percussive sources like drums in many situations because of the tonal smoothness they add.

One of the sources on which the hardware LA-2A really shines is vocals, »



The MW Leveler offers a close sonic emulation of the LA-2A, and its expanded control set helps you dial in various vintage characteristics.

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» and the MW Leveler is perfect for that application, too. It will comprehensively reign in the dynamics of the track while at the same time adding a warm sheen.

MOTU equipped the MW Leveler with several extra controls that weren't on the hardware versions. The four Opto-Coupler buttons at the bottom emulate different iterations of the LA-2A hardware. The terms 'Fast' and 'Slow' on those buttons indicate their attack characteristics. The Response knob also affects attack time.

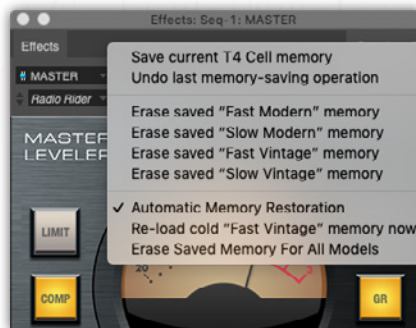
MW Leveler's circuit emulation of the original is so accurate that it mimics the behaviour of the photoresistor cell in the detector of the hardware unit, which had to 'wake up' for a moment to reach its optimal state when first turned on. In addition, the cell will behave differently depending on the characteristics of the audio signal you run through it while it is warming up, achieving a response that is truly unique to that audio material.

Because the plug-in acts the same way, MOTU included a Standby button. You can use it to turn off the processing of MW Leveler without affecting the readiness of the emulated cell, which is not the case if you use DP's standard plug-in bypass button. If you want to compare the sound of a track with and without the MW Leveler on, use Standby to turn the unit off and on, and you'll get the fully 'warmed up' response instantly upon activating the plug-in.

Clicking on MW Leveler's meter opens a dialogue box for saving and recalling the cell state for the plug-in's Vintage Slow and Fast and Modern Slow and Fast modes. Since the cell state can be signal dependent, you can truly 'dial in' its sound for audio material in a specific track. The commands in this menu allow you to preserve and instantly restore that same dialed-in sound. For example, if you achieve results with a certain vocal track that you really like, you can save that cell state and then apply it to other vocal tracks, without going through the warm-up process for the other tracks.

In the Zones

MW Compressor is a multiband dynamics processor that's useful both in mixing and mastering because it allows you to split the audio into up to three user-definable frequency bands — as delineated by two adjustable crossover frequencies — and process each one separately. It's essentially three compressors in one.



Click the MW Leveler's meter to access a well-equipped menu for saving and managing cell states for the four different variations of the LA-2A emulation (Vintage/Modern/Fast/Slow).

The ability to target the compression based on frequency can be very useful. Let's say you're compressing a stereo drum track where the snare is a little loud.

range and the high zone on the cymbals'. Then use the middle zone's threshold and ratio controls so that you're getting more attenuation on the snare.

Setting the frequency bands is made easy by using the solo buttons, which let you listen to a band in isolation. The bypass feature, which allows you to turn off the compression to any of the zones, is also quite handy.

MW Compressor's ability to target compression to specific frequency regions makes it well-suited as a master bus compressor or on instrument or vocal busses. It would also be a boon in a mastering situation when you're trying to alter the characteristics of a particular source within a stereo mix with as little impact on the rest of the frequencies as possible.

“One of the sources on which the hardware LA-2A really shines is vocals, and the MW Leveler is perfect for that application, too.”

You want to attenuate the snare more than the kick, while affecting the cymbals as little as possible to avoid making them sound 'washy'. You can set the low zone to focus more on the kick's frequency

In next month's Digital Performer workshop, we'll look at the other three members of the MasterWorks family: MW Limiter, MW Gate and MW Equalizer. ■■■



With three user-selectable bands, MW Compressor gives you the ability to apply different amounts of compression to low, mid and high frequencies.

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Expanding Horizons

JULIAN RODGERS

In a recent Pro Tools column I wrote at some length about Avid's Pro Compressor, highlighting it as the star of the Pro series of plug-ins. Now standard with active upgrade plans and subscriptions, these have become almost stock plug-ins, and they're very powerful. They might not have the charisma of hardware emulations but they have comprehensive features and well thought-out interfaces, and sound great. You can probably tell I'm a fan.

No plug-in from the Pro series better illustrates this depth of features than the Pro Expander. How interesting can a gate be? Well, turns out it can be more interesting than first impressions suggest.

Could You Expand On That?

A particularly useful control which is overlooked by many, possibly because is isn't well understood and probably also because it has an unfamiliar name, is Hysteresis. Pro Expander and Avid Channel Strip both have a Hysteresis control, and it enables the plug-in to respond differently to sounds crossing the threshold 'from above' compared to sounds crossing the threshold 'from below'. In other words, it offsets the threshold at which the expander or gate closes compared with the threshold at which it opens. This offset is set relative to the main (open) threshold.

To really understand hysteresis, try setting up a simple kick-drum track using a pattern from the Boom! virtual instrument, with an 808 kick on beats one and three at 120bpm and the decay set to its full length. Instantiate a Pro Expander plug-in after Boom!. If you set the kick on beat one at full velocity but the kick on beat three at medium velocity and set the Pro Expander as shown in the screens, you should get both kicks passing through the gate. You'll hear that both kick drum beats get cut off sharply by the very fast release, but the softer kick on beat three gets cut off more quickly, as its tail falls below the threshold sooner.

More acceptable results can, of course, be achieved using a longer release time, but in this case it might be more appropriate to lengthen the Hold time. As the name implies, this leaves the gate open for longer

There's more to Avid's Pro Expander plug-in than meets the ear...

before the release phase kicks in, and in this context, holds the gate open the same amount of time for both kicks, with markedly different-sounding results. Once you've tried that, set the Hold back to its minimum and instead try opening up the Hysteresis control. This time, the gate stays open longer because the gate's closing threshold, for signals falling in level, has been lowered relative to the threshold for rising signals, which are opening the gate. Unlike Hold, it is a level-based parameter, and in this case it gives different results for each kick because they differ in level. Because the softer kick

exceeds the threshold by less, it still reaches the threshold during its decay sooner than the loud kick.

Running Deep

As is so often the case, sometimes the cure is worse than the disease. For instance, if you're trying to reduce the level of unwanted bleed onto a source, the sound of the expander/gate opening and closing can easily become more distracting than the noise it was trying to disguise. Longer release times can help, but using the Depth control can help more. This sets a maximum

amount of gain reduction to be applied, so that rather than cutting the sound completely when it falls below the closing threshold, it just gets reduced. Often, as little as 6dB

A good way to explore the effect of the Hysteresis control is to feed Pro Expander from the Boom! drum machine, using a kick-drum sound with a long tail.





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» of expansion or gating can be enough to push the noise or bleed down out of harm's way, and 12dB is usually enough for all but the most obnoxious of hi-hat spill on a snare mic.

The Depth control can be particularly useful when exploring some of the less traditional applications for gating and expansion, as can the different detection modes at the top of the window. Smart is, as the name suggests, intelligent enough to sort out most scenarios, but I find Average useful when setting up tremolo effects. There is no dedicated tremolo plug-in in Pro Tools, and of all the possible workarounds, I get the best results using Pro Expander keyed from a drum track or other periodic audio source that isn't routed to the rest of the mix. Using short attack and release times can create a chopped effect; lengthen the Hold value to adjust the duty cycle (ie. the relative durations of the loud and quiet parts of the tremolo effect), and increase the attack and release times to soften the modulation. The Depth control sets the depth of the effect precisely.

Very similar effects can be created by using expansion instead of gating, and changing the knee setting to vary the sharpness of the modulation. Whether you see experiments like this as pointless time wasting or an enlightening investigation of your tools very much depends on your point of view, but these interdependencies are what makes dynamics processing such a fascinating area.

Fowl Play

The Depth control comes into its own with one of the most useful and most overlooked features of Pro Expander: the Ducking mode. Most people are familiar with ducking in its best-known application, namely automatically reducing the level of music when a radio DJ speaks into his or her microphone. This is simple to set up by using the mic as an external key for a dynamics processor controlling the level of the music. Most people use a compressor to produce the necessary gain reduction, but ducking as found in Pro Expander is actually a tidier way of achieving the same end, since it reduces the level above the threshold by a fixed amount rather than applying more gain reduction as the voice gets louder.

Ducking is also used in music production to push the level of other elements down to make room for a vocal or a solo instrument — or indeed to make the whole mix 'pump' against the kick drum in some dance styles. A more subtle application is

to duck the level of effects returns so that they bloom during pauses but are pushed back during phrases. This can be achieved using externally side-chained compression but, again, 'real' ducking gives more precise control over the gain reduction.

Join The Dots

When gain reduction is taking place, it's displayed on the graph in the upper central portion of the window, which plots input level against gain reduction. The dancing yellow dot represents the level of the incoming audio (it's red in Channel Strip) and is very useful for understanding the relationship between your attack and release times and the source material. If your time constants are set too slow to catch an incoming peak, you'll see the dot overshooting the line; this shouldn't be seen as something bad, but it is significant, and it is perhaps easier to understand with a gate

than it is with a compressor. For example, when used with a very fast release, the dot follows the line perfectly, but with a longer release time, the dot tracks back downwards diagonally when the gate closes, giving a visual indication of the effect the time controls have on the functioning of the gate. Lots and lots of people have seen this action without realising what it means, myself included!

The relationship between dynamics processors and audio signals follows logical rules, but really getting under the skin of these processes can be frustrating. As well as doing all the jobs you would expect from an expander/gate, Pro Expander is also a really handy educational tool for understanding the way it all works, and one thing I'm sure of is that the more I know, the more I find there is to know. If you feel the same, spend some quality time with Pro Expander. **////**

Going Up

When we talk about expansion we are usually referring to downward expansion, where signals below the threshold are progressively turned down as they fall below the threshold. Pro Expander also offers the much less common upward expansion, where sounds are turned up as they rise above the threshold.

Upward expansion makes the loud stuff even louder. This is the opposite of compression, and can be useful to introduce greater dynamics into a performance, or to undo over-zealous dynamics processing, but things can easily get

out of control. The Depth control can be used to limit the amount of gain increase, capping the maximum variation this effect can introduce as a sort of audio safety net. One quirk of Pro Expander is that the gain-reduction meter stops working in upward expansion mode. This is correct, because in upward expansion mode the processing applies amplification rather than gain reduction, but it does mean that when setting up upward expansion, the only visual feedback is the difference between the input and output meters.



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Secrets Of The Mix Engineers: Konstantin Kersting

After unknown busker Tones and I teamed up with unknown producer Konstantin Kersting, she soon became very well known indeed!



PAUL TINGEN

In 2017, 24-year-old Australian Toni Watson was living out of a van and busking in Byron Bay. Two years later, as Tones and I, she has a number-one single in more than 30 countries. Discovered and managed by music lawyer Jackson Walkden-Brown, Watson spent the whole of 2018 busking and gigging, using her setup of Roland Go:Keys and Casio keyboards and a Boss RC300 Loop Station. As the crowds got bigger, Walkden-Brown teamed up with Lemon Tree Music, a management company with

another former busker, Tash Sultana, on their roster.

Tones and I's first single, 'Johnny Run Away' was uploaded on February 15 of this year, immediately went viral, and ended up double platinum in Australia. The follow-up, 'Dance Monkey', was released on May 10, and gradually grew into a global phenomenon. By mid-November, the song had enjoyed its fifth week at number one in the UK, and its 15th in Australia.

Monkey Business

The staggering success of 'Dance Monkey' hasn't only been life-changing for Watson,

but also for her main producer, Konstantin Kersting. In his studio in Brisbane, Kersting engineered, produced and mixed both 'Johnny Run Away' and 'Dance Monkey' as well as two other songs on Tones and I's six-track debut EP.

Kersting's first taste of success in this genre was the song 'Better' by young Australian singer Mallrat. "I co-wrote that song with the artist and it became a breakthrough thing, because it was both my first foray into collaborative writing, and into electronic music production, where pretty much only the vocals are recorded, and the rest is programmed. When you

record a band, most of the time you really are just a facilitator. You make sure your signal chains and gain staging are right, and you coach them during recording, but that's it. By contrast, co-writing can sometimes feel more creative and exciting. That song became quite successful in Australia, and from there I started working more with signed artists, mostly in the alternative pop genre."

Kersting's studio in Brisbane includes hardware instruments such as a Moog Voyager, Korg Minilogue, Behringer Model D, Fender Jazz Bass and Stratocaster, and quite a few pieces of outboard. Amongst them are an eight-channel SSL X-Desk, an SSL X-rack with four EQs and two preamps, Lynx Aurora 16, Apogee DA-16X and Crane Song HEDD converters, UA LA3A, LA2A, 1176, Smart Research C2 and Drawmer 1968 compressors, Sound Workshop 262 spring reverb, Roland R880 digital reverb, Roland RE200 Space Echo, and an API Lunchbox with JLM 99V preamp, JLM LA500 compressor and PEQ500 equaliser.

"The studio definitely is a hybrid setup now," comments Kersting. "I have a few synths that I use heaps, like the Korg Minilogue, mostly for pads and twinkly things, and the Moog Voyager for bass and for classic lead sounds, and any weird, crazy arpeggios. Until recently I also had a Roland Juno in the studio, but I ended up getting rid of the latter, because the TAL plug-in version sounds pretty much the same and is less noisy. For bass I often use

either the Model D or the Voyager, and/or an Arturia soft synth. My sounds in general are a blend of hardware and soft synths. I just like touching things every once in a while and not just clicking everything in. It might not necessarily sound better, but it makes me feel better about my job.

"I think it's also important that the artist actually plays some parts, wherever possible. Tones always played several parts in the tracks we worked on together. Many of these parts were not complicated, and we could have easily drawn them in, but I feel like it makes it more exciting to actually play them. It also gives you more of a human feel, because you get slightly different velocities on certain hits. In this super-perfect music world, people still like things that sound slightly more human. Playing things in also makes the artist feel more involved, and when they hear their track played somewhere, they think: 'I played that!' It is very important to me to make artists feel comfortable, and like they are part of making the song. You want to have the artist walk out with a song that they are really proud of, and I feel like you get that way more when the artist is in the room with you, rather than just presenting them with the finished track at the end of the production."

Starting Points

It was this artist-friendly hybrid setup that Toni Watson encountered when she entered Kersting's studio in late 2018. "Tones came in with a bunch of demos —



'Dance Monkey'

Written by Toni Watson
Produced by Konstantin Kersting

about six of them. They were all based on the loop arrangements she had created as a busker. 'Johnny Run Away' was the first song we worked on, and this actually already had a demo that she had made with another producer. The demo was really great in terms of structure and overall vibe and gave us a good idea of where to take the track. We just ended up re-producing it all at my place with a more modern take on the overall production.

"By contrast, we started 'Dance Monkey' without a demo. She played me the song in the studio the way she played it live, with her busking arrangement, which was based on a few loops. I also had seen her Instagram video of her doing the song like that. Her busking version

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Konstantin Kersting

» had that main eighth-note keyboard line, the kicks in the pre-chorus, the four-to-the-floor kick in the drop, and the drop bass line. It was a pretty sparse arrangement, but it gave me a pretty good idea of where to go with the song, and how to turn it into a pop production that could go on the radio.

“The sounds that she used for her live show were really cool, but we did not use any of them for the final version. Improving on these sounds was one of the main aims of the production. Her original keyboard sounds worked well, so we tried to keep the sounds in a similar vein, but for example her bass sound was from her Casio keyboard, and instead we played the part in on the Voyager and the Model D. We also put in some extra parts, like 808s in the pre-chorus, and we made sure the claps sounded really good. I ended up with five different kick sounds in the chorus, making sure the blend was right to get the right sound. In general, it was a matter of taking elements from the original demo, and then improve on them and create the dynamics and forward movement in the song.”

Growing Pains

Kersting paid particular attention to making the song build effectively. “I try to do that

delayed gratification thing with most of my productions, with the last chorus being the biggest moment. I don’t like it when every verse and every chorus are the same. Sometimes you have to give away more earlier on, but I love it when you can get a song to build like that, and the last chorus comes in and everyone goes: ‘There it is!’ You have to let the listener know that you’re still with them, and are not just repeating loops, and that there’s something different going on for each section of the song. If you give everything away right from the start, by the time you get to the last chorus, it can get a little boring.

“Most of the building of ‘Dance Monkey’ was done here in the studio. Tones’ version was more or less the same all the way through. In the final version the first chorus has just a kick, the bass line and a snap, and the second pre-chorus and chorus adds some more elements. The second chorus actually is a double chorus, with backing vocals and a guitar and keyboard entering, and the final chorus has big group vocals, guitars, keyboards, and so on. I was filling things out, and was really intent on giving that final push in the last chorus, adding things that she could not do with her Loop Station.

“The first thing we laid down was the piano, because it’s the most important and

driving part, and after that the beat, and then the bass part. I think we started with an actual piano sound, but we ended up changing that to a more synth-like sound, and used the Korg Minilogue for that. Toni played the main keyboard parts in and then I’d quantise these parts, and run them through whatever synth we felt was the best fit.

“The beat was next, and it was all programmed in on the computer. We started the kick and the claps and filled that out with details. Initially we had claps everywhere, but then I decided to scale that back, because it was too intense. Many of my drum samples are from a company called That Sound, in Nashville. I have all their stuff, and it’s really cool. Some of my drum samples come from Splice.

“Once the beat felt good, we moved to the bass. Tones played it in, and I then locked it to a grid, making sure it still sounded human in terms of the velocities she had played. She played the part on my MIDI keyboard, and I then ran those parts through my Moog Voyager, Model D and a Moog bass sound from the Arturia Mini V-3. It’s an emulation of the Minimoog, and is part of their V5 synth collection. I also swung the bass in a particular way, making it a bit more jagged, and this turned out to be very important in terms

Sun, Sea & Studios

Konstantin Kersting is originally from the German capital Berlin. He arrived in Australia in 2007 as a teenage exchange student and, wanting a beach and a non-urban, Australian lifestyle, settled in the coastal town of Caloundra. "I played violin from age four or five," recalls Kersting, "and played in orchestras, but let go of that when I turned 16, because playing the violin seemed hopelessly uncool. I switched to bass guitar, and played in jazz bands for ages, and continued to have music lessons, which gave me a background in music theory. I also play guitar and keyboards, and a little bit of drums.

"A few years after I arrived in Australia, I got a Bachelor of Music degree at Queensland University of Technology in Brisbane. After one year there I also started working as an assistant in the studio for one of my lecturers, Yanto Browning, who became my mentor. At Airlock Studios — owned and operated by Ian Haug of Australian band Powderfinger — I worked my way up from assistant to second-in-house engineer, and after a few years I eventually went completely freelance."

It was only four years ago, when he moved to his current studio, that Kersting started getting into "the more electronic side of music production", a move which was kickstarted when he participated in a 2016 Mix With The Masters seminar with star mix engineer Michael Brauer. "I learned things about mixing that have become part of my approach, but mostly it was seeing how the other people at the seminar worked that pushed me towards a different perspective on things. After that seminar I got more into the electronic side of producing, which is a whole different thing. Some people are amazing studio engineers, but they would not know how to create a cool trap beat. I had to relearn a lot of things, and today electronic productions are 90 percent of what work on, mostly with female pop singers."

of driving the song along. Once we had the bass sounding right, it was a big moment in the studio, even though it was still early in the day!"

Finer Points

The arrangement for 'Dance Monkey' sounds deceptively simple, but a lot of detail went into the programming. For example, Kersting eventually filled in and refined the keyboard arrangement to 19 tracks. "In addition to the main Korg Minilogue synth, there's a piano sample, which sounded a bit too much like a real grand piano, so I added an upright sample and a steel drum sound. I found that more exciting than just having a straight piano sound. In addition, there are tons of little keyboard parts, with pads entering in the second chorus, using sounds like from a Juno, and Solina, and Farfisa organs, plus some more contemporary touches like vocal chops. The pads are mostly playing the same parts, the odd different inversion aside. The idea is that all the different sounds make one cohesive sound together."

Most of the recording and production took place in a single day. "She was around for the whole day, and we were throwing ideas around and having a great time. I guess the main thing was to make the drop feel really good, make the bass line pump and make sure the kick sounds good. I mixed the song the next day, and that mix existed for a while, but then Tones had the idea of having the group vocals in the final chorus. She wanted her friends on the track, and also because she lives in a different place she recorded these group vocals locally, and sent them to me. I mixed them in, and after that there were a few recalls, with Tones each time saying, 'Turn up the group vocal!'"

It's very common these days to start the mix process during recording and production, often rendering the final mix an afterthought. Kersting is not fond of this approach. "I try to keep >>



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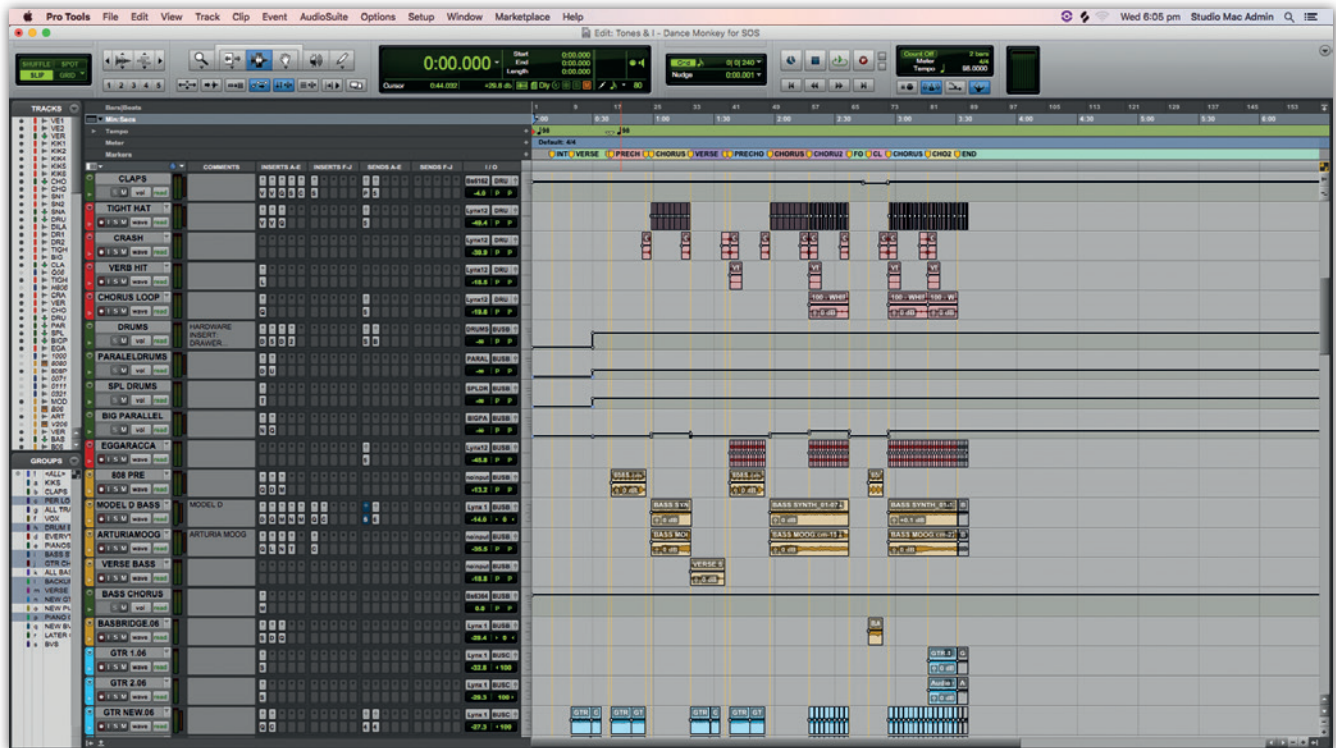
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This excerpt from the (very large!) Pro Tools session for 'Dance Monkey' showcases Konstantin Kersting's careful organisation, and also his extensive use of subgroups and parallel processing, especially on the drums.

» production and mixing separate. I obviously create a blend during production so I have an idea of the track, and there'll be some vocal effects and vocal throws and some basic EQ to make everything sound good. But that's it. Mixing is a totally separate process for me. I know people want rough mixes, but I kind of hate doing them, because you end up with demoitis, and are fighting the rough later on. It's nice to have lots of wiggle room for improvement when you get to do the actual mix!"

Different Space

Because Kersting doesn't do any serious mixing during production, the first stage of his mix process involves getting his session mix-ready. "I go into a mix zone, mentally, and I print any software instruments that may still be in the session, make sure everything is organised in a way that makes sense to me visually, get all my busses in, and my aux effects, and so on. It really is a totally separate process for me. I have a template, but it's always changing, as does my master bus.

"When I first started mixing years ago, I did all sorts of crazy things in analogue, with splitting things out with crazy parallel compression, but the recall issue just became too frustrating. The more I'm doing, the more I'm moving away from outboard, though I could not do without my Roland Space Echo, which is a really

cool piece of kit, or my Moog Voyager, which sounds awesome and no emulation of it is as good. And I love the C2 on the master bus. These are some of the key pieces of hardware that I continue to use."

The extremely well-organised Pro Tools mix session for Kersting's 'Dance Monkey' mix contains 116 tracks. His colour-coding scheme has most aux tracks coloured dark green; audio tracks for drums (top) are red, followed by bass (yellow), guitar (blue), keyboards (light green), backing vocals (pink), group backing vocals (pink-red) and lead vocals (purple). At the bottom of the session there are 20 more aux tracks, split into 10 aux effect tracks (again dark green), five vocal compression aux tracks, four group tracks for keys, drums/bass, guitars and backing vocals, and a master bus.

"All my busses are added as part of my mix process," explains Kersting. "I tend to do most of my processing on these busses, rather than on individual audio tracks, because I know that the listener is, for example, not going to listen to five separate kicks, but to a blend of all of them. Normally I'll only add plug-ins to individual tracks if there's something particular bothering me about them."

Drums

The layered kicks, snaps and claps are thus fed to their own busses for processing,

which involves EQ and various forms of saturation ranging from subtle tape and console emulation to more obvious distortion. "FabFilter Pro-Q2 is my favourite EQ. I'll use it any time I want to pull something out or add something. I'll only use another EQ if I want the specific sound that EQ adds — Neve, SSL, API and so on. I use Slate's Virtual Mix Rack generally for console emulations, especially the Neve desk one, and sometimes the API one. I think they sound really good and they are really subtle, and they help a lot if you have many instances of the plug-in. That warms up the sounds in a nice way. It's similar with the Slate Virtual Tape Machine, which I think sounds slightly more subtle than the UAD tape emulation. SoundToys Decapitator and Devil-Loc and FabFilter Saturn are all good for distortion. The Pro-C2 is an amazing compressor that does exactly what you want it to do. I use the FabFilter stuff a lot, because it is so good and so straightforward to use, and it does not colour the sound."

All Kersting's audio drum tracks end up going through his 'Drums' aux, either directly or via the individual instrument aux tracks. Some of these latter tracks, and the 'Drums' aux itself, have sends to three parallel tracks: 'Parallel Drums'



Konstantin Kersting's Moog Voyager was one of the instruments layered for the 'Dance Monkey' bass line.

2500. The Drums bus also has sends to the 'SPL Drums', adding some attack and sustain to the drums, and to the 'Big Parallel' bus, which has the Novatron set to 'Punish' mode, and a Pro-Q2 to roll off some high end, because when you do parallel compression you sometimes end up with super-bright stuff. I added the Novatron because I felt that the drums in the chorus needed more impact. I created a blend of these four drums aux tracks to get one cohesive sound."

Music

The 808 which Kersting added to the pre-choruses is part of the bass section, and is the first track of this section. The bass and guitar have a slightly different routing structure, with no parallel processing, and there are tons of plug-ins on the inserts of the Model D and Arturia

with the SoundToys Decapitator and Kush UBK1 compressor, 'SPL Drums' with the Native Instruments Transient Master and 'Big Parallel', with the Kush Novatron compressor and Pro-Q2.

Kersting: "This main 'Drums' aux bus has an insert to my Drawmer 1968 hardware compressor, and there's a Decapitator adding some saturation, and yet more compression from the UAD API

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The main piano-like synth part came from Kersting's Korg Minilogue.

» Moog audio bass tracks. A send from the Model D bass goes to a 'Bass Chorus' aux with the SoundToys Microshift plug-in, and the two main guitar parts each have sends to a 'Guitar Slap' aux with the SoundToys Echoboy and a 'Guitar Reverb' aux with the FabFilter Pro-R.

"There's a Decapitator on the 808, because you need a bit of distortion on them to make them cut through on small speakers. I check this on my Auratone 5C monitors. I even try to make sure you can hear 808s on a phone speaker. You have to make sure that your mix sounds good on all those devices, as well as on your main studio speakers, which in my case are the Quedest VS2108s.

"There are no fixed rules. It is about whatever gets me to make it sounding the best. In this case that meant lots of plug-ins on some of the audio tracks, particularly the Model D bass, which is the main bass you hear in the track. I'm pulling out quite a bit of sub and high mid-range with the Pro-Q2 and sub with the FabFilter Pro-MB, because the bass was blowing out my sub. The Decapitator and the Dada Life Sausage Fattener add distortion in the lower mid-range, again to make sure the bass sounds good on small speakers. In addition, there's a Kush Audio Novatron compressor to keep the bass in place. Regarding the keys, many of the plug-ins on them were added during production,

because I was layering several sounds to get the right textures and vibe."

Vocals

As already noted above, the vocals are divided into three sections: harmony vocals (which sound like a medieval choir), the group backing vocals, and the lead vocals, which are split up over 10 tracks. Each of these three sections has its own group aux, with tons of plug-ins. Most work clearly went into the lead vocals, as the audio tracks also have a fair amount of plug-ins, mostly Waves Tunes LT, FabFilter Pro-C2 and Avid D3 De-Esser. The 'Lead Voc 1' aux has nine inserts and 10 sends, plus there's a supplementary 'Lead Voc 2' aux, with five inserts and five sends, adding yet

another layer of effects.

"There is actually not that much happening!" insists Kersting. "All these plug-ins do a tiny bit. I split out the lead vocals over several tracks to be able to treat each slightly differently. I felt that the Waves Tune plug-in sounded best on Tones's vocals, so I used that, just to nudge a few notes here and there. Plus there's a compressor, the C2, and a de-esser on each of the audio tracks. Three tracks are marked '-8', and they have the SoundToys Little Alter Boy, which lowers them by an octave, and there's a track with the Tal Vocoder, which is the vocoder for the middle eight.

"All these vocal tracks apart from the vocoder track go to the master vocal bus, 'Lead Voc 1', and on that I have the FabFilter DS de-esser, and then a hardware insert to a Urei LA2A, which is not doing a lot. It's quite a slow compressor, and I always have it on the limiting setting, just taking off little tiny bits. Then I have the UAD Neve 1073 emulation, set to the drive input, like you would on an actual Neve preamp. So I turn it up a little bit, and then turn the fader volume down, which I find gives a really nice sound. Then I have the Slate VMR, with a Neve console emulation, an 1176 emulation, another harmonic saturation thing, and a slight compressor.

"After that, I have the Pro-Q2 again taking out some mid-range, adding a little bit of top end from 5k onwards, and getting rid of low end. Then it goes into the Slate VMR, which is adding harmonic saturation again, then it goes to the Pro Tools de-esser, because I added some highs with the EQ. And then it goes to the Pro-MB multiband, which is working pretty much in the same mid-range as where I cut with the EQ. I must have wanted to tame her vocals a little bit more. It's also taking out some



Konstantin Kersting still uses hardware processing, especially on the master bus, where his Smart C2 compressor was employed on 'Dance Monkey'.

Vocal Tones

One of the things that has helped drive the success of 'Dance Monkey' is Toni Watson's very distinctive vocal sound. Konstantin Kersting explains that this isn't the result of monkeying about in the studio. "I used my Miktek CV4 large-diaphragm microphone on her, and that went into my SSL XR627 preamp, which sounds really good on vocals. I ran it through the SSL XR425 EQ after that, and then the Urei 1176 and into the computer via my Lynx Aurora 16. In general, I try to minimise the amount of compression on the way in. Tones' voice is so expressive, the 1176 was just about containing the peaks, as opposed to adding a sound.

"We did maybe four or five takes, and that was it. She's a really good natural singer, and the way it sounds is just the way she sang it. It's unique, and unlike anyone else, and I guess it's why people have taken to the track. I barely did any tuning, apart from manual corrections on a few notes, and there's no added distortion, or things like that. I thought of the song as a left-field pop song, and that kind of stuff does not need to be 100-percent perfect. The main thing I had to do was to control the sharp, high mid-range that her voice has, using EQ and multiband compression. I did this during the mix."

of the 's' sounds again, around 10k. Finally, there's a gate, to take care of any noise from the LA2A outboard compressor.

"These are the inserts on the lead vocal bus. After this are the effect sends. The first five go to a slap, a long delay, a reverb, a small plate and the vocoder track. The second round of sends go to the five orange aux tracks, which all have compression. I'm mixing into compression with the levels of the compressors set, Michael Brauer-style, and I then blend these five tracks together to create a cohesive vocal sound. The compressors are the Kush Audio UBK1, which can be super-saturated and quite squishy-sounding, the Slate emulation of an 1176, a Slate VCA compressor [*wrongly labelled in the session*], an Empirical Labs Arouser, and for the Vox Distortion bus, the Decapitator.

"It looks like a lot, but again, it is just a whole bunch of compressors doing small things together. I find that my vocals sit a lot better in the track since I started doing this. They are easier to mix in, and I need less EQ. I actually create my blend of the compressors first, because generally my vocal sound will be kind of dictated by the blend of those compressors."

Master Bus

The vocal tracks go directly to output 4 of Kersting's Apogee DA-16X, while the rest of the session is sent to the unit via his four 'global' aux tracks, respectively keys (A) and backing vocals (D) to input 7-8, drums and bass (B) to 1-2, and guitars (C) to 5-6. This makes the entire session fit his eight-channel X-Desk.

"The X-Desk sums to a Crane Song HEDD, and from there into a hardware passive JLM PEQ and a Smart C2 compressor, again doing very little, at the most taking off 3dB, and then back into Pro Tools. In Pro Tools I have whatever sounds good on the day on my mix bus. In this session I had the Slate VMR, with a Neve console emulation and iZotope Ozone 7. That goes to my Slate Digital FG-X mastering processor, which is adding some transients, and from there it goes to my Kush Audio Clariphonic DSP, which I love. You put it on and it is like a blanket has been lifted off your mix! At the end of the chain are the Slate VTM tape emulation, and the FabFilter Pro-L, which is just there to make sure that there are no unexpected peaks. I never add limiting even when I send out listening copies. My mixes generally come in at -12LUFS, which is already louder than Spotify. The mix that I send for approval is also the mix that I send to mastering!" **////**



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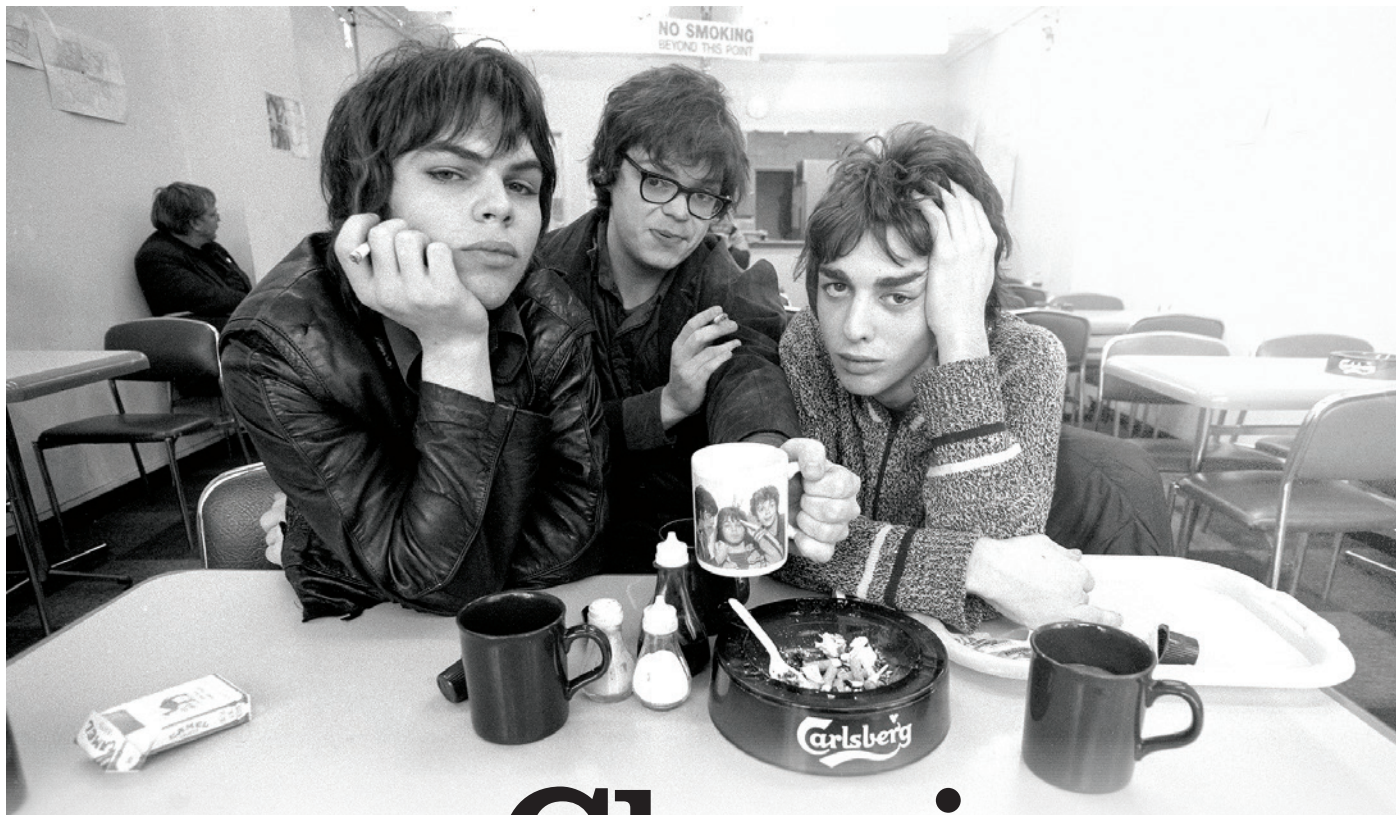
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TOM DOYLE

In the crowded field of mid-'90s Britpop bands, Supergrass stood out. For a start, they were surprisingly young — singer Gaz Coombes was only 18 when they released their 1994 debut single, 'Caught By The Fuzz'. They were also in possession of a strong, boyish retro rock-band look and a different set of influences from their peers: Ziggy Stardust-era Bowie, Lou Reed, Iggy Pop.

But it's for their UK number two single of 1995, the irrepressibly bouncy 'Alright', that Supergrass are best remembered. In the song's video, their image was frozen forever as a Monkees-styled trio capering around on bikes and in a bed on wheels rolling along a beach, as Coombes sang a cheeky lyric about teenage delinquency: smoking fags, sleeping around, crashing a car in a field.

However, at the time, the success of 'Alright' was both a blessing and a curse for the band — Gaz Coombes (vocals/guitar), Mick Quinn (bass/vocals) and Danny Goffey (drums). It made their singer in particular a reluctant household face in that summer of 1995. "When 'Alright' went mental, we were in America," Coombes remembered in an interview with this writer four years later. "We got back and suddenly every fucker was recognising you. I never wanted to be a rock star. I just wanted to be in a band."

Almost 10 years after they split — and

Classic TRACKS

Supergrass 'Alright'

When producer Sam Williams discovered Supergrass he knew he had to capture the band's infectious energy on tape.

in the wake of Gaz Coombes enjoying a successful and critically-acclaimed solo career — Supergrass have re-formed. 2020 will see them back on tour, in support of a new career retrospective box set, *The Strange Ones*. Producer Sam Williams, who discovered the band and oversaw the making of their debut album *I Should Coco*, clearly remembers the day that he first encountered the trio in the street in Oxford in 1993.

"I was in a music shop and I came out and saw the boys standing on the pavement," he says. "It was one of those classic, surreal moments. I'd grown up with strong references to early Beatles and the Monkees and the cartoon kind of culture of larger-than-life '60s-looking bands. They

didn't look like anything that you'd seen in real life for a long, long time. Danny was wearing a blue velvet suit with red Bowie hair and Gaz had the kind of Neil Young sideburns.

"The feeling was that it was just an immediate, magnetic attraction. Something in me just went, 'I don't know who that is. But I know that's a band and I know I'm going to produce them.'"

First Impressions

As a musician and producer, Sam Williams shared many reference points with the young members of Supergrass, but by this stage in his career, he had gained much more experience in the music industry. As the son of Len Williams, founder of the

London Guitar School, and the brother of guitarist John Williams — a member of classical rock band Sky and also a soloist, whose rendition of 'Cavatina' famously became the theme from *The Deer Hunter* and a worldwide hit in 1978 — Sam Williams had been in and out of recording studios since he was a teenager.

Growing up playing bass, sax, clarinet, piano, drums and then becoming a singer, Williams had joined his first band at 11 and, by 16, had played bass on 'International Language', the debut single by Richard Strange, formerly of proto-punk band Doctors Of Madness. "They were interviewing him on Radio 1 when the single came out," Williams remembers, "and he was saying, 'Oh yeah, I got this schoolboy called Sam Williams to play bass.' And I was thinking, 'Well, look, if this is possible, then anything is possible.'"

Living in Cornwall by this point (where his father had relocated to found a monkey sanctuary), Sam Williams began to visit his local recording studio, Sawmills. At 18, driven to become a producer, arranger and artist, he was employed by Sawmills as an assistant and moved into the remote, residential facility.

"It's in a place called Fowey on the southwest coast of Cornwall," he explains, "and only accessible by boat or by walking down a railway line. I moved into one of the cabins that was part of the residential thing. Sawmills was just a stunningly beautiful place. A complete sanctuary as well, because obviously it was quite remote. It was like my rock and roll university."

At Sawmills, Williams was trained in old school analogue engineering and to-tape recording. "The guy that owned Sawmills at that point, Simon Fraser, was a fantastic mentor and teacher of a lot of recording techniques," he says. "I quickly got a feel for mixing and rigging up a lot of tape delays. I also got a taste for editing, because a lot of what I wanted to do wasn't easily doable or possible on conventional tape recording. So, we were into the land of click tracks and two-inch tape, but I'd be chopping up two-inch tape to get the results that I wanted that would be closer to the way that we would sequence and loop now."

Obviously, for an engineer, taking a blade to a reel of two-inch tape during a session required a lot of nerve and confidence. "I tell you what, man, it was really scary," Williams laughs. "Even on a basic multitrack edit where you wanted



Artist: **Supergrass**
Track: **'Alright'**
Label: **Parlophone**
Producer: **Sam Williams**
Engineer: **John Cornfield**

verses and choruses from two or three takes, you'd have a lot of pieces of tape hanging around in the studio over different bits of furniture, waiting to be assembled. And if anybody walked in and said something, it could disrupt your concentration and you could be in hell. Some of that did happen on a little bit of editing in that period of the first Supergrass record."

By the time Williams met Supergrass, he was already a veteran of countless sessions at Sawmills and the frontman of his own band, the Mystics, who'd signed to Fontana Records. "Those were the recordings that I initially played to Supergrass when I met the boys," he recalls. "Just as a kind of calling card of... 'Well, this is what I've been doing.' To say, 'Look, this is how it'll sound if I produce and if we go to Sawmills.'"

At the time, comically, and almost like the Beatles in *Help!*, the three members of Supergrass were living virtually door-to-door to one another in a set of cottages in the village of Wheatley in Oxfordshire. Sam Williams first saw and heard them play in a local pub. "I was blown away," he enthuses, "because it was super tight and super hard and very visceral and incredibly connective. Danny's drumming was drawn from a lot of different things — Mitch Mitchell, Keith Moon, and from Charlie Watts for the groove-orientated thing as well. A very creative player, a really great feel player. But intense energy.

"They were playing very fast, so it didn't take them long to play eight songs. They had elements of the Buzzcocks

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The live room at Sawmills.

» and they had elements of the Pistols a little bit with the guitar sound. But again, none of it in particular in terms of an homage or a pastiche. And it was incredibly concise songwriting, a bit like the speed at which the material was delivered.”

Williams first offered to help quickly nail down demos of the band’s songs, recording them playing live on a Fostex Portastudio at Danny Goffey’s cottage, after randomly placing microphones around the tiny room. “Literally what I call Lazy Miking,” the producer laughs, “where you throw a mic on the floor. We were just capturing. I think we may have overdubbed some vocals.

“They would play in a room where you could just get them and me in, and they’d play at full volume. I thought, ‘This is a really good indication as to how this record should sound and feel.’ So, we used the four-track recordings to get to a place where we had a pretty comprehensive sketch of pre-production for six tracks. We had a good synopsis, if you like.”

One of the six songs they demoed together was ‘Alright’, in its original, scrappy form, built around pumping, staccato piano chords. “That was done on a very out-of-tune piano at Danny’s cottage,” Williams remembers. “We’d done everything on a four-track format that we didn’t care about, but that was good enough to make choices and decisions about the arrangements. To get a good

sketch of everything down and an overview of the material that we wanted to cut.”

Sawmills

In the spring of 1994, Sam Williams and Supergrass first travelled down to Sawmills for a five-day session agreed as part of a production deal with the studio’s owners. “We took the boat with the gear in it down the river and they just loved the place,” says Williams. “I could see they got it and it was the right environment for them. It wasn’t going to be overwhelming or over-spec in any way. It was just right... the feeling of containment and kind of like a little bit of naughty isolation.”

Together there the team began working with Sawmills’ in-house engineer, John Cornfield. “My approach as a producer was to enable the making of a record the way I imagined Sam Phillips would work at Sun, or the way I imagined the culture would be at Stax or Motown,” Williams explains. “It was an in-house feeling about it, especially in terms of collaborating.

“I was working with John Cornfield, who is a world-class engineer,” he stresses. “And coming at it from a different point of view that was ideal as an old school team. It allowed me to have a relationship with the band that was fundamentally based on playing, arranging, having fun, and getting the energetic lines of the production right.”

True to his original idea, Sam Williams

wanted to recreate the intensely concentrated sound of Supergrass’ four-track demos. “When I was in the cottages recording that immense racket in a small space,” he says, “I knew how I needed to record *I Should Coco*, which was with the amps mainly in the same room. Because that’s all the band had known.

“I also wanted some element of controlled bleed, even into drum mics. Which is not ideal, if you’re looking for a kind of perfect, separated sound. But I knew it would help in that way that it had with some of the Spector things. I could hear the element of bleed that I liked.”

Williams’ approach had the band playing while wearing headphones, but at the same time still able to hear and feel the familiar blasts of sound from the drums and amps. “If you close a band off too soon when they’re young and give them headphones,” he argues, “they lose physical contact with the acoustic elements of the velocity and the transients and the way that they make contact with an instrument. John and I had a little chat about: ‘OK, we’re gonna do it this way.’ We did spread [*the amps out*] a little bit as the record progressed, but not massively. The energy of the band was in their eye contact and volume.”

The picturesque location of Sawmills certainly helped to create the right mood for the sessions, too. “It was not a big live room,” says Williams. “A nice-sized

Gaz Coombes and Sam Williams, mid-'90s.



room, but with an immensely cool, creative playing vibe. The windows looked out onto a completely isolated creek with swans swimming around on it, and with the woods on the other side.

"If a take was finished or half-finished, we'd get into canoes and row over to the other side of the creek and have a listen to a playback through the windows of the studio. Have a little smoke, go back again, do another take. It was that kind of atmosphere. They loved it."

The control room at Sawmills is centred around a Trident 80B console. "I just love the board," says Williams. "It's a transparent-sounding board. It's got immense mojo, but you also feel like it doesn't obscure the contact with the music. There's a monitor section on the 80B which has this different EQ from the main channels. John Cornfield said, 'Look, Sam,

when you're gonna pop 3k on the guitars, do it there.'"

Monitors-wise, at the time of the recording of 'Alright' and *I Should Coco*, Sawmills offered the choice between Quested 212s and Yamaha NS10s. "The control room was small," says the producer. "A very shallow space. So, you're not looking at a textbook studio for monitoring. But I have to say, still one of my favourites for containment. The sound was very focused, very punchy. A bit like having a big pair of headphones on."

Tracking was done to the studio's Otari MTR-90 MkII 24-track two-inch. The evening before the five-day session was due to begin, Williams and Cornfield set up the facility's Premier drum kit and prepared a drum sound for the next day. "It was [AKG] 414s for overheads," says Williams, "and [Shure] 57s top and bottom on the

snare. Probably Sennheiser 421s on the toms and the kick would have been an old school [AKG] D12. We were tracking pretty quickly the next day.

"My whole production ethos was about capturing it quickly, before it went off the boil. Because this was a young band and this was about capturing a kind of energy that does not hang around. You're not gonna benefit from spending two days getting a bass drum sound."

Straight To Tape

In keeping with this approach, the basic tracking for the six songs Sam Williams and Supergrass recorded that week at Sawmills — 'Caught By The Fuzz', 'Alright', 'Strange Ones', 'Sitting Up Straight', 'Mansize Rooster' and 'Lose It' — were cut with the minimum of takes. "When I'm rolling tape, I want to get it, if I can, inside one or two cuts," emphasises the producer.

"Danny was aware that it was down to him to deliver it in that time spec. In the worst-case scenario, I think with a difficult track like 'Lose It', we may have gone six cuts in a row. And he's sweating and dripping, because every time he's giving it up in that immensely physically demanding style that he drums in. But we would very rarely need to go more than two cuts and often they'd get it in one."

The team were similarly unfussy about the choices of guitars and amps. They simply went with the equipment that the band had at the time, namely, Gaz Coombes's Fender Telecaster or Epiphone SG played through a Sound City 2x12 combo amp and Mick Quinn's Carlsbro

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The control room at Sawmills is still based around the same Trident console.

» bass and amp. "His bass made a farting, blown-up sound which was very much key to the distortion," says Williams. "Same with the vocal sound on 'Lose It'. We had a [Shure] 58 going through a distortion pedal for Gaz to sing straight through the Sound City amp, and that was it."

As the live takes were being laid down, Sam Williams would typically be in the control room, directing the proceedings and trying to bottle the band's energy: "When they were delivering cuts as a power trio, I wanted to be making calls in the spirit of the three-hour sessions that I grew up with as a kid that don't go beyond that time zone. I wanted to have the confidence to stop a take when it was necessary and restart it. In other words, to optimise a cut while it's happening, so that they get the best that they can."

"But most of the time we didn't have to do that," he adds, "because they were playing great. 'Alright' was cut in the space of 10 minutes — two takes, first and second halves cut together from the two-inch tapes. That was it. Then overdubbed with piano and the other things."

To recreate the pub-ish piano sound that featured on the demo of 'Alright', Sam Williams took some liberties with the studio's Rönisch grand piano. "We nearly fell out with the studio there," he laughs. "I got in there with John Cornfield and I just detuned everything. Not randomly, because I knew what would cause that effect. I didn't need to destabilise the whole piano. I just needed to destabilise two out of three strings [of the chord]. So, if I went sharp and

flat on both sides of a relatively accurate string, then you got that pub thing.

"The piano needed to sound inherently out-of-tune. But it didn't go down well with the studio because obviously you've got to put it all back in again. But it worked perfectly for 'Alright'. I think we may have tracked it in octaves, and it did give it a massive sound."

While a guide vocal was always committed to tape during live tracking sessions, typically Williams would re-record Gaz Coombes' lead vocals — and Mick Quinn's characterful falsetto BVs — after the fact, using either an AKG C12 or Neumann U87. "Sometimes guide elements were kept," the producer says. "But often we'd go and cut vocals properly. Mickey was doing incredible backing vocals that were like another personality in the band. Very difficult to record, at the extreme velocity and pitch he was using. But we'd get them right."

On the last of the five days at Sawmills, Williams mixed all six of the tracks that the team had recorded that week, including, of course, 'Alright'. Ahead of the mix session, John Cornfield had set Williams up with a variety of effects sends on the Trident, including to two Revox reel-to-reels used for tape echoes, an EMT 140 plate reverb, a Roland Dimension D stereo chorus, an Eventide DSP4500 harmoniser and a Universal Audio 176 valve limiter.

"The 176 was the original [Bill] Putnam thing which was on the Sinatra and Beach Boys records," Williams recalls. "John used to rig it for our parallel distortion on Gaz's

vocal. There was always a tape slap running on every vocal. There was also an AMS [DMX 15-80S] delay but, most of the time, I'd use tape echo if I could.

"I'm used to getting very hands-on on that board," he adds. "So, John would always be amazing in setting me up with a session built on a great engineering basis. I could get in and do my thing as a mixer and cut and craft and shape sounds."

Round Two

As soon as they were completed, Sam Williams brought the first six Supergrass track masters to the attention of Radiohead's managers, Chris Hufford and Bryce Edge of ATC. Williams remembers, "Chris was pretty much like, 'Well, that's the first six singles laid out. Can you ringfence this? And we'll come back on it.'"

The result was Supergrass signing to Parlophone Records and then returning to Sawmills in the summer of 1994 to complete *I Should Coco*. "It was a golden summer that would never come again," says the producer, "and that's when we cut the rest of the record."

In this second half of the album's sessions, Williams and the band stretched out more in terms of playing and production. "I'd be starting to sit in on keyboards," says Williams. "On the second sessions we had a Vox Continental, so I would be in the room playing on 'I'd Like To Know'. We started using Wulitzer [electric piano] or other keyboards."

Another key track, the mid-paced, early Pink Floyd-y 'Sofa (Of My Lethargy)',

featured much instrument-swapping between Williams and the group, the producer moving onto bass to connect with Goffey as a rhythm section, as Quinn changed over to guitar and Coombes to piano. "I was a kind of floating auxiliary player," says Williams. "So, it was a flexible musical thing that could move around very easily like that."

Varispeeding tape was a trick that Williams and the band began to use more during the second set of sessions. The two-inch master of propulsive rocker 'Lenny', for instance, was sped up to achieve the right feel for the track. "That track was 6 to 10 bpm slower," says Williams, "We'd got an accurate cut of 'Lenny' that was too slow, too rock, and we sped it up considerably. We erased the bass and the guitar, and Gaz and Mickey re-recorded them over a sped-up drum track which then had the exact tempo. Whatever it took, we would do it. There wasn't any kind of purist idea of how you do it."

On 'We're Not Supposed To', meanwhile, Williams took a four-track home demo the band had made, experimenting with tape-speed chipmunk-y voices, and embellished it at Sawmills. "That is so Danny and his sense of humour," he laughs. "I took all the crazy things that he'd done from the four-track cassette, spun it onto two-inch and then rebuilt the guitar, the bass, everything around it."

"Congas were overdubbed with that kind of early T Rex/Bolan thing as an influence. Two acoustic guitars, the left/right [panned] stereo thing, and an elastic band kind of bass sound. So, it's actually quite a polished production around a complete bit of lo-fi. It was all very arranged chaos and probably one of the most complex productions, actually, although some people would mistake it for a comedy track."

Elsewhere, two tracks from the original sessions, 'Mansize Rooster' and 'Sitting Up Straight', were re-recorded during this final stretch of making the album. "Because the songs were requesting an alternative approach to the production that wasn't benefitted by speed," says Williams. "You can hear that we've spent more than an hour [laughs] getting a drum sound. Not that it was bad in the first place, 'cause it suited it perfectly"

"So, in other words, if you had the options open, cut everything at no-brainer speed to start with. Anything that doesn't make it with that methodology, then apply the remit of more expansive, more detailed

production to it. But not the other way around. And that way you're never gonna miss energy, you're never gonna miss capture. You're only gonna expand naturally into things that require it."

Williams thinks that overall the rooms at Sawmills contributed hugely to the tight, fuzzy and energetic sound of *I Should Coco*: not just the live room playing space, but also the control room during the mixing. "I later noticed that people like Flood would mix in rooms that often didn't have a huge space," he says, "that were very flat and contained and based on that principle. I think, considering its limitations, it was one of the best-sounding, punchy, tight, controlled rooms to mix in."

Postscript

For Sam Williams, his main memories of working with Supergrass on their landmark debut album are of himself and the band laughing. "I've never laughed so much making a record," he says. "To the point where you actually had to stop recording. I'm very grateful and honoured that I've been lucky enough to connect with them."

The subsequent chart success of 'Alright', however, he only remembers as

a blur. "It rushed by us like a high-speed train," he says. The producer however is in no doubt as to why the song was set to become regarded as a classic track. "It's incredibly positive," he reasons, "and, actually, it's got two sides to its coin. It's got a very British pub knees-up kind of energy. It's everything that teenage life is about — sex, drugs, rock & roll. It's saying something that you can only really know in that window of time when you're kind of leaving school and before the engagement of other issues becomes unavoidable as an adult."

"But, also, it's looking at identity. It's got a duality which is beautiful: 'Are we like you?/I can't be sure.' Because they were at that age, it was authentic. It was the real thing."

Ultimately, Williams is as thrilled as Supergrass' legions of fans are that the band have decided to reunite. "Yeah, I'm delighted," he states, "because they're a great live band. When I speak to people about Supergrass, I often get the same thing: 'God, weren't they great?'" **///**

Supergrass: The Strange Ones 1994-2008 is out January 24th on BMG.



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OLIVER HELDENS

DJ & Producer

Laptops, loops and layers are the tools that have taken Oliver Heldens to the forefront of the future house genre.

PAUL TINGEN

Six of the top 10 artists in *DJ* magazine's Top 100 for the year 2018 were from the Netherlands, including chart-topper Martin Garrix, Hardwell, Armin van Buuren, Tiesto and Afrojack, with the name Oliver Heldens appearing at number eight. Still only 24, Heldens is one of a new wave of Dutch EDM stars who is following up the breakthroughs made by van Buuren and Tiesto. Born in Rotterdam in 1995, he's a digital native who is equally at home in pop music or a swathe of EDM subgenres.

"Before I began producing, I played piano and keyboards," Helden recalls, "but when I started producing on computers, I stopped playing eventually. I now just click in the melodies and chords as they pop up in my head. So there's no need to play keyboards anymore. When I hear a classical music composition, I can easily click in the notes and just remake the composition. It's the same when I have a musical idea: it only takes a few minutes to enter the melody and chords with a mouse. But I do think that playing piano in the beginning helped."

The teenage Heldens became very adept at this way of working, and major success came when he signed to the Dutch label Spinnin' Records at the age of 18, on the basis of his song 'Gecko'. Released in late 2013, it made some inroads in Holland and neighbouring countries. Half a year later, a new version called 'Gecko (Overdrive)', featuring



British singer Becky Hill, was far more successful, reaching number one in the UK. It was a meteoric rise for the Dutch teenager, helped by the fact that Spinnin' was the happening label in the newly hot EDM genre.

Heldens recalls how he got swept up: "When I was 11, I got into dance music and house music in particular. I was buying compilations of this music and afterwards I fell in love with jumpstyle and hardstyle. I adored people like Fedde Le Grand, Laidback Luke, Chris Lake, Green Velvet and so on. When I was 12, I discovered electro house and club. I went to high school and they threw really big school parties where they booked people like Hardwell and Chuckie. From that moment onwards I wanted to make music myself. A neighbour friend who was a little older was already producing things using Fruity Loops, and he showed me what he was doing, and we'd dance to his music!

"I downloaded Fruity Loops and started making beats on that as well. Because I had the demo version, I could not save what I was making, so every time I made something I'd export a little bit of a part. I'd be exporting all the different parts, like in bits of seven or 15 seconds of audio, and I'd then pull everything together in Windows Movie Maker. That was my first experience of making beats. In the very beginning, I was only working with samples, but by the time I was 15 I got more serious, and was reading about beat-making and started watching tutorials. I went in-depth and got the full version of FL Studio. That's when I discovered all these virtual instruments, like Sylenth, and Native Instruments Massive and reFX Nexus."

Building A Brand

Three years after getting "serious" Heldens had found a record deal and major success. While he never equalled the chart success of 'Gecko' and 2014's 'Last All Night (Koala)', the young Dutchman has become a major player on the international music scene, his releases under his own name being very successful on various online streaming platforms. In addition, Heldens spread his wings in other directions as well. He started his own Heldeep record label, released songs under the name HI-LO (Oli H backwards), and has remixed tracks by

the likes of Martin Garrix, Coldplay, Calvin Harris, the Chainsmokers, Katy Perry, David Guetta and Chic. Heldens is also a regular performer at the world's biggest EDM festivals, and this year he signed with RCA, marking a new chapter in his musical adventures.

As his music goes global, however, Heldens himself has returned to his origins — specifically, his old room in his parents' house. "Yes, I'm back in Rotterdam, in my old bedroom! We completely rebuilt it to make it sound really good. For me it doesn't matter where I work, because I just work in my laptop, entirely in digital. My monitors are the old Genelec 1030A's, and I have a sub. I also have a four-way monitoring system from Eve, the big ones. The two systems sound very different, with the Eves more open-sounding, while everything sounds more together in the Genelecs. My soundcard is the Prism Sound Lyra, I don't know how many inputs

Oliver Heldens: "When 'Le Freak' turned 40, the record company wanted a big dance remix, and asked me to do it. That was a big honour for me!"

or outputs it has, and I have the Mackie Big Knob that the monitors and soundcard plug into, so I can select the monitors.

"My dad is a big synthesizer and audio freak, and he has things like a Minimoog and a Hammond and a Fender Rhodes. A few years ago I had his Minimoog in my studio, which has a really sick sound. Right now, I'm playing around with a Casio Casiotone 405, which is really fun. But I don't really use hardware synths in my tracks. For me it is not really practical. I prefer to use digital stuff, inside my laptop."

Fruitful Loops

Elaborating about the goings-on inside his laptop, Heldens says, "I use Windows and a Mac right now. But Fruity Loops works better on Windows. I haven't tried the Mac version of Fruity Loops yet, but just got a new PC laptop, an HP, with the best drivers, and four TB of storage space. I see a lot of people around me working on Macs, but I don't think it matters what platform you work on. I still use Fruity Loops. I am very happy with it, though I saw some other people using PreSonus

Studio One, and that appeared to me like the future of music software. I also have it now, and want to switch over to using it, but because I am so used to working in Fruity Loops, and it fits my workflow, I haven't switched to Studio One yet.

"In terms of plug-ins, I like the parametric EQ and Gross Beat in Fruity Loops. Gross Beat is really useful for side-chaining. Nowadays you have all those LFO plug-ins and the Kickstart plug-in by Cableguys and Nicky Romero, and they kind of do the same thing, but I still like to use Gross Beat. For reverbs I love to use Valhalla. I still use the three synths I mentioned above — Sylenth, Massive and Nexus — but at the moment I also love Xfer's Serum, and Native Instruments' Kontakt for more organic sounds. Spectrasonics' Keyscape also has really good sounds, and I like to use Native Instruments' Guitar Rig. I spent a lot of time doing sound design, though

I don't mind using presets. If I find a sound I like, I'm happy to use it.

"In a lot of cases I layer sounds. For example, in a song like 'Gecko', the sound that you are hearing is not a bass sound, but a mid-range sound and

a high sound, which is layered with a bass sound. When I do have bass and sub-bass sounds I tend to use Sylenth, Nexus or Serum, which are my go-to plug-ins for bass sounds. In a number of tracks, I use just 808 kicks, which I will put through Guitar Rig, like a big guitar amp, and this will create a huge bass sound. Sylenth is great for creating clean sub sounds."

Getting Together

Of his writing process, Heldens says, "It varies a lot. I don't really have a particular way of working or one specific thing that I always start with. Sometimes I start with a melody, sometimes with chords, sometimes with a bass sound, sometimes with a rhythm, sometimes a vocal. Anything can be a beginning, and then you work from there, adding the other parts of an arrangement as needed. I tend to create like 140 songs a year, which is a lot!"

Over the last year, Heldens has released four singles: 'Fire In My Soul' (featuring Shungudzo), 'This Groove' (with Lenno), 'Summer Lover' (featuring Devin and Nile Rodgers), and 'Cucumba' (with Moguai), and at the time of writing he was >>



A career highlight for Oliver Heldens was working with Chic legend Nile Rodgers at Abbey Road Studios.

» about to release a track called 'Turn Me On', a collaboration with UK beatmaker Riton featuring singer Vula.

Collaborative writing is the norm these days, and Heldens describes his approach thus: "Both 'This Groove' and 'Fire In My Soul' started during writing sessions with other musicians. For 'This Groove' I was in Sonic Vista Studios in Ibiza with Lenno, a Finnish producer, and the singer, JHart. We wrote 75 percent of the song on the first day, and Lenno and I then created

the production. Lenno uses Logic, so we sent files back and forth, adding stuff and changing things. I wanted to do something more clubby, more dancy, kind of inspired by the Studio Jack disco tracks. So I came up with the bass line melody, and then Lenno added the piano, and we carried on creating the entire track. We then worked with JHart, and he came up with some really fresh vocal hooks. For some reason the melody 'every time I hear this groove' [from the 1984 disco track 'Time To Move' by Carmen] kept sticking in my head, so we downloaded the a cappella and pitched

it down, to make it fit with our track. It really worked.

"'Fire In My Soul' was written in LA, in a session with Warren 'Oak' Felder, who usually does more pop and urban productions, so that was an interesting combination. Oak works with two writer-producers, Trevor Brown and Zaire Koalo [a production and songwriting duo called *The Orphanage*], who were there, and we also had a really cool singer-songwriter in the studio called Alexandra Shungudzo Govere. I have been into African music lately, and Shungudzo is half Zimbabwean, and Oak is half-Turkish, and we started out showing each other all these African gems. This meant that we had a vibe when we started writing. Trevor played some chords, and then Oak and I created a bass line to that. We were all inspiring each other, and it happened really quickly. Shungudzo wrote the vocal hook and melodies in a really short time. The song was written on that day, and I then took it home to work on the production and make it more Heldens."

Freak Out (Again)

One of Heldens' most memorable experiences was working with Nile

»

House Building

There's a lot of snobbery and confusion about the term EDM, which is despised by many, and its hundreds of subgenres. Oliver Heldens is said to make future house, which is related to deep house, which itself is a subgenre of house music. Future house itself can be split in future bounce and future trance, and so on, and on. So what does Heldens make of this?

"I know they call what I do 'future house', but I don't really think about that. To categorise music in genres is very human, it is kind of how our brain is programmed, trying to put things into categories. It helps a lot of people to make it more *overzichtelijk* [a Dutch word meaning

clear, structured or uncluttered]. But I just try to make the best tracks possible. I just try to make music that really excites me. I don't try to put myself in a corner too much. I don't really think that much in genres. In any case, there is so much music nowadays that is a hybrid of several subgenres, and the things that don't fit in one genre or one corner these usually are the best tracks. It's also what I try to do in my sets. I would never do a set with only future house. That would bore me, even though I love future house, of course. But I also like to put in some more techno or more electro stuff, or more disco house."

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Not only a producer, Oliver Heldens is also one of the world's most high-profile DJs.

» Rodgers, one of his all-time heroes. "He has always been in the top of my list of people I want to work with, and two years ago I emailed him and sent him some of my ideas, and there was one song he really liked, and he wrote a song to it and played guitars over it. We didn't finish the track — I'm still working on it — but when 'Le Freak' turned 40, the record company wanted a big dance remix, and asked me to do it. That was a big honour for me!

"They sent me the stems, and I loaded everything in Fruity Loops. In most cases when I do a remix, I just use the vocal, and create an entirely new production, but in this case I wanted to use the guitars and strings as well. I wanted to remain true to the original and not turn it into a completely new track. I created a heavy bass line drop, which goes well with the original guitar and vocals.

"Nile and his team and the label were really happy with my remix, and they invited me to go into Abbey Road Studios in London. I did a few sessions with him there, with several different singers, like Rebecca Ferguson and Craig David. For the first session I had some instrumentals ready, and another session started with

an idea Nile had for a song. He played the idea, and I hummed a cool bass line to it, and when he heard the bass line he started playing something different, and suddenly we had a completely new track with just his guitars and my bass line. I put the bass line in Fruity Loops, and within half an hour we had a completely new song, with really fresh vocals from Craig. Only one of the songs I have done with Nile has so far been released, 'Summer Lover', but we still have several other unreleased songs."

Multitasking

In general, Heldens seems to be branching out to slightly more traditional music industry activities, like writing in commercial studios with others, as opposed to alone in his bedroom, and also signing to RCA and further developing his Heldeep label. "I signed with RCA so they can release the crossover singles," the Dutchman explains. "I put the more club tracks out on Heldeep. 'This Groove' was released on Heldeep, but RCA might take it over. They call it upstreaming. Heldeep has been growing and growing, and last year we did many Heldeep events, like hosting a stage at various festivals. I have two

guys working for me at Heldeep, and we also work together with Spinnin'. I also have a company called Noise House, which produces radio shows. Together we do the Heldeep Radio show every week. My brother is really helpful with that."

With club and pop song releases, remixes, his own label, a radio show and live performances, Heldens has his hands in many projects. It's how he likes it, he says, "I try to do many different things at the same time, though in the spring time I'm often focused on creating specific edits for my DJ sets at Ultra and other concerts. I do this in Fruity Loops. I do a lot of mashups. It's really fun to take a cool underground record and edit it in a way that makes it sound even better, and that works live. When I am on stage I use Denon CD players and a mixer.

"The winter is a time when many DJs are at home, and take time off from touring. During that time, I tend to go more into the studio and create tracks. I've been working on some solo Heldeep club records, but I also created many songs last year that need finishing. I'm focused on singles and tracks for Heldeep, but I may eventually do an album or EP, because I have so much music!" ■■■

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PROFESSIONAL AUDIO HARDWARE AND SOFTWARE

In Session Audio Riff Generation Outside In Edition

Kontakt Instrument

★★★★★



I reviewed In Session Audio's original Riff Generation title in the December 2018 issue of *SOS*. The package managed to combine a number of familiar concepts — a library of predominately synth-based sounds, pattern-based sequencing, creative effects, flexible randomisation options and adaptive key/scale matching — into an elegant user interface to deliver something that was very much greater than the sum of its parts. In particular, the combination of the last two of these features meant the 'generation' bit of the library's title was very apt; this was an instrument that could generate some inspiring, pattern-based, musical ideas, often with a hypnotic quality that would work in all sorts of electronic music styles or soundtrack creation.

In Session Audio have now released Riff Generation Outside In Edition. It is built around the same UI (as covered in the original review) but with a different sound set and including some engine tweaks introduced since the original title first appeared. The 'Outside In' label indicates the origins of the new sounds. All of these are totally new and derived from 'real' acoustic and electronic instruments rather than created from 'in the box' sources. These include vibraphones, xylophones, acoustic and electric guitars, pedal steel, bass, cello, bells, pianos as well as a number of hardware synths. Again, the emphasis is on sounds with a shorter duration and there are many with a percussive nature, allowing you to easily create a rhythmic element within your sequenced patterns.

In terms of the engine itself, In Session Audio have taken advantage of new reverb and delay options built into Kontakt 6, so the effects section now offers you some additional creative choices. However, perhaps the most significant change — and one that I noted on my own wish-list in the original review — is the ability to divide the 256 pattern steps available into eight shorter 'scenes', with key-switching between them on the fly. This instantly makes the instrument easier to use in a compositional context. It is also useful to be able to copy/paste between scenes as you experiment, so you don't lose a good idea from further use of the various generation options.

I have to say I think the new sounds are excellent and, as an alternative to the synth-dominated sounds of the original release, bring a more organic feel to patterns you create.

SAMPLE LIBRARIES

For example, when used together, the various guitar harmonics and palm-muted sounds can create some beautiful, haunting atmospheres within the right pattern. Equally, the sub-set of percussive sounds mean you can add an additional rhythmic element to your sequences. If you liked the original Riff Generation, then you will undoubtedly also like the Outside In Edition. No, it's not cheap, but it sounds fabulous, is undoubtedly inspiring, and is also a lot of fun to use. Well worth exploring. *John Walden*

\$249.99

www.insessionaudio.com

Musical Sampling Boutique Drums Ruby & Jolene Kontakt Instrument

★★★★★



While top-end virtual drum instruments such as Superior Drummer 3 are both highly desirable and amazingly capable, for some users, they are beyond their budgets and quite possibly overkill in terms of features offered. Fortunately, for those looking to get something that's both lower cost and simpler, there are some interesting alternatives. One such option is the Boutique Drums series from Musical Sampling and there are two titles currently available; Ruby and Jolene. Both are modestly priced and offer a super-simple user experience. Both also require Kontakt 5.8.1 or later.

The two libraries share the same very streamlined user interface, but differ in terms of the drum sounds on offer. Ruby is perhaps aimed at conventional rock and heavy rock, while Jolene has a more vibey sound and would perhaps suit indie, pop and (yes, Jolene) country styles. In each case, the UI provides you with eight complete drum mix (the blurb uses the term 'baked-in') presets and, while you can mix-and-match individual kit pieces (kick, snare, hi-hat, etc) between the various presets and adjust their mix balance, that's pretty much it in terms of sound-shaping. This is a deliberate 'keep it simple, stupid' design decision by Musical Sampling; you pick a preset and go.

The sampling itself feels fairly detailed, though, and the drums themselves have a good dynamic response as well as round-robin sampling on the hi-hat, rides and toms. As well as each kit piece being mapped across a conventional range of MIDI notes, you also get three batches (each recorded at a different

tempo) of eighth-note hi-hat patterns; hold a single key and these will play back tempo-matched to your host. The sampling here includes different degrees of hi-hat opening and a very neat option of using the Mod Wheel to crossfade between these in real time. You can switch these patterns between half and double time to create 1/4 or 1/16 grooves. You can also engage a Humanization setting

which simulates

varying degrees of timing adjustment to incoming MIDI data so programmed parts are not too mechanical.

That's about it in terms of features.

In use, the sounds

are very solid and very usable. As advertised, it doesn't take you long to pick a sound and get playing. I'm sure this will suit some users down to the ground. Personally, I would have liked an option for fine-tuning the degree of ambience although, in part, that's what the different mix presets offer. The simplified concept here is perhaps taking things in a similar direction to that offered by the Get Good Drums products and, if metal is more your thing, I suspect they might be a better bet. However, if you consider that even the GGD approach is OTT, then Boutique Drums Ruby and Jolene keep it even simpler while still offering some solid acoustic drum sounds. *John Walden*

Ruby \$99, Jolene \$79.

www.musicalsampling.com

Freshtone

Lost Tapes Vol 3 Acidized WAV & Apple Loops

★★★★★

If you are wondering when Freshtone Lost Tapes Vol 2 was reviewed in *Sound On Sound*, take a look at the July 2014 issue. This follow-up library certainly took

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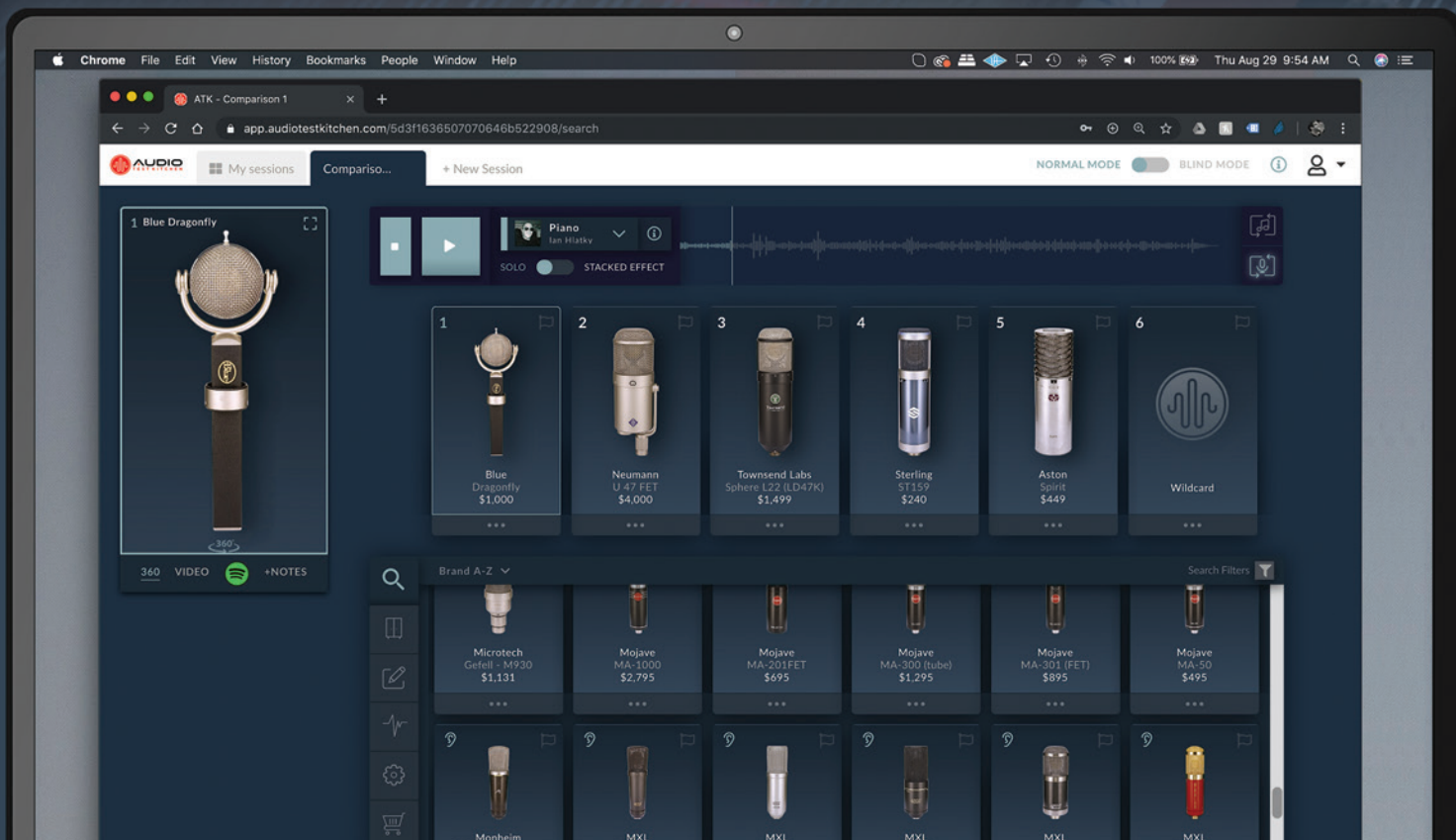
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» a long time to appear, but it continues where its predecessor left off, by providing soul and psychedelic-influenced funk loops that sound like they could have been sampled from records cut in the late '60s or early '70s.



Once again, the sample content was recorded at East London's Gizzard Studios, using the facility's Alice Stancoil mixer, ribbon mics, Studer tape recorders and vintage instrument collection. In fact there are so many similarities between the two volumes that I wouldn't be surprised to learn that this one was recorded at the same time as its predecessor and has been kept on ice until Volume 2 sales started to fade.

In all there are over five gigabytes of samples, but it would be more accurate to say that the library is 2.7GBs, because each recording appears in Acidized WAV and Apple Loops formats.

The first folder contains 89 full band instrumental recordings, mostly lasting about 15 seconds. These ensemble pieces include drums, percussion, bass, guitars, electric pianos and brass sections, so they are a bit full-on to use as the basis of custom compositions, but they do demonstrate how the rest of the collection's samples fit together. There's also a folder called Cutdown Jams, which includes all the sections apart from the brass.

The remaining nine folders contain the loops of the specific instruments, so if, for example, you like the guitars in the arrangement called '119bpm MADISON full jam 1.WAV', its six guitar parts can be found in the Guitar Loops folder, all starting with the name '119bpm MADISON'. If, on the other hand, it is the Rhodes lick that you have a use for, the Keys Loops folder is the place to look.

Although the solo instrument loops generally have plenty of atmospheric amp hum and harmonic distortion, they don't exhibit any spill from the other instruments, which tends to suggest that they weren't recorded by a band

grooving together in the same space.

Overall, though, the performances are excellent and the arrangements are spot on. It's all there, from busy drums, to restless basses and biting brass stabs. The producers must have spent most of their lives either watching '70s American TV cop shows or trawling through countless old vinyl collections. It would have been great to

have some arrangements that feature strings for that even bigger orchestral sound, but considering the collection's very reasonable retail price, that's probably asking too much.

My only real criticism is that sometimes the timing seems slightly too tight, possibly as a result of the recording and editing processes used to get clean and tempo-accurate samples, but that's a small price to pay to avoid the sort of copyright infringement issues that come with vinyl record sampling. *Tom Flint*

£36

www.freshstonesamples.com

Spitfire Audio Hauschka Composer Toolkit Kontakt Instrument

★★★★

Recorded in Berlin's all-analogue Vox-Ton Studios, this library is based on a collection of prepared piano multisamples created by film composer and sound designer Volker Bertelmann under his stage name Hauschka. The piano tuner might have been horrified to witness Hauschka stuffing wedges, ping-pong balls, gaffer tape, saxophone reeds, toy drums, a tambourine, necklace and (the *piece de resistance*) a Christmas cake wrapper into his precious Steinway D-274 grand, but the resulting sound set was worth it: all the resonant, jangly, percussive and quasi-industrial timbres one associates with prepared piano are here, and a lot more besides.

Where this toolkit earns its keep is the imaginative processing applied to the raw recordings, which take the piano out of this world. In addition to two clean mic channels are four processed signals: distortion, Hauschka's flangy, distant-sounding pedalboard effects, a 1960s Binson Echorec tape delay and weird pitch-modulated treatments generated by Hauschka collaborator Francesco Donadello's modular synths, the latter a good bet for your horror film scores. A nice hall reverb, delay effects and frighteningly loud tape saturation are also available.

The library employs multiple GUIs for its presets: some feature Spitfire's beloved pegboard-style grid, others run in the sound-designer-friendly Mercury synth (a great source of LFO-driven psychedelic wobbles), and all can be loaded into a conventional Kontakt controls

panel. The library runs on Kontakt or Kontakt Player 5.6.8 or higher, consists of nearly 16,000 samples and requires 25GB of disk space to install.

'Plug Hits Grid' contains a mixture of jangly prepared-piano timbres and short, muted clean notes reminiscent of pizzicato strings and palm-muted electric guitar: add a 16th-note delay with a little feedback, and you're instantly transported to early 1960s guitar instrumental territory. One ear-catching preparation was created by placing small metal tea lights on the piano strings, producing a gentle, bouncing rattle which is very effective on high-pitched chords. A straight piano multisample is also included — though not exactly delicate, its strong, muscular sound would fit well with grandiose orchestral arrangements.

More atmospheric and floaty effects can be found in the Swells Grid, a set of fast repeated single notes and octaves with a built-in crescendo-diminuendo; use only the modular signal with lashings

of reverb for these performances, and the piano is transformed into heavenly thrumming harps with a hint of celestial choir. Other highlights include dreamy reversed samples, glitch hits and a 'drum kit' of atonal percussive noises featuring some great thumping bass notes and snare-like metallic

thwacks which you can sequence into kickass industrial grooves.

The icing on the cake is a set of 50 'artist presets' created by the Spitfire team, which add ethereal, eerie and deranged sci-fi effects, tambura-like bass drones and some captivating electronic soundscapes. All in all, an interesting collection which will inspire creative composers with left-field leanings.

Dave Stewart

£249

www.spitfireaudio.com



Audio examples of this month's libraries are available at www.soundonsound.com.

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THE MIX REVIEW

COMMERCIAL PRODUCTIONS ANALYSED

We examine the production of some recent hits to help you brush up on your listening skills.



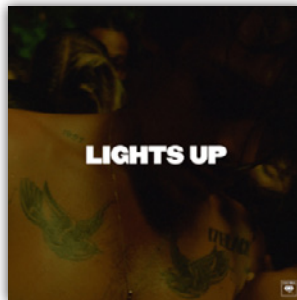
DOMINIC FIKE
'3 NIGHTS'

Two things really stand out from this production. The first is the vocal performance, which covers a tremendous expressive range. It starts off with the chilled attitude that permeates the layered opening chorus texture, as well as forming the baseline character of verse 1. After eight bars, though, it assumes a more playful mood, with nice little articulations such as the skipping “company” at 0:50 and the leap up to “like” at 0:57. Then after bar 16 he shifts up an octave into a cool raspy chest voice, but interspersed with lovely moments of layered falsetto. That high-register lead sound picks things up again in verse 2, but with a noticeably more assertive delivery, leading into my favourite moment of all: the gloriously ragged “too fucked up” at 1:53. But even though this seems like the performance high-point, he’s still got one more trick up his sleeve at 2:06, a brilliant bit of patter singing which sashays between straight and triplet rhythms so fluidly that you can scarcely see the joins — one of the most assured bits of creative groove-surfing I’ve heard in a while.

My second production highlight is more subtle, and best heard during the introduction. Notice how the backbeats seem to be accompanied by a burst of filtered noise. I suspect this is simply part of the claps sample they’re using, but there’s always the possibility that it’s been overlaid separately. Either way, this noise burst ends pretty much in tandem with its subsequent kick drum, and I think this helps make the kick stand out more. It also creates a kind of musical connection between the clap and kick, so that the programming feels less like a bunch of isolated events, and more like a kind of musical relay where each event is driving us forward to the next. This is something that I’ve always associated with people like Portishead and the Prodigy (I’m showing my age...), but it’s really easy to transport the idea into pretty much any style, as long as there’s some long-tail drums component that can be cut off by another — perhaps an open hat being stopped by a closed hi-hat, a sustained 808 kick by a snare, or a reversed cymbal by a kick. *Mike Senior*

HARRY STYLES
'LIGHTS UP'

The layered vocal texture used for most of this song is unusual on two counts. Firstly, it’s not common to hear a vocal line being doubled at both the upper and lower octaves, an arrangement that adds timbral variation without actually supporting the harmonies any further. And, secondly, the doubling lines sound so tight and

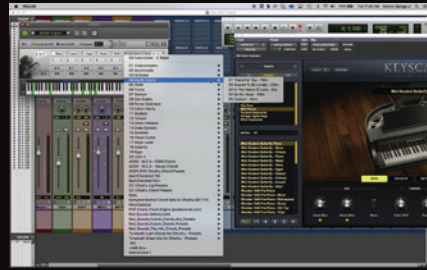


electronic that I suspect they’ve been derived directly from the lead line using pitch-shifting. I have no inherent beef with this, though, because even though it does make the combined vocal texture sound more like

an artificial effect (rather than a group of natural vocalists), I really don’t think that’s a concern within the context of this production’s prevailing hazy, tripped-out vibe. Speaking of which, I love how the (presumably) intentional pitch-drift of the electric guitar part first heard at 0:28 contrasts with the super-tight vocal tuning to enhance the production’s kind of ‘seeing double’ wooziness, almost as if the vocals and guitar are shifting in and out of focus with each other.

Another odd aspect of this song is its structure. Ask yourself this question: what’s the main chorus section? On first listen, I’d have said it was the section starting at 1:18. It’s uptempo and high-energy. It begins on and candences back to the key’s home chord of Bbm. It has a repeating lead-vocal melody and lyric, along with that feel-good “Shine” backing vocal refrain. But, er, that section only appears once in the whole song! So is it the drop-down sections, which contain the “lights up” lyric, and the second of which closes out the song? But, um, they sound more like breakdown pre-choruses than choruses. Or maybe it’s the opening texture when the beat first arrives at 0:19 — after all, that’s the section that seems to get the most track >>





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Illmind: Producing A Track From Scratch In One Hour

New York-based producer/engineer, songwriter and music educator Illmind has wide-ranging credits extending all the way from 50 Cent, Drake, J Cole and Kanye to the smash soundtrack for Disney's *Moana*. In this month's exclusive video feature, we watch him build a catchy track in his home studio in just one hour, from nothing but a single dry vocal sample.

www.youtube.com/soundonsoundvideo

» time overall, and there are repeated lyrics there too. But, er, that really feels more like a verse. Taking a step back, though, does it actually matter? So what if we don't know quite where we are in the song, or where it's going next? Doesn't that just reflect the freewheeling stoner aesthetic of the musical statement as a whole?

That aside, the rather bland 17-second instrumental intro might also raise some eyebrows. Wasn't it supposed to be a basic rule of chart pop that you get to the main hook(s) and/or lead vocal as quickly as possible? Well, I wonder whether we might be witnessing a tactic to increase streaming statistics here. For instance, Spotify counts a 'play' when the listener makes it more than 30 seconds into a track, as I understand it. Working on the assumption that most people will want to hear some of the actual song before deciding whether to skip, I reckon simple curiosity will likely carry them through that intro — much as curiosity carries most of us through those 15-second pre-video ads while surfing YouTube. At that point there's only time for five bars of the song before the 30-second mark, and I reckon few listeners would feel confident skipping any song that quickly. Had the track started straight in with the beat, on the other hand, the 30-second mark would have arrived during the 13th bar. Given that that's the beginning of the fourth four-bar iteration of fairly similar musical material, I reckon there'd have been a significantly higher likelihood of skippage before then. *Mike Senior*

JOEL CORRY

'SORRY (I DIDN'T KNOW)'

If you've not heard Monsta Boy's original UK garage release of this song from 1999, then be warned that it's a slightly unsettling listen because, frankly, its *bona fide* retro production feels a whole lot cooler and more contemporary than Corry's currently charting retro-flavoured happy house cover. Certainly, if you'd asked me blind which of the two came out of a '90s bargain bin, I'd probably have chosen Corry's on cheese-factor alone.

Leaving my musical snobbery to one side, there's a simple harmonic trick on display here that's worth shining a neon glow stick on. In a lot



CLASSIC MIX

OASIS

'Wonderwall' (1995)

Despite Robson & Jerome denying it the UK top spot, this song went on to become Oasis's best-selling single in the UK, and it's not hard to find reasons. Liam's instantly recognisable vocal tone, of course, must rank high on the list, balancing super-focused midrange, cutting nasality, and (on higher-register lines such as "I don't believe that anybody" at 0:33) a hint of tastefully grainy break-up.

However, I really rate the drums here too. The choice of brushes is a great production move, as this instinctively feels like it better matches the arrangement's prominent acoustic guitar and solo cello. Delaying the drummer's second-verse entry by three beats is also inspired, not only because it's so unexpected, but because it prevents the drums obscuring the cello's first entry and also adds stress to Liam's vocal rhythms on "street that the" — wisely resisting the temptation to plant a snare under the word "backbeat", which I don't think would have been nearly as attention-grabbing. And that fill after the first chorus is also a classic!

The harmonies are impressive as well, because of the way Noel manages to puzzle out a satisfying progression using five different chords (A, B7sus4, D9, E5sus4 and F#m7) which all share two notes (A and E). This means that those notes can be sustained as upper pedal tones pretty much throughout — no wonder U2's guitarist Edge has apparently said he wishes he'd written 'Wonderwall', because those pedal tones would have been ideally suited to his trademark long-tail echo effects.

What mystifies me a little with this song,

of house tracks, the harmony involves a series of seventh and ninth chords, but with the traditional functionality of the chords weakened in two ways.

Firstly, any major chords are given major sevenths to counteract their tendency

towards a dominant function; and, secondly, the chords tend to track the bass line in parallel motion, which undermines any sense that the dissonances are obeying traditional principles of voice-leading. The fundamental four-chord progression around

though, are the disparities between the original album and the version on the band's

Stop The Clocks (2006) and *Time Flies* (2010) greatest hits collections. Now I can understand why the drum fill after the first chorus appears to have been drastically EQ'd, because the kit was clearly mixed to leave plenty of room for the harmonic and melodic parts, which means it does sound rather thin in isolation at that moment. But why has the entire mix apparently been polarity-inverted for the compilation? I accept that some people feel this makes a sonic difference, but if the improvement were so cut and dried, then it's curious that the very same mastering engineer didn't repeat the polarity-inversion move for his 2014 album remaster.

Then there's the fact that the left and right channels appear to have been swapped for the compilation. Compare the positioning of the clean guitar counterpoint under "all the lights that lead us there are blinding" (1:11), for instance, or the electric guitar's sustain at the end of chorus one. But what's weirder is that this doesn't remain consistent, since the verse sections don't seem to have been channel-swapped — the opening acoustic guitar is noticeably brighter and pickier on the left side in both cases, for instance. What could possibly have been gained by doing this? It makes no sense to me at all. Yet if it's a simple goof, then isn't it actually the kind of thing you'd expect a top-flight mastering house to be quality-controlling for, especially with their highest-profile artists? *Mike Senior*



which Corry's whole track is constructed (E7-G#m7-C#m9-D#m7) unequivocally ticks both those boxes, so no surprises there, but the pace of the chords doesn't stay the same throughout the timeline. So the opening double prechorus features a simple one-chord-per-bar harmonic rhythm, but when the following double chorus arrives that rhythm accelerates to two chords per bar, as well as adding an element of syncopation by pushing chord three an eighth-note early and by pulling chord four an eighth-note late. It's hardly rocket science, but it does very effectively increase the sense of urgency and energy exactly where you'd hope.

Mike Senior ■■■



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SOS WHY I LOVE... COLLECTING GEAR

SIMON HORROCKS

Most people's early morning routine consists of stretches and health shakes, but these days mine is more about fine-tuning my email alerts, scanning social media for new pedal releases, and checking that Roland's marketing team haven't dedicated another calendar date to their vintage kit. (OK, so 808 and 909 day, fine, but what about 202, 303, or even 106?)

My name is Simon and I have Gear Acquisition Syndrome.

It's a tough confession to make, but regardless of cost, brand or expediency, if an object transmits a noise, features shiny potentiometers and a curiously busy back panel, I'll be adding it to my growing studio arsenal, and I usually get the tremors if I don't.

Today is no exception. Firstly, and most importantly, my SOS subscription copy is due — chock full of tempting reviews and features to get the circuits flowing. Furthermore, a new audio interface awaits its

unboxing in my man-cave... and joyously, an original, pristine Speak & Spell will arrive by courier and will be forwarded, almost immediately, to my bespoke circuit bender.

I have also finally managed to convince my wife (she from whom I hide brown packages) that high-end, rare sound equipment makes sense as a serious financial investment, and armed with our — sorry, her — inheritance windfall, I now have high mandate to research such a topic.

The magpie instinct in me has been forever present. Back in the day it was vinyl that would get me high; the sniff of a record shop within 20 minutes of arriving in a new city. Charity shops, jumble sales and car boots moved me on to harder stuff. As I gravitated to music making, there'd be regular trips to London's Tin Pan Alley. A single tear is now forming as I recall the Turnkey shop, with its downstairs DJ area and the infamous synthesizer museum. I could continue this paragraph with my remorse for sadly departed high-street

electrical stores and music marts where I would regularly pester a salesperson, not to mention the fear of touching anything in the posh piano shop, and the grumpy bloke who sold second-hand guitar pedals in the indoor market.

So now I purvey the landscape digitally. The social media groups and marketplaces, the endless forum posts and Wiki searches to iron out ridiculously convoluted compatibility issues. Jim Heywood genuinely deserves a knighthood for his 'therapeutic' Synth Memes group, where GAS-heads like me can gather and feel slightly more normal about the way gear acquires us.

And so, as my gadget mountain grows, the wish list now extends to more storage racks to house it all. The studio itself is in need of a rethink. Even the loft space is running out with all those empty boxes, kept for the non-existent resell.

You'll be pleased to know I have considered income streams to compensate these treacherous outgoings. Writing

online reviews of all the guff I have bought is the obvious one. I've even browsed the Chinese wholesale sites to see if I could start up my own musicians' enterprise flogging kazoos or modular units, seeing as all this shopping has turned me into an expert.

The real money spinner, however, is to actually make music itself. It's a remarkable revelation but I could conceivably use all this rottenly overpriced gear I have spent years investing in.

Alas, the most regrettable part of this story is that I am not making as much music as I should. Please don't let me be the only person this is happening to, but each time I sit down to create a momentous new chord, patch, or lyric, my phone pings with news of an item I'm bidding for, or a review from my favourite vlogger.

So how on earth can I rid myself of these horrific consumer-driven addictions? On reflection, perhaps waking up with a smoothie and some yoga would be the killer cure for my repetitive shopping injuries. ■■■

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