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WANTS VS NEEDS

Over 10 years ago I started a leader column by saying: "The cold reality is that the musical hardware and software available five years ago almost certainly already did everything that a talented engineer/producer actually needed to create great recordings." OK, a few more genuinely useful polishing tools have become available over the past decade, but in reality, what developments in technology since then have actually improved the quality of your final projects? As I pointed out in a leader column written some years before even that one, the real divide between pro recordings and home projects comes down to musical ideas, playing skills and the space used to make the recordings — the technology needed to record them was perfectly adequate even back when everything was 16-bit. As long as you have decent monitors or headphones so that you can hear what you are doing, you can make perfectly good release-quality recordings using budget audio interfaces and microphones and a 'lite' DAW. In most cases you can also get by with the plug-ins and sample sets that already come with your DAW, though I have to confess that I do occasionally give in to the temptation to buy third-party plug-ins and sample libraries when I find something that can do a job that my basic plug-ins can't. What I do find frustrating, though, is when one piece of CPU intensive software causes my computer to wimp out, then when I look at the activity monitor, I find one core of my machine is maxed out and the other seven are doing almost nothing.

So, what do I actually want from technology? Apart from a cure to the above load sharing issue, which would be a great start, my wish list starts with a computer that has enough power to run my music software with capacity to spare; I'd like it to be upgradable without having to ditch the whole thing to buy a new model every three or four years; I'd like the software to be perfectly stable and I'd like upgrades in the operating system not to kill off some of the old software upon which I still rely. I'd also like to see new machines with enough of the peripheral ports that I need to plug in my various drives, controllers and monitors. But then I'd also like to be able to cure all health problems, be around to see world peace, live under a government that has the best interests of its citizens at heart and have the major online music distribution/streaming companies pay musicians a fair cut — so it looks like I'm destined to be disappointed for a while longer yet.

Still, it's not all bad news, as some things have definitely improved over the past decade — my six-year-old Mac Pro still just about manages to get the job done, and the spinning beachball of death doesn't visit quite as often as it once did. So maybe when you come to read a *Sound On Sound* leader in another 10 years time, probably delivered by laser retina projection sunglasses, everything will work perfectly all the time. Place your bets please.

Paul White
Editor In Chief

"The real divide between pro recordings and home projects comes down to musical ideas, playing skills and the space used to make the recordings."

SOUND ON SOUND

Sound On Sound North America,
1083 Vine Street #727
Healdsburg, CA 95448 USA

Sound On Sound Ltd (Head Office)
Allia Business Centre,
King's Hedges Road
Cambridge, CB4 2HY, UK

T **+44 (0)1223 851658.**
sos@soundonsound.com
www.soundonsound.com

http://soundcloud.com/soundonsound/sets/sos-podcasts

Editorial
sos.feedback@soundonsound.com
Editorial Director Dave Lockwood
Editor In Chief Paul White
Technical Editor Hugh Robjohns
Features Editor Sam Inglis
Reviews Editor David Glasper
Reviews Editor Matt Houghton
Reviews Editor Chris Korff
Production Editor Nell Glasper
News Editor Matt Bell

Subscriptions
subscribe@soundonsound.com
www.soundonsound.com/subs
Circulation Manager Luci Harper
Administrator Nathalie Balzano

Administration
admin@soundonsound.com
Managing Director/Chairman Ian Gilby
Editorial Director Dave Lockwood
Marketing Director Paul Gilby
Finance Manager Keith Werthmann

Production
graphics@soundonsound.com
Production Manager Michael Groves
Designers George Nicholson Hart, Alan Edwards, Andy Baldwin
Classified Production Michael Groves

Marketing
marketing@soundonsound.com
Business Development Manager
Nick Humbert

Online
support@soundonsound.com
Digital Media Director Paul Gilby
Design Andy Baldwin
Web Editor Adam Bull
www.soundonsound.com
http://twitter.com/soundonsoundmag
http://facebook.com/soundonsoundmag

Advertising
paul.dacruz@soundonsound.com
Tel: 707-569-6021
Advertising Manager Paul DaCruz
Regional Sales Associate David Carson

Distribution
distribution@soundonsound.com
Worldwide
Pineapple Media Ltd
Robinwood, 3 Nightingale Way
Denham, Buckinghamshire
UB9 5JH, United Kingdom
Tel: +44 (0) 1895 831942
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Hear at last: Ocean Way's RM1-B

Some products are announced immediately before shipping, others have a rather longer gestation period — like the aesthetically stunning stainless-steel RM1-B large-format ribbon mic from Ocean Way Audio. It might seem familiar; it has appeared in the *SOS* news pages before, following an outing at AES Los Angeles in 2016, and was even filmed at that show by the *SOS* video crew (the original news item and video can be seen at https://sosm.ag/RM1-BNews16). However, the prototype mic exhibited there was never widely released as seen; original designers Cliff Hendricksen and Allen Sides subsequently embarked on a long programme of design improvements, refining the interaction between the magnet and the ribbon, and making the mic sturdier to prevent damage when shipping.

Mechanical improvements have also been made to render the mic less prone to falling when stand-mounted. This was a concern due to the sheer weight of the RM1-B's neodymium assembly, which generates what the designers call "the greatest magnetic force ever available in a ribbon mic". The "high-intensity neodymium-powered magnet super circuit" contributes to the mic's relatively high output level (comparable to that of premium condenser mics, according to Ocean Way), as well as its enhanced sensitivity and frequency response (allegedly 20Hz to 25kHz, and without the high-frequency roll-off typically associated with ribbon mics). Thanks to a discrete-component, custom-built phantom-powered mic preamp included with the mic, low-noise performance is also promised.

Of course, bespoke design like this is rarely associated with budget equipment; the RM1-B ships for a suggested retail price of $3250 in the USA. But with the mic finally ready for commercial release internationally, we will be getting our hands on one for review very soon — so we can put Ocean Way's claims to the test at last!

www.oceanwayaudio.com

Spitfire on a roll with new orchestral releases

Spitfire, the widely respected sample library/virtual instrument producers, have had a busy few months, even launching their own record label, SA Recordings, in January dedicated to the output of their in-house composers and other emerging artists (see www.sarecordings.com). Back in the world of virtual instrument libraries, Spitfire followed the launch of their Studio Strings library last Summer (see https://sosm.ag/SpitfireStringsNews) with similar Brass and Woodwind collections in December and January. Now the content of all three libraries can be obtained in a combined library/instrument, Spitfire Studio Orchestra.

As with the earlier Studio Strings, all of the recordings for Spitfire's new Brass and Woodwind libraries, and therefore those of the new combined instrument, were carried out in the dry acoustic surroundings of AIR Lyndhurst's Studio One, providing a contrast with the earlier, more reverberant Symphony Orchestra libraries. All of the new instruments feature Spitfire's usual attention to sonic detail, with a profusion of instrumental articulations and various available mic placements, and all corners of the orchestral family represented, from piccolo down to contrabass tuba. Even less common instruments such as the double (or contra-) bassoon and the cimbasso are included. Spitfire Studio Orchestra is available in Core and Pro versions; the Pro version includes many more articulations and several selectable mic placements, whereas the Core version offers only a single 'Tree' mic setup. It costs $999 for the Pro version and $549 for the Core version.

As well as covering traditional sounds with the new Studio Brass, Woodwinds and combined Orchestra libraries, Spitfire also recently released an orchestral collection for composers interested in more off-the-wall sounds. London Contemporary Orchestra Textures, as the name suggests, was recorded with the help of the famous LCO and used as its recording venue a decommissioned US aircraft hangar in rural England (pictured), formerly used for testing fighter jet engines (and therefore heavily soundproofed). Focusing on extremely unusual combinations of instruments that make full use of the venue's natural 10-second reverb, the collection marries these recordings, made with various user-selectable mic placements, to Spitfire's innovative Evo Grid playback interface, a virtual pin matrix for layering and playback pitch selection which allows the layering and composition of some unique sonic textures. LCO Textures retails for $299.
www.spitfireaudio.com

UAD adds three new plug-ins with v9.8 upgrade

As is now standard practice for Universal Audio, the latest update to the UAD software that ships with the company's DSP-powered hardware interfaces, UAD cards and DSP accelerators includes 14-day demo versions of various new plug-ins; users who perform the upgrade can pay to keep them at the end of the trial period. Three come with the UAD v9.8 upgrade:

- V76 Preamplifier (below) is an emulation of the valve-based mic preamp designed in the 1950s by Germany's IFR, or *Institut für Rundfunktechnik* (the country's still extant Broadcast Technology Institute). The original device was widely used by studios all over Germany, as well as Decca and Abbey Road studios in London. The virtual version models the entire valve- and transformer-based signal path (including the impedance, gain staging and circuit behaviour of the original), offering up to 76dB of gain with the classic V76 sound.
- Antares Auto-Tune Realtime Advanced doesn't need much explanation; as the name suggests, it's the latest version of Antares' classic plug-in, usable in real time with low latency.
- Diezel Herbert Amplifier (below right), modelled by Brainworx, emulates Peter Diezel's much-loved 180W Herbert guitar amp, originally launched in the early 2000s. The plug-in offers users 120 modelled recording paths with a selection of speaker cabinets and microphones, all recorded in Brainworx's studio through a Neve VXS72 console.

The UAD v9.8 update is available from www.uaudio.com/uad/downloads. At the end of their 14-day trials, UAD v9.8 users may purchase the V76 plug-in for $149, the Auto-Tune plug-in for $299, and the Diezel Herbert amp for $149. Owners of the existing Antares Auto-Tune Realtime plug-in can upgrade for $49.
www.uaudio.com

Roger Mayer updates classic mastering limiter

Roger Mayer is the electronics engineer responsible for the Octavia pedal used on the 'Purple Haze' guitar solo, the RM58 mastering compressor/limiter and the more recent 456 analogue tape simulator (reviewed *SOS* November 2014: https://sosm.ag/RM456). He has now announced a new two-channel version of the RM58 which incorporates the 456 tape emulator.

The revised RM58 is a Class-A, dual-channel, FET-based feed-forward limiter that offers dynamic gain reduction; in other words, there is no user ratio control, just 21-position stepped threshold and output controls, and the limiting ratio constantly changes based on the audio being received at the input. The designer claims this produces "a very natural, musical-sounding result" with a stable stereo image even when high gain reduction is being applied. Built mainly from stainless steel with a universal power supply and a six-layer, triple-shielded PCB for low-noise performance, the hand-built new RM58 retails for $3500 in the US.
www.roger-mayer.co.uk

K&M hit three score and ten

König & Meyer, the German microphone stand and accessories manufacturers, turn 70 this year — a perfect example of a company that produce such reliable, high-quality products for so long that they have literally become part of the furniture where studio owners and users are concerned. Still a family concern — Gabriela König, granddaughter of one of the founders, is now Managing Director — the company continue to innovate, producing 95 percent of their designs and completed products to stringently certificated eco-friendly standards at the company headquarters in Wertheim, south of Frankfurt.

At NAMM recently, K&M introduced a new series of lockable, hard-wearing $470 loudspeaker/lighting stands in discreet white for events such as weddings, and the award-winning Omega E (pictured), an $1699 height-adjustable keyboard stand which accommodates loads of up to 80kg and can be adjusted via Bluetooth with a smartphone app.
www.k-m.de

Antelope power on with Edge Go

Antelope's Edge Go, which had just started shipping at the time of writing, is the latest addition to the company's range of modelling mics, and surprised us all by being a USB-powered mic — a first for a modelling mic as far as we are aware. A dual-capsule, large-diaphragm, variable-pattern condenser mic like the earlier Edge Duo, the Edge Go has all the built-in vintage dynamics processing, reverb, and classic modelled condenser emulations offered by the Duo and handles all A-D conversion itself, at up to 24-bit, 192kHz.

Antelope's intention was that adding an Edge Go to a laptop-based recording setup should ensure you have everything you need to make high-quality vocal recordings; to this end, the mic receives power and transmits its audio output via its USB-C connector. Latency-free monitoring of the output is also possible via a 3.5mm output jack. Users have full control of the mic's features from their desktops via a PC/Mac-based app, and the mic ships with various processing presets, making setup quick and easy for beginners who need a sound for specific applications — for example podcasting, radio broadcast, or voiceovers for YouTube or live gaming streams.

The Edge Go is supplied with a stand, pop filter, shockmount, USB-C cable and hard carrying case, and retails for $1595.
https://en.antelopeaudio.com


DP
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MOTU

FX50 & FX80: new nearfield actives from Fluid Audio

Fluid Audio's first FX-series coaxial monitors, the affordable FX8s (which, logically, featured eight-inch LF drivers), were reviewed in *SOS* February 2015, winning praise from Paul White for their detailed sound and imaging. The company have now returned to the coaxial concept, updating the FX8 design to create the FX80, and has also introduced a smaller design with a five-inch woofer, the FX50. As with the FX8, an immobile waveguide is used for the concentric treble driver; Fluid claim this prevents the high- and mid-frequency smearing that occurs in concentric designs that use the (moving) cone of the woofer as a treble waveguide.

The new speakers also feature Class-D amplification (100W for the FX50, 140W for the FX80; the FX8 had a built-in Class-A/B design) with mid and treble trim and adjustable low-frequency roll-off controls for room tuning and optimum subwoofer integration (the FX8 lacked these controls). There's a DSP-controlled crossover for a smooth handover from the LF to HF driver, and a composite woofer cone is employed (the FX8 used paper cones). The FX50 uses a slightly smaller one-inch silk-dome tweeter than the FX80, which features the 1.2-inch design from the FX8.

Best of all, like the original FX8s, the FX50 and FX80 remain keenly priced, retailing respectively for $299.98 and $498 per pair.

www.fluidaudio.com

Warm Audio launch new valve & FET condenser mics

Warm Audio have gained a reputation for recreating various much-loved classic pieces of studio recording kit for a fraction of the price the originals now command. Now they've announced two new microphones, the WA-251 and the WA-84.

The WA-251 is modelled on the famous Telefunken ELA M 251E (to give the original mic its full name), and as such is a large-diaphragm, three-pattern (omni, figure-of-eight and cardioid) valve-based condenser mic, designed for vocals but usable on just about anything that produces noise in a studio or live context. The WA-251 features Warm's own brass AKG CK12-style WA-12-B-60V edge-terminated capsule, a Slovakian JJ 12AY7 valve, an American CineMag transformer, and a custom Gotham Cabling seven-pin power cord.

The WA-84 similarly takes its lead from the classic (and long-discontinued) Neumann KM84; like its forebear, it's a small-diaphragm, FET-based condenser mic suitable for use on instruments of all kinds, live or in the studio. The new mic features a removable cardioid capsule designed to what Warm Audio claim are "vintage specifications"; no other capsules are currently available, but clearly Warm Audio have designed the mic with future replacements in mind. The mic also includes a heavy nickel large-core CineMag USA transformer, high-quality polystyrene, tantalum and WIMA capacitors and a Fairchild FET. It's available in black or nickel finishes, and should be shipping by the time you read this.

The WA-251 retails for $799 in the US and ships with its own power supply, shockmount and wooden box; the WA-84 is $599 and comes with a carrying case, a shockmount, windshield and mic clip.

www.warmaudio.com

Japanese sound designers feel the width with MS EQ Comp

Although theirs is most likely a new name to *SOS* readers, you've probably heard the work of Osaka-based sound and software designers The Internet Company before; over their 30-year history, the company's behind-the-scenes work has been incorporated into products by Roland, Yamaha and Sonnox, as well as various Vocaloid libraries. Now marketing their own Mac and PC software products, such as the Sound It! software audio editor and F-REX real-time demixing app, the company have just released MS EQ Comp, a VST stereo/M-S encoder/decoder plug-in with built-in control of level, EQ, and dynamics. Suggested applications include selective mix processing and stereo width enhancement/adjustment. Costing a mere $29, it's available now.

https://internetmusicsoft.com

NEWS

Softube, virtual instrument specialists and makers of the Modular virtual Eurorack environment, have released two new products: a plug-in version of the esoteric (some say impossible to use) Valley People Dyna-mite processor from the early 1980s, and a new Mutable Instruments virtual Eurorack instrument, Rings, which is only available for Modular. The latter is a 'multi-resonator' plug-in which allows users to create virtual resonant plates, strings, membranes and tubes, define modes and points of vibration, and then excite the models with noise-based impulses or external audio sources. Rings can apparently produce sounds ranging from drones to percussion and plucked and bowed string instruments. It costs $39. Their version of the Dyna-mite, meanwhile — available in VST, AU, AAX and Console 1 formats for $149 — provides a software version of the vintage device in all of its inscrutable glory, but also provides two simplified versions, Dyna-mite Slam (with the compressor/limiter functions) and Dyna-mite Gate (for gating and expansion).
www.softube.com

Synth designer Tatsuya Takahashi, who was interviewed in February's *SOS*, has written in to correct the attribution of a quote about circuit design which he mentioned as a personal source of inspiration in his conversation with article author Kim Bjørn. The quote, in the third paragraph of the article, was correctly attributed to a Moog engineer, but the name was incorrect in the printed version of the article. The Moog employee in question was actually Chief Engineer Cyril Lance.
https://sosm.ag/TatsuyaTSOS

There have been tone-shaping guitar pedals designed to make electric guitars and basses sound like synths before... but few have been quite so playable and fun as the new Mono Synth and Bass Mono Synth pedals from Electro-Harmonix, which caught our ears (and eyes) at NAMM in January, offering guitar users instant access to monophonic string synth, bass synth and searing lead sounds. With just five main rotary controls, the pedals are easy to use: 'Dry' and 'Synth' set the level of your unprocessed guitar and generated synth sound respectively, 'Sens' adjusts how responsive the synth sound is to your playing dynamics, and 'Ctrl' allows you to adjust a single key parameter for each of the 11 preset synth-type sounds accessed via the final knob, 'Type'. Favourite sounds can be stored as user presets with a tap of your foot, and external inputs also allow preset selection and adjustment of a further parameter per sound via connected pedals. Both new stompboxes cost $164.70 each.
www.ehx.com

Given that headphones are effectively microphones in reverse, it seems odd that mic legends Neumann had never manufactured any... until January, when they announced the NDH 20. According to Neumann, the new closed-back, foldable design is aimed at studio-based and FOH users who like to track and mix on headphones; the designers therefore strove for high acoustic isolation, portability and a neutral, uncoloured tone suited to reference monitoring and mixing, not unlike that exhibited by the company's KH studio monitors. With an impressively flat frequency response allegedly extending from 5Hz to 30kHz, the NDH 20 uses new 38mm drivers with neodymium magnets for maximum sensitivity and low distortion, connected by an adjustable headband made of flexible steel. Supplied with a soft cloth bag and two detachable cables (one coiled, one straight), the NDH 20 retails for $499.
www.neumann.com

TH-U, the latest version of Overloud's TH guitar amp simulator and processing plug-in, was announced at NAMM in January and has now been released. TH-U adds 15 new amps and 14 new cabinets, making for a total of 89 modelled guitar amps and four bass amps, and 50 guitar and two bass cab types, together with 77 pedal and rack effects. There's a new delay and reverb known as Shimmer, and the built-in looper now has true multitracking abilities. The cabinet impulse response loader has also been revamped and improved, and the amp models have been made customisable with the addition of the Amp Tweaks window, which allows users to exchange virtual valves in the amp models. Entire guitar rigs comprising amps, cabs, mics in various positions and room acoustics can now be captured via the new Rig To Model feature, and complete rig models can be loaded via the new Rig Player. TH-U costs $299.
https://overloud.com

Moogfest, the annual electronic music festival founded to honour sonic pioneer Robert Moog, returns to Durham, North Carolina April 25-28. At the time of writing, the initial lineup of participating artists has been announced, including Matthew Dear and Martin Gore of Depeche Mode amongst others. As usual, there is also an extensive programme of electronica-related workshops and exhibitions. Tickets range from $249 to $499 for a VIP ticket. For more details, see the link below.
www.moogfest.com

Solid State Logic SiX

Analogue Mixing Desk

It may be the smallest mixer they've ever made, but the SiX shares more than just heritage with SSL's legendary large-format consoles.

HUGH ROBJOHNS

To many, the large-format SSL mixer represents the very quintessence of the recording console. But although still a very important part of SSL's business, the market for such consoles has changed — the majority are probably now sold to educational institutions and very well-heeled personal studios, rather than commercial music facilities. Smaller project studios now drive much higher volumes of pro-audio gear sales, of course, and these typically have very different requirements. SSL have catered for this market for quite some time, with several more affordable, compact, but very high-quality mixers and outboard designed for contemporary DAW-based workflows.

More recently, however, SSL embarked in an interesting new direction, with their impressive Fusion Stereo Analogue Processor (reviewed in *SOS* December 2018). This was a novel product for SSL, not only because it introduced new functionality and addressed new applications, but also because it was the first to be manufactured wholly in China. Most of SSL's product lines are built from sub-assemblies manufactured in Britain but, since SSL's acquisition by the Audiotonix group (in which SSL now sit alongside Allen & Heath, Calrec, DiGiGrid, DiGiCo and Klang:technologies) the company have had access to a very high-quality manufacturing facility in China. As a result, SSL are now developing a range of products that wouldn't have been viable if built with their established manufacturing techniques. This promises to bring some seriously high-quality gear within reach of project-studio users. The Fusion has already proved very successful, and now SSL have embarked on their second overseas adventure: the new SiX console.

SSL SiX $1499

PROS

- The most affordable SSL yet!
- A genuine mini SSL in terms of technical performance, features and capability.
- Thoughtfully flexible and versatile design with masses of mic gain available.
- One-knob channel compressors and adaptable EQ on mic/line inputs.
- Effective DI inputs with plenty of gain and headroom.
- 12-channel summing bus with G-series bus compressor.
- Talkback channel can be employed as additional mic input, complete with the listen-mic compressor.

CONS

- USB or digital interfacing options would have been nice.
- No input polarity reverse switches.

SUMMARY

The most affordable SSL ever, this is an impressively well-built mini-console. Its well thought-out and versatile capabilities should make it very attractive indeed for a wide range of small-scale but quality-conscious applications. If paired with a decent audio interface, the SiX could double up as both a high-quality analogue front end for your DAW and a monitor controller.

SiX Of The Best

Apparently, the SiX (so named due to its input channel count) was conceived a couple of years ago, but initial build costs meant it was forecast to retail at over £2000, which was thought to make it commercially unviable. Building it in China has allowed a much lower manufacturing cost without sacrificing component or build quality so, after a year in production development, the SiX is now a welcome reality.

The design is classic SSL, with ultra-clean 'SuperAnalogue' circuitry throughout, mostly derived directly from existing SSL consoles. DC servos are used to manage inter-stage coupling, so there are no electrolytic capacitors in the signal path, and the SiX's high-end performance is clear from its impressive technical specifications — which I confirmed though tests with an Audio Precision analyser. For example, the maximum output level is +27.5dBu, and the worst-case noise floor (ie. with all channels routed to the mix bus) is better than -85dBu, giving a dynamic range of around 112dB. With just a single channel routed to the output (such as when recording a single vocal mic) the noise floor is better than -90dBu, giving a dynamic range of more than 117dB. The THD+N figure is equally impressive, at 0.0015 percent, and the console's -3dB bandwidth extends from 4Hz to well over 80kHz, which should ensure precise transient and time-domain behaviour with minimal in-band phase-shifts. So there's clearly been no corner-cutting here — the SiX delivers genuinely superlative, professional, high-end performance.

Opening the smart Apple-esque shipping carton reveals a mini-console that resembles other compact desktop mixers, but even at first glance there's a distinct air of purposeful professionalism and quality here, with five long-throw faders and a generous sprinkling of knobs and buttons, all in SSL's familiar style. The SiX is wedge-shaped and half rack-width (kits are available for rackmounting a single SiX, or two of them arranged side-by-side), with two mono mic/line and two stereo line input channels to deliver the six inputs — but there's *much* more to this little mixer than that!

The headline features start with the two mono mic/line channels, which both also have DI capability, one-knob compressors, two-band EQs, balanced insert points (the sends

doubling as direct recording outputs), and two stereo cue sends. Alongside are two stereo line channels. While these also have access to both cue sends, they lack the EQ, compressors and inserts. Two external stereo line inputs are also provided and are routable into the main mix bus A, the foldback outputs or the monitor bus.

In addition to the stereo mix bus A there's an alternate stereo mix bus B, a simple but capable monitoring section with main and alt speaker feeds, and talkback (complete with SSL's infamous Listen Mic Compressor, or 'LMC') to the two stereo foldback outputs. In total, this mini-SSL can accommodate 12 inputs for analogue summing, and there is even a simplified G-series bus compressor for a bit of 'mix glue', plus balanced inserts on the mix bus A path.

Most of the physical inputs are, conveniently, on the top panel, while the outputs, inserts and ancillaries are accessed from a recessed panel at the rear. The mixer is powered from an external universal 'line-lump' PSU, and is convection-cooled through vents under the front and at the rear. In use, it gets noticeably warm, with the top of the rear panel reaching a toasty 41 degrees Celsius after a few hours.

SSL's design team have clearly tried to make the SiX as flexible and versatile as possible for the widest imaginable range of applications. They see these as including tracking and analogue summing in project studios, small location recording sessions, general mixing and monitoring duties for small-scale audio-for-video post-production, voiceover, podcasting and broadcast/streaming facilities, and even on-stage source mixing and monitoring for musicians in live-sound applications.

For me, an obvious elephant in the room is the SiX's lack of direct computer interfacing — the SiX is 'just' an analogue mixer. That will doubtless disappoint some potential customers, and could limit its market in the face of properly integrated »

The SiX features several channels of compression, including two one-knob channel compressors, a 4:1-ratio stereo bus compressor, and a listen-mic compressor for the talkback input — which, incidentally, can be used with a little creative routing as another mic input channel at mixdown.

» preamp/DAW interface/monitor-controller products from the likes of Apogee, Audient, Focusrite, UA, and some less high-end small mixers. On the upside, though, I can't think of any other compact desktop mixer that matches the SiX's sound quality and feature set, let alone its gorgeous styling. And who knows? A combined mixer/interface combo is well within SSL's capabilities for the future... A SiX-Extra perhaps? Meanwhile, you'll need an audio interface that has at least a stereo input and output if you want to hook this mixer up to your DAW software.

In Depth

A mixer's functionality can usually be inferred from its connectivity, and that starts at the rear of the SiX in a recessed vertical panel. Located here are the main stereo mix bus A outputs on XLRs, with mix bus B's outputs on quarter-inch TRS sockets. All are electronically balanced — there are no audio I/O transformers in the SiX. Two more pairs of TRS sockets provide the balanced main and alt monitoring outputs, while another two pairs deliver the two balanced stereo foldback outputs.

For the uninitiated in 'SSL-speak' I'll explain, so that we're all clear on the terminology, that what you may think of as the channels' aux sends are called 'cue sends' here. These cue sends don't actually have dedicated outputs of their own, but become a selectable source for the artist foldback outputs, which would normally be used to feed the performers' headphone amplifiers or stage monitors. And in case you were wondering, these foldback outputs can, of course, also be used for effects sends or clean feeds (mix-minus) and so forth, if required.

The external line-lump power supply connects using a five-pin XLR — an interesting choice, since only two of the pins are actually needed — and delivers 15V DC at 3.3A (which is why the console runs slightly warm). DC-to-DC converters inside the mixer generate all the required symmetrical power rails and phantom voltages. The screen terminals of the audio connectors are grounded directly (less than 0.5Ω) to the safety earth connection at the PSU's IEC mains inlet, and a spectral analysis of the mixer's residual noise floor showed it to be very clean, with a few (mostly mains-related) spikes all below -120dBu. An on/off button is tucked away immediately above the XLR but doesn't isolate the PSU from the mains supply, obviously.

Various other I/O connections are accessed through a pair of 25-pin D-sub sockets wired to the familiar AES59 (Tascam) standard. The output socket carries insert sends for stereo mix bus A as well as the two mono input channels (the latter being usable as direct record feeds). There are also (parallel-wired) duplicates of both the mix bus A and main monitor outputs, which could be useful for feeding hardware meters, perhaps. The input D-sub connector receives insert returns for mix bus A and the two mono channels, and also accepts 'alternative' (mono) line inputs, which are selectable sources for the two mono channels' stereo cue sends (see below).

If you've been counting, you'll realise that this particular input socket has two unused channels — and I was surprised that SSL haven't seen fit to make constructive use of them, since several practical opportunities spring to mind. For example, direct I/O access to the talkback section's LMC compressor might have been nice, or a dedicated stereo PFL output for external monitoring (always handy in a broadcast/post-production application), or even duplicating the mix bus B outputs to make it easier to sum bus B into bus A for clean-feed (mix-minus) applications. Of course, these potential ideas would break the convention of all analogue AES59 channels operating in the same direction... but rules are made to be broken after all, and it wouldn't be for the first time!

Moving to the front panel's connectivity, the two mono channels are each provided with an XLR for the mic input and a quarter-inch TRS socket for the line/high-impedance DI input; individual phantom power and the appropriate input type are selected on adjacent buttons. The two stereo channels are equipped with pairs of TRS sockets, cross-normalled to ensure that a single left-channel input automatically appears as a dual-mono signal in the centre of the stereo image. Two more pairs of TRS sockets accept the two external stereo line inputs, with independent level controls located in the master section.

Surprisingly, there's no internal talkback mic, but a third XLR on the top panel (with »

The rear panel plays host to the various analogue outputs as well as the inlet for the external DC power supply.

» independently switched phantom power) accepts an external talkback microphone signal, and it's positioned perfectly for a gooseneck-type mic to plug straight in. There's also a conventional quarter-inch stereo (unbalanced) 'engineer's headphone' output socket here, too.

Signal Flow

Working through the signal paths, the mono channels adopt the familiar practice of padding down the line input when selected (by a nominal 9dB in this case) and routing it through the mic preamp. The gain range for the mic input is marked +6 to +72 dB. My bench tests revealed the range to be 0 to +71 dB, but this was measured from the input to the main output, with the faders at zero and the pan-pot in the centre position — the range would be offset 4.5dB higher with a hard pan, or if measured at the insert send. Although there's no input pad, the maximum mic input level is a very healthy +20dBu before things start to sound edgy, and the incremental gain spread around the control's rotation is relatively uniform. I noticed a slight 'gain rush' at the clockwise end but it was barely perceptible compared with so many budget preamps and it wasn't a problem in practice at all.

Bench tests revealed that the line input is actually padded down by 8dB, so that the gain control then ranges from -7 to +61 dB, and that applies to the DI mode, too, which simply raises the input impedance of the line socket from 10kΩ to 1MΩ (the mic input presents the traditional 1.2kΩ). Another button activates a second-order (12dB/octave) high-pass filter with a corner frequency of 75Hz, but there is no polarity reverse option, sadly. At maximum preamp gain, the residual noise floor from the mic preamp measured -55.5dBu, so the EIN figure comes out just under -127dBu, which is very good.

A bypassable two-band EQ comes next. Each band offers standard ±15dB low- and high-shelf equalisation, with the corner frequencies at 60Hz and 3.5kHz, respectively. However, each band can be switched separately into a bell mode, for which the centre frequencies are 200Hz and 5kHz. This ingenious arrangement provides considerable versatility, and I found the options to be very effective in real-world applications.

Following the equaliser is a bypassable 'one-knob' compressor, which, though inspired by the channel compressors of SSL's large-format consoles, is actually an entirely new design. Equipped with a 'traffic-light' gain-reduction meter, the single threshold knob spans a range of +10 to -20 dBu to determine how much 'squash' is required, while the release time and ratio are fixed at benign settings of 300ms and a hard-knee 2:1. The attack time, though, is programme-dependent and varies between 8 and 30 ms, which is fast enough to exert reasonable control over the dynamics without being overly aggressive. An automatic gain make-up system preserves the output level, adding up to 10dB of boost for the lowest threshold setting.

Next comes the balanced insert point (post-EQ/post-compressor) and the send is always active, as already mentioned. Since the high-pass filter, EQ, and compressor can all be fully bypassed, the insert send can deliver the preamp's direct output, if required, for ultra-clean direct-record tracking. The balanced return is selected by a button alongside the channel fader, and the post-insert/pre-fade signal is displayed on an eight-segment bar-graph peak meter scaled from -21 to +24 dBu. The fader is configured with 10dB of gain above the unity position, and the channel pan pot applies 4.5dB attenuation at the centre (relative to each end). By default, the panned channel output is routed to the main stereo mix bus A. A large button near the bottom of the fader routes it instead to mix bus B, if desired, but also serves as a channel mute if mix bus B is not being used. A smaller, latching button activates PFL (with a yellow status LED).

The two stereo line channels are much simpler affairs, offering just a level trim control (-10 to +20 dB) at the top of the strip, a stereo bar-graph peak meter alongside the stereo fader, the bus B/mute and (stereo) PFL buttons, and a balance control instead of pan.

I previously mentioned the two stereo cues, which can be accessed from all four input channels, with individual level controls and 'on' buttons. The two mono channels also feature pan controls for each cue send, and 'Alt' buttons for cue 1, which replace the channel's signal with that alternative external line input arriving through the D-sub connector. The cue sends are all normally derived post-insert but pre-fader, »


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Despite the smaller size, the SiX looks and feels like a 'real' SSL, and the familiar coloured knob caps aid navigation of the (inevitably slightly crowded) controls.

» but they can be changed to post-fader using buttons in the foldback master section, which reconfigure all cue 1 and/or all cue 2 sends globally.

Master Section

The stereo mix bus A output fader (with 10dB of gain above the zero mark) is on the right-hand side of the console. Immediately above is a group of four buttons with status LEDs, the first of which activates the mix bus A insert return. The others enable the additional summing bus contributors: External 1, External 2 and, rather unusually, the stereo cue 1 bus (so that those alternative mono external inputs described above can be employed in the mix).

As a result, mix bus A sums two mono input channels, four more from the two stereo line input channels, another four from the two External stereo inputs, and yet another pair by way of the mono channels' cue 1 external inputs. That makes 12 summing inputs in total (10 line and two mic/DI/line), which is quite an impressive feat for such a compact device. This feature also works for bringing stereo effects returns into the master stereo mix, or for cascading an expansion mixer, of course.

Mix bus A's signal path is quite simple, with the mix bus summing amplifiers driving the master insert send, and then the bypassable return feeding into a (simplified)

G-series bus-compressor ahead of the main stereo output fader and output drivers. Mix bus B is even simpler, with the mix bus summing amps driving a rotary fader (with mute button) before the output appears at the rear-panel connectors. If you were wondering (because they're not entirely obvious at a glance), the mix B rotary fader and mute button can be found nestling amongst the profusion of knobs and buttons in the monitoring section, alongside the main output fader.

Interestingly, while the stereo bus compressor is based directly on the classic and much-loved G-series design, it has been simplified and uses a more modern VCA chip here (although the SSL gurus assure me it sounds exactly the same!). Its three controls are located at the top of the desk, alongside the output metering, and comprise threshold (-20 to +20 dB), gain make-up (0 to +20 dB), and an 'In' button. There's also a five-segment gain-reduction meter (0-15 dB). The attack and release times are fixed, as is the very soft-knee 4:1 compression ratio — settings which I'm told were chosen to reflect those most commonly used on the big studio consoles.

Monitoring

The monitor section is reasonably versatile, with mix bus A, mix bus B (both post-fade/post-mute), External 1 and External 2 (post-level controls) all available as sources which can be auditioned individually or in mixed combinations. There's one potential 'gotcha' to watch out for, though: with no source selected there's no monitoring output at all, and it's not entirely obvious if the buttons are depressed or not! I was also surprised to find no way to monitor the foldback outputs to check what the artists are hearing, though the two stereo cue sends can be auditioned on the headphone output. A large, 11-segment bar-graph meter displays the monitored signal and is scaled from -21 to +24 dBu, with greater resolution above the 0dBu mark than the input meters. Dedicated segments for +15, +18 and +24 dBu make alignment with standard DIN, EBU and SMPTE A-D converters very straightforward.

Input channel PFLs always override the selected monitoring source(s) when activated, and there's also a mono button (introducing -3dB attenuation of the summed stereo signal), monitor mute, and output routing to either the main or alt speakers. Frustratingly, while SSL have seen fit to provide an adjustable -3 to -30 dB dim facility, they've not included a right-channel polarity reverse button, so it's impossible to check the stereo difference signal (something I find essential!).

A single rotary level control, which is the largest knob on the console, sets the listening level for the main and alt outputs, and the engineer's headphones have a separate volume control. Normally, the headphone output carries the selected monitoring signal (including the PFL override), which is derived after the mono button but unaffected by the dim, mute and monitor level controls. The headphones can also audition the cue 1 or cue 2 mixes (as opposed to foldback 1/2), if desired, which is useful if an artist is working in the control room.

More usually, performers would use the mixer's two stereo foldback outputs for their studio headphone amp, and each foldback output has its own rotary master level control. Logically enough, foldback 1 carries the cue 1 signal, and foldback 2 the cue 2 signal, but these can be replaced individually with either of the two External inputs or with talkback (all using separate latching buttons). I was surprised that there are no LEDs to warn when talkback is active, particularly since the buttons are mechanically latched types. However, this latching arrangement does enable the talkback signal path to be used as a third mic input by routing the relevant foldback output back into an input. This makes it practical, for example, to record a drum kit, with kick and snare mics through the two mono channels and an overhead/room mic via another line channel (and squashed through the LMC if desired).

SiX Appeal?

The SSL boffins have done a very good job with the console ergonomics on the whole, especially given this mixer's very small footprint and generous feature set. The panel layout is inevitably quite 'busy', though, and I found I had to look carefully to find the function I wanted even after using the desk for a week. Thankfully, there are status LEDs almost everywhere that matters, and coloured knob caps help to identify the different features (although the adjacent proximity of channel and cue 1 pans on the mono channels often confused me). The monitoring section requires careful use, too, and some print shading under the monitor section controls might help to separate them visually from unrelated functions.

Internally, the SiX uses surface-mount components almost universally, as you'd expect, but the standard of construction is everything you'd hope and desire of an SSL product; it really is outstanding. The lower panel I/O connects to a very high-quality PCB on the base of the unit carrying the balanced receiver and driver circuitry, and that is linked via a couple of ribbon cables to a second PCB suspended under the front-panel controls. This board is enclosed within its own steel casework, which provides additional screening and mechanical support. Servicing might not be terribly easy, but it's certainly a solidly built piece of kit which oozes quality, ruggedness and longevity wherever you look!

Although there are a few surprising and potentially disappointing omissions (the lack of input polarity reverse buttons, a stereo-difference monitoring facility, active talkback status LEDs, and foldback AFLs), I have to say that this is a lovely little console which performs as well, and is as nice to use, as any of its much larger siblings. It really is a remarkably versatile and capable little mixer which slots in very elegantly alongside the X-Desk and XL-Desk in SSL's range. It would seem logical to assume a larger version will follow in due course, now that SSL have access to more cost-efficient manufacturing, but to my mind, much more appealing to many prospective users would be a version featuring an integrated audio interface. Who wouldn't want an affordable SSL console providing the front and back ends of their DAW-based studio setup?

Sound wise, the SiX's performance is exemplary. Headroom margins are generous, noise and distortion are as low as best practice allows, and the bandwidth is huge — all of which means this desk sounds completely transparent. And yet, the one-knob compressors and EQ on the mono input channels can be used creatively to shape sources when necessary, and the stereo bus compressor does exactly what is needed to glue a mix together.

It's impossible for me not to fall in love with this sweet but extremely capable little console, and think up countless applications and justifications for purchasing one — and plenty of people will feel the same!

Alternatives

Small desktop mixers may not be uncommon, but there is no other that is so compact, yet so well-equipped and versatile as the SSL SiX — and certainly nothing of this size that can match its technical performance.

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Patch & Tweak

Book Review

The team behind *Push Turn Move* shift their attention to modular synthesis.

SIMON SHERBOURNE

Patch & Tweak follows on from last year's *Push Turn Move*, but where the first book from Kim Bjørn's team was an exploration of design and user interfaces in electronic music, this time the focus is all on modular systems, and especially Eurorack. As such it may be what you were hoping for the first time around.

While *Patch & Tweak* isn't the first book about modular synthesis, previous efforts have tended toward the monochrome in both tone and imagery. The fact that publishing this glossy celebration of modular is economically viable, with it's 350+ edge-to-edge full-colour pages, is a testament to how popular an interest modular has become.

The closest comparison on my book shelf is the rather beautiful *Stecken Schrauben Spielen*, published by SchneidersBuero in 2008 as something between a book about the emerging Eurorack scene and a brochure. Like that book, *Patch & Tweak* manages to combine a bestiary of modules with interesting and engaging snippets of back story, comparison and usage ideas.

As in *Push Turn Move*, the tour of concepts and hardware is broken up by a large number of interviews and patching tips from leading lights in the scene, from artists to makers, sound designers to YouTubers.

Symbolism

After the foreword from synth royalty Suzanne Ciani, there's a concise Modular 101 section, covering the basics of synthesis, sampling and synth architecture. Perhaps the team worried that diving straight into too much theory would put people off. I would have been happy with a bit more. Thankfully, as the book settles into its taxonomy of synth modules it continues its gentle instruction, in the context of the hardware categories.

You might wonder how well patching concepts can be communicated via the silent medium of print. To aid in this the authors have revived and revised the idea of a common set of graphical icons for drawing patch schematics. They propose to 'open source' these for general usage and build a web tool for archiving and sharing patch ideas.

The book explains the various modular formats past and present: of course Eurorack, but also Serge, Buchla and Moog. The lion's share of the book is then a tour through the various categories of modules that make up the modular universe: oscillators, VCAs, sequencers, samplers, clocks, etc. For this, Kim and co stick to Eurorack devices. This may disappoint if you're primarily interested in one of the other synth ecosystems, but I think it was a good decision, ensuring all the wonders presented are for the most part mutually compatible, and contemporary.

Analogue

Much has been written before about the experiential difference between using hardware synths and their software equivalents. Ciani's foreword puts it in terms of "the way it engages both our minds and bodies". There's an obvious analogue (if you please) with the contrast of a physical book or magazine versus a Kindle, website or video. A book invites you to flick, explore, let your attention be caught, and enjoy the tactile interaction. *Patch & Tweak* is a great resource for beginners and modular veterans alike, and every visit to it rewards with some new morsel of information as well as the simple pleasure of browsing.

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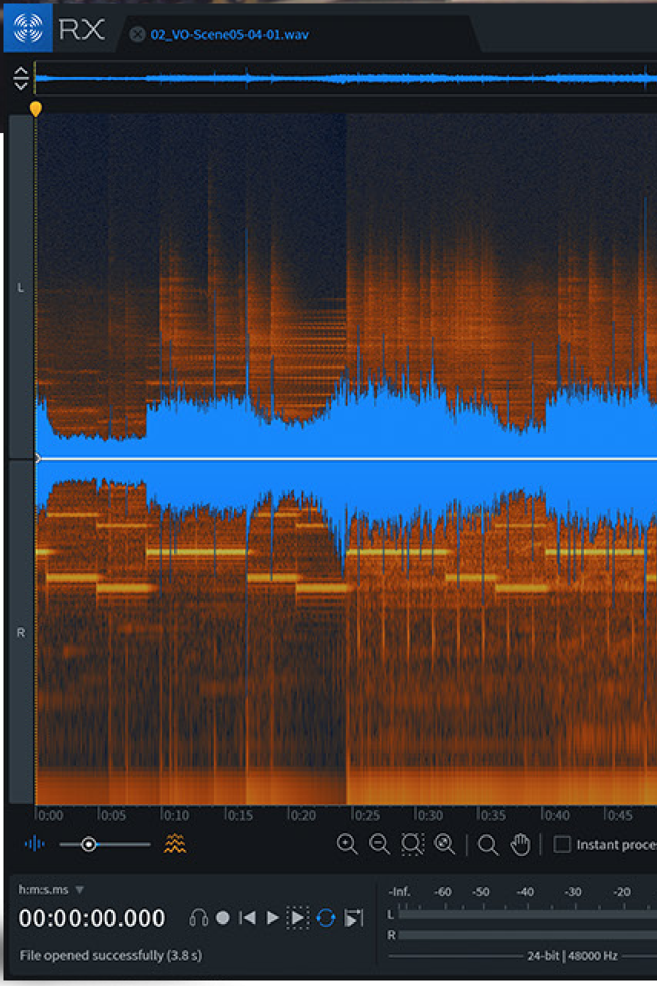

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Polyend/Dreadbox Medusa

Synthesizer & Sequencer

The Medusa combines analogue and digital, synthesizer and sequencer — and even throws in a controller for good measure.

PAUL NAGLE

The driver behind any collaboration must surely be to play to the strengths of those involved. In today's example this means a blend of Polyend's digital technology with the analogue skills of Dreadbox. I was already acquainted with the latter, having previously enthused over the Erebus synth, so I was keen to discover what Polyend brought to the table — literally in this case, since the Medusa is a tabletop synth, sequencer and controller rolled into one.

The synth engine features six oscillators (three analogue, three digital), an analogue filter, five LFOs and five envelopes, while the sequencer boasts up to 64 steps of notes and synth parameters. There are three voice modes, one of which is a sort of 'paraphonic plus' where chords of up to six notes can be played and sequenced. And as a final bonus, the pads can be put to general controller duties with a DAW or other synths.

Legend tells us that Medusa is associated with being turned to stone. Personally, I was keen to discover how it coped when associated with a stoner...

Hardware

The Medusa is a thoroughly black and white experience, its low profile metal box sporting an 8x8 pad matrix, two neat and petite OLED displays and a number of buttons, knobs and sliders. Operationally, it's a game of two halves with a minimal

transport and menu section sitting between the pads and synth controls.

Generally the knobs and sliders feel OK, although the knobs wobble slightly. The audio connectors are more solidly held in place and consist of a single audio output, an input (for processing audio via the analogue filter) and a phones socket — all on quarter-inch jacks. The round white buttons aren't the snappiest or most responsive though; occasionally I hit Play on the sequencer but the Medusa completely failed to notice.

Having a layout with the pads on the left-hand side and the synth controls on the right felt slightly contrary to me as a right-hander. The small silicone pads have a pleasant squishiness and each one generates data in the X plane (pitch bend, side-to-side), Y plane (mod wheel, up and down) and Z (channel aftertouch, pad pressure). A recent firmware update also added velocity to the list of attributes transmitted from the pads in its controller guise. With a generous choice of scales and pad layouts, Medusa is therefore a performance surface that will take time to fully appreciate.

MIDI is supplied via regular five-pin connections or USB, with the latter also used for firmware updates and patch dumps. Power is from an external 15V adaptor and the power-up sequence includes a calibration routine for the analogue oscillators, after which you're good to go.

Hybrid High Jinks

Polyend have crammed quite a lot of synthesizer into an area measuring just 227 x 170mm. Inevitably, this has been achieved by sharing controls and by tricks such as using single buttons to step through values. At any time, you can edit either trio of analogue or digital oscillators — a 'Digital' button lights to indicate when the latter are live on the controls. Regardless of type, every oscillator has four standard waveforms (sawtooth, square, triangle and sine) and three choices for octave shifting. You can squeeze a further octave up or down from the Tune knob, a control that switches from semitones to fine tune at the press of the appropriate button.

You quickly realise that a number of the controls are 'analogue only'. For example, the PW knob changes the pulse width of the analogue square waves and the two Sync buttons synchronise analogue oscillators 2 and 3 to the first. The same limitation applies to the FM knob. Oscillator 3 serves as a modulation source for 1 and 2 and the filter cutoff frequency, the knob's action tied to a pair of selection buttons. This is exponential FM and therefore a source of clangorous cross modulation, ripping filter noises and typically edgy material rather than the more refined linear FM.

The oscillators are further supplemented by a versatile noise generator with a built-in filter that morphs the noise colour from brown through to violet — or deep and boomy to high and hissy, if you prefer. All seven audio sources are blended in the mixer, along with any external input signal (for which no on-board attenuation is provided).

Lacking oscillator sync, PWM or cross-mod, it might look like the digital oscillators have been left behind. Fortunately, the Wavetable knob offers a clue to something extra lurking beneath the covers to redress the balance. To explore further, you must select wavetable playback by stepping through the regular waveforms until all four waveform LEDs light up.

The Medusa possesses 20 wavetables, each with 30 waveforms. Browsing the available tables involves the menu system, entered by a press of the unlabelled encoder in the middle. Turning the encoder scrolls through various options and the last of these is Wavetables. Right now it's just a list of numbers, which isn't terribly helpful, but Polyend say there may be alternative tables in the future, along with the possibility of names.

Auditioning each in turn, I found the first wavetable to be packed with glitchy transitions, the fifth is vocal in nature, several are organ-like and some feature smoothly morphing synth waveforms. While it's not the most comprehensive wavetable implementation out there, it's a welcome addition for fleshing out the Medusa's sonic range.

The Wavetable knob selects the wave within the current wavetable and, as a bonus, the right-hand OLED supplies a graphical waveform view. Naturally I wanted to try automatically sweeping through the table or giving »

Polyend/Dreadbox
Medusa **$1199**

PROS

- A versatile blend of analogue and digital synthesis.
- Paraphonic, capable of up to six-note chords.
- A different type of synth playing surface.
- Grid storage of parameter modifications is wonderful.
- Step sequencer includes parameter recording.
- Solidly made.

CONS

- The sequencer is basic and rather menu-bound.
- The pads are small and not the most responsive.
- MIDI spec could be improved.
- Some aspects feel unpolished; the divide of duties between Notes and Grid modes feels awkward.
- Shared controls, so less immediate than it might be.

SUMMARY

The Medusa is a powerful and often subtle hybrid synth whose killer feature is the ability to store up to 64 complete sets of parameter tweaks in every patch. Also acts as a control surface and sequencer.

The Medusa's front panel. Around the back are MIDI in, out and thru ports, a USB port and quarter-inch jack sockets for audio out, headphone out and audio input.

each digital oscillator its own wave, which required digging into the Medusa's modulation system.

Modusa

Modulation is a straightforward affair in which a single source, either an envelope or LFO, is aimed at a single destination. The source is selected by button, at which point dedicated Amount and Target controls come into play. There's also a shortcut method of allocating the target, by holding a source button and moving one of the panel controls (or, in the case of FM, pressing one of the FM buttons). With the exception of the envelope stages, most of what you'd expect to modulate is up for grabs — including LFO waveforms, rates and amounts. Once you grasp the concept, it's therefore easy to set up modulation of the pulse width, a sweep through the wavetable or a whoosh of noise. When you aren't tweaking controls, the default state for the right-hand OLED is to dynamically represent the movement of all 10 modulation sources.

It's worth noting that PWM applies to all analogue oscillators with a square wave selected, but you are able to modulate the position in the wavetable independently for each digital oscillator. In doing so, there are a couple of operational idiosyncrasies to grasp. First, and rather annoyingly, the target list is obscured by a box saying 'OFF' until you activate the LFO or envelope by double-clicking its button. Secondly, there's no visibility of the modulation source that's currently live on the controls. Only after you move an envelope slider or LFO knob does the relevant button flash, which seems needlessly obtuse.

The envelopes have an initial delay stage and a Loop button. And while the envelope only loops during 'note on' it's still a great bonus, especially if you start running out of LFOs. LFO speed is displayed in Hz (yay!), in the range from 0.01Hz to 30Hz. When synchronised to the current tempo, the values switch to note values, from six through to 1/64.

Granted, the method of 'single source to single destination' is no substitute for a full modulation matrix, but it scores by being fast and uncomplicated. The LFOs have an extra trick up their sleeves too: fully morphable waveforms from sine through to sawtooth, taking in square and triangle on the way. The only omission is a random (S&H) source.

Analogue Filter

The Dreadbox analogue filter offers three modes: low-pass 12dB, low-pass 24dB or a high-pass 6dB slope. Successive firmware updates have gradually shunted the loud and ugly resonant squeal towards the very end of the knob's travel, although it's still lying in wait for the unwary. (When scrolling through modulation targets, you quickly learn to keep the modulation amount low to avoid the speaker-shredding peak as you pass resonance in the list.)

While the filter isn't quite so fluffy and sweet as the Erebus', it's a splendid way to add warm squelch and fuzziness to the analogue and digital waves. The high-pass filter is an underwhelming but necessary tool for thinning out the super-thick textures that half a dozen sometimes detuned oscillators can produce. As you'd expect, the note tracking knob allows high notes to be brighter.

The synth engine's final cog is found in the small Play Mode section and offers glide plus three voice modes. In the first of these, Mono, all six oscillators are stacked to deliver some of the Medusa's most obese solo and bass sounds — or delicate, shifting layers if you've chosen to modulate the oscillator levels. The next voice mode is P1, which invokes three voice paraphony, each voice having a layered digital and analogue oscillator. Lastly, P2 allocates the oscillators as six separate voices, each with its own amplitude envelope. The proviso is that they are still all positioned before the single filter.

Digging into the Config menu to specify a Voice Priority of 'Next' tells the Medusa to play each oscillator on a round-robin basis. If you then open the filter fully you can achieve pretty credible six-note pads and chords. Combining this mode with the oscillators set to different waveforms is a superb way to demonstrate that paraphony isn't necessarily inferior to regular polyphony, just different.

Finally, if you're ever stuck for inspiration, Polyend fitted a Random button to randomise synth and grid parameters.

Pads, Modes & Performance

Up to 128 patches can be stored in two banks (A and B), with each patch

CP88

CP73

» comprising patch, sequence and grid data. The latter consists of up to 64 pads' worth of parameter modifications. Patches aren't named but selected using the Load button in conjunction with the pads. A nearby hold button is used to drone the last triggered note (or chord) at the current sustain level.

Operationally, the pads are used in two distinct ways: Notes mode or Grid mode. In the former, the pads trigger notes and transmit control values based on finger position and pad pressure. Since the pads are quite small, I found it a challenge to play with any great precision or smoothness. And sometimes my touch was simply too light for every hit to register, especially when playing left handed.

In Grid mode (the Grid button is lit), each pad replays any data stored in it, with the type of data shown by various brightness levels — more about this later. For now we'll remain in Notes mode, where a number of squares are lit to indicate the root notes within the current scale.

To see the root and scale values, you need to enter the menu system — a list of top-level menus for defining pad behaviour, sequencer settings, etc. Spinning through until you reach the Scale entry, you're able to choose from an impressive 40 scales and modes, covering the familiar and the exotic, eg. Marva, Hirajoshi, Iwato, Yo and Enigmatic. Although there's no user scale option, I reckon it'll be quite some time before you work your way through the ones on offer. Pad layouts and root notes are selected in the same manner.

Layouts are numbered from 1-8, followed by 'Guitar', which sets out the notes in a way not unlike a guitar fretboard. As you audition each, the pad display shifts to indicate the new position of root notes.

Playing a synth from pads is obviously a very different experience to playing a keyboard, but once you get past the unfamiliarity, fresh phrases caused by new finger positions should soon burst forth to surprise and maybe delight. Oddly enough, this was when I most missed the ability to name patches as that could have helped me remember the key and scale used in each.

Also within the menu system are the assignments for pitch bend, mod wheel and aftertouch (X, Y and Z) with targets already familiar from the modulation section. You could therefore wiggle your finger left and right to bend pitch, push it upwards to open the filter and apply pressure to increase volume — and the allocations you prefer may be saved on a per patch basis.

In another menu — Config — you assign the MIDI channels, synchronisation and so on. For its controller roles, you can choose whether the Medusa sends its data via five-pin MIDI or USB. It even supports MPE — polyphonic expression for those devices that support it (sadly I didn't have one to try).

Via a special mode you can address the six oscillators independently via a (fixed) selection of MIDI channels. This acts like a kind of 'multitimbral lite' mode when controlled by an external sequencer. Sadly you can't capture these multi-channel sojourns into the Medusa's own sequencer though.

Other parameters here include setting whether panel tweaks should take place instantly or only when the knob or slider passes through the stored value. Calibration is there too, and from time to time you'll need to perform this manually since the analogue oscillators occasionally drift. This is more noticeable when stacked with the digital oscillators and may even be considered a feature by the perverse. The Medusa doesn't feature a master tune, although hopefully this may be added in the future.

Firmware & MIDI

I worked my way through a number of firmware revisions before starting this review. Version 1.1 added a means of updating the firmware and dumping/restoring banks of patches via a dedicated app, MedusaTool. I was unable to get this working on my Mac (it has quite an old OS) and on my Windows 10 laptop, the app obstinately remained as a tiny box despite my best efforts. However, since Polyend have not provided any means of dumping patches and banks via regular SysEx, it's important to stick with it. (Maybe I'm unusual in having a store of patches in SysEx format dating back to the 1980s, but I can report these still load painlessly into synths today. I doubt this would be the case if I required bespoke software from the time!)

The MIDI spec currently feels incomplete and slightly idiosyncratic. I've mentioned in the review text that velocity is ignored by the synth engine, but some of the standard MIDI CCs have been oddly allocated too. The most significant of these is CC7 (volume), which has been hijacked for oscillator 1 tuning! The Medusa doesn't respond to MIDI Program Changes either, so patch selection is always a manual affair.

Sequencer

As far as the sequencer is concerned, the operational modes again play distinctly different roles. Notes mode is firmly designated for the recording of notes and chords, which are replayed using the values of the current patch. Any X, Y and Z performance data you happen to play at the time is ignored, as is velocity (all notes are stored at maximum velocity). I suppose this makes some kind of sense as the synth engine doesn't respond to velocity, but it limits the possibilities for sequencing external gear.

When in Grid mode, the sequencer triggers stored notes plus Patch Modifying data — a similar concept to Elektron's Parameter Locks. This PM data can be recorded freely during playback or by selecting a step (or steps) into which parameters should be captured, or cleared from. There seems no limit to the amount of parameter tweaks held, although sadly there's no way to see what they are prior to a clearing operation. Instead, Polyend employ varying degrees of brightness to show whether a step contains no data, note data, PM data or note plus PM data. However, the different levels aren't always easy to distinguish and there's no avoiding the thought that coloured pads would have made life much easier.

In use, the sequencer is simple if slightly menu-bound. You have to enter the menu to set the tempo and swing values but you also need to go in every time you want to change the pattern length (1-64 steps) and direction (Forward, Backward, Pingpong and Random). If those are the kind of things you'd typically expect to do live, it feels somewhat awkward. The sequencer runs at a fixed 16th-note resolution, which is, again, limiting.

As every patch holds a single sequence and there is no chaining or song mode, you quickly learn to embrace simplicity and make extensive use of Patch Modifying data to spruce things up. Fortunately, this is a source of many pleasant surprises, eg. adding fast LFO modulation on certain steps to simulate note repeats, tying specific waveforms to certain notes or simply adding envelope and noise tweaks to create percussive grooves.

Slipping back into Notes mode for a moment, you can transpose a running pattern by holding the Hold key and pressing a pad. However, switching during playback causes the sequencer to switch between honouring the PM data and ignoring it. Since this can lead to significant tonal leaps, you'll need to find a way to either avoid Notes mode or incorporate these leaps into your performance. Although you can edit notes in Grid mode, you can only record them in Notes mode — regardless of whether you're using the Medusa's own pads or an external keyboard.

If you recall a patch while a sequence is running, it switches immediately rather than at the end of the pattern. And if you save during playback, there's a rather unpleasant glitch, which kills off any hopes of developing patterns and sequences in a live situation.

Conclusion

Unless you're Perseus, there are multiple ways you can approach the Medusa. It could be a MIDI controller transmitting notes and performance data into a DAW environment, or perhaps a stand-alone pad-based synthesizer that breaks the usual keyboard conventions. Some will find the built-in sequencer perfectly suitable for melodies, bass lines and chords, especially given the power of Grid mode, where you can sequence what amounts to a different patch on every step if it takes your fancy.

Even if you forget the sequencer and work exclusively in Grid mode, the Medusa offers a rare degree of choice. You can prepare and trigger up to 64 notes or chords, each with specific synth settings, which can inspire genuinely different performance techniques. If you have recorded only parameters into the pads, the notes can be supplied by an external MIDI source as you introduce modifications dynamically.

By itself, the Medusa synth engine is capable of a good range of analogue and digital tones. It's ready to deliver percussion, sound effects, rip-roaring solos, snappy basses and even pads. It's aided considerably by the various voicing modes, the six notes of paraphony and the Dreadbox filter.

At the price, it might struggle to make a huge impact unless its quirky feature set appeals. Each firmware update has brought the Medusa closer to completion and more are scheduled. It would be nice if Polyend addressed the general matters of visibility and accessibility, the lack of names for patches and wavetables, the under-developed MIDI spec and inability to dump and restore patches in SysEx format. Then again, it might already tick enough boxes for some; it's certainly a fresh take on the concept of synth, sequencer and controller.

Alternatives

There are a number of controller solutions out there, many of which are aimed at your DAW rather than stand-alone use. If a pad controller isn't essential, a number of variations on the synth/sequencer theme come to mind. The most colourful might be the compact and powerful **Synthstrom Deluge**, a multitimbral synth, sampler and sequencer with built-in effects and up to 64-note polyphony.

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Waldorf Quantum

Polyphonic Synthesizer

Waldorf's new flagship synthesizer is complex, versatile and hugely ambitious.

GORDON REID

Waldorf are a company with a patchy history. They have released unfinished products — most notably the Wave and the Q — and they even went bankrupt in 2004 only to reappear two years later with much the same product line as before. But they nevertheless retain a reputation for designing interesting products and, while recent developments have tended to be smaller and lower cost, the company are now aiming high again. I must admit to having had some trepidation when I agreed to review the Quantum. Would it fulfil its promise, or would it shower me with 'this feature is not yet implemented' messages as one Waldorf flagship did in the past? Let's find out.

Although the Quantum appears to be a conventional eight-voice analogue/digital hybrid polysynth I can almost guarantee that you'll fail to get the best from it if you plunge in without researching its capabilities. It's not that it's arcane, but there's a great deal going on within it and the 'I never read manuals, it shrinks my testicles' brigade is going to suffer in comparison with players who take the time to understand it. But don't fret about this. Once I had learned the Quantum, I found it to be quick and straightforward to use. But why the concern? After all, the control panel reveals that it's just a three-oscillator-per-voice polysynth with a couple of filter sections, three LFOs,

three contour generators, something called a 'Komplex Modulator', and three effects sections, right? Wrong. Very wrong.

The Oscillators

Each of the three oscillators can host your choice from four sound generation methods and, depending upon the method chosen, all of the LEDs associated with its knobs will light up in a different colour — cyan to denote a wavetable oscillator, green for a waveform (virtual analogue) oscillator, and so on. You can control seven of the most useful parameters for each oscillator type using the large, friendly knobs on the front panel, while many more are accessible from the large touch-sensitive screen. And, to ice the cake, you can even save and recall individual oscillator setups.

Perhaps because of Waldorf's long-standing relationship with wavetable synthesis, this is the first oscillator type offered. The Quantum includes 85 factory wavetables, and these are real, 128-stage wavetables that you can travel through in meaningful ways. You can select the table used, its pitch, key tracking and pan, the position and brightness of its spectral envelope, the amount of noise applied to the steps in the table, and whether any drive or gain is applied. You can also determine how you move through the table, how quickly, and whether the steps are interpolated for smooth timbral changes or stepped with various quantisations for grainy transitions. In addition to using things such as LFOs and contour generators to sweep through the table you can use MIDI note numbers to determine the playback position as well as the resulting sound's noise level and brilliance, so each key can have a different but related timbre from the next. You can also create your own wavetables, either by typing a phrase into the Quantum for text-to-speech-to-wavetable conversion, or by analysing an audio file held in its flash memory. The results can then be saved to the internal memory or to an SD card. (Unfortunately, it doesn't seem possible to give your own wavetables meaningful names, and any that you generate are just called User X, where 'X' is a number, which is going to be a problem once you've created a large library of them.) If you like the glassy, sometimes fragile sounds of the type pioneered by the earliest PPGs, or the evolving sounds at which wavetable synthesis can excel, you'll love this oscillator type.

The next is called Waveform, and this generates the classic analogue waves — sine, triangle, sawtooth and square, with additional white and pink noise options. A Warp function then shapes these into intermediate waveforms and adds things such as pulse-width modulation. However, its apparent simplicity hides several advanced features. One of these »

Waldorf Quantum
$4299

PROS

- A PPG, a VA polysynth, a sampler/granular synth and physical modelling synth in one box... Wow!
- It offers huge synthesis power, and the range of sounds is stunning
- Given its depth, the editing system — although not perfect — is quick and intuitive.
- Version 2 will soon add FM synthesis and more to the sound generation already available.
- It feels solid but it's also attractive — you'll look good standing behind one.

CONS

- Until the MIDI/USB problem is fixed, it may be unusable for some potential owners.
- When a sound passes through the analogue filters it's reduced to a single audio channel.
- The digital filter section is global, not per-voice.
- The use of SD cards rather than USB sticks was a poor decision.
- The review model had a screen flaw.
- It's not cheap (but I don't think that it's overpriced either).

SUMMARY

The Quantum is a massively flexible polysynth. It lacks the polyphony of some of its competition, but it can sound superb and I have to admit that, during the review period, I found it to be quite inspiring at times. Once the teething problems are ironed out, it will be hard to ignore it.

» is Sync. Nothing special here you might think, but there is; the sync oscillator isn't one of the other oscillators, it's part of the oscillator that you're programming. Taking this concept even further, each oscillator offers up to eight 'kernels' that share the same waveform but can have independent pitches for kernels 1 to 4 (which are duplicated for kernels 5 to 8 if used) and independent positions in a stereo spread. To test this type, I started with a single oscillator and used it as the basis of a simple eight-voice, single-oscillator-per-voice polysynth. This placed me firmly in Juno-60 territory. Sure, the sound wasn't the same, but experimentation yielded sounds that I would have been happy to substitute for a Juno. Adding the second and third oscillators and selecting appropriate filter types (which we'll come to presently) took me into the realms occupied by Prophets, Oberheims, Jupiters and the Memorymoog. Even with its unusual PitchVar (pitch variation) parameter, the Quantum didn't ever sound quite like any of those synthesizers, but this isn't a criticism; I programmed many two- and three-oscillator-per-voice patches that I would have been proud to use, and one would have to be an analogue fanatic to deny their quality or musicality. Furthermore, in its monophonic mode, the Quantum also makes a fine virtual analogue monosynth.

The Particle synthesis engine is based upon stereo multi-samples, allowing you to replay them in conventional fashion or in short snippets called grains. The Quantum comes with a sample library of around 1GB within its 4GB internal memory, but you can also import WAV, AIFF and AIFC samples from SD cards, record the synth's own output for resampling, or use its in-built recorder to capture external sounds. I thought that it would be long-winded and brain-twisty to create a multi-sample patch, but this proved to be wrong. I recorded myself singing 'aahhh' at multiple pitches and then selected the Particle option in oscillator 1 and positioned each of the samples across an appropriate region of the keyboard. (I could also have sung at multiple loudnesses and distributed the additional samples across different MIDI velocity ranges, but I didn't.) Next, I chose suitable start and end points for each sample, looped them, fine-tuned them, shaped the results using a filter and the audio signal amplifier, and added chorus, delay and reverb to obscure my awful singing. The results were remarkable and the whole operation had taken no longer than 10 minutes. In fairness, the Quantum isn't a substitute for a dedicated sampler but, if you're happy to work within its limitations, it can yield great results. In addition, if you fancy inflicting a bit of gratuitous violence upon your samples, you can extract grains from them, determining the position from which the first grains are generated, their lengths, the attack/decay amplitude contour applied to each, and the way in which the point of generation wanders through the sample. There are also pitch parameters that allow you to distribute the grains in various ways. Oh yes, and each oscillator can again generate up to eight kernels, so all but the spikiest sounds can be made to sound smoother and more musical. If you're interested in a bit of sonic mayhem and want to derive unlikely sounds from interesting starting points, this is one way to do it.

Unfortunately, Waldorf supply very little information to explain the Resonator oscillator. To quote: "For a better understanding on how the Resonator works, we recommend initialising a sound program and starting with a default Resonator. Try out all the parameters to become familiar with the functionality of this powerful sound creation tool." In other words, they don't tell you what's going on, and even the limited information supplied is confused in places. For example, the manual talks about the partials "fading out exponentially" when it means that their amplitudes are an inverse function of their frequency. Waldorf could do much better because the underlying concept isn't that complex: an Exciter — things such as clicks, noise bursts and samples — energises a Resonator, causing it to emit sound according to the various parameters on offer. You can control the attack and decay of the Exciter and whether it's applied once or whether it repeats, as well as the spread, structure, response and damping of the partials generated by the Resonator, and these parameters allow you to create a wide »

The Rear Panel

The Quantum's rear panel starts with a headphones output and its associated volume control, followed by unbalanced main and auxiliary output pairs. Next there are stereo audio inputs for real-time processing or 24-bit recording of external signals, followed by control inputs for a sustain pedal and a single expression pedal. The digital I/O starts with USB Type A (MIDI in from an external controller) and USB Type B (MIDI in/out). Unfortunately, the USB B interface doesn't carry audio, which is disappointing. These are followed by a slot for an SD card, and five-pin DIN sockets for MIDI in/out/thru. The power input is an IEC socket (hurray!) for the internal, universal power supply.

50 Series

range of sounds that would be difficult to synthesize by any other method. When excited by an impulse, the Resonator excels at things such as plucked strings and hammered sounds including bells, chimes, and electro-mechanical pianos, while experimenting with extended Exciters offers very different results; with the decay of the resonators set to maximum, I obtained some weird and wonderful sustained sounds. I suspect that this is the oscillator type that offers the greatest hope of creating something new and interesting, although it might also be the one that causes you the greatest frustration as it attempts to do what it wants rather than what you want.

Filters, Amplifier & Effects

Let's assume that you're generating the initial sounds that you want — whether using a single type for all three oscillators or combining different types to create all manner of complex timbres, and with or without ring modulation (which I haven't yet had a chance to mention). If you now access the Filter/Routing page, you'll be presented with a display that shows which sound is going where, with the colours of each synthesis block corresponding to choices made elsewhere in the system. Following the oscillators, the Quantum offers three filters in two blocks — a parallel 'dual analogue' filter block, and something called the Digital Former — and you can choose to place these in parallel, or feed the output from the analogue filters into the Former, or *vice versa*. Oscillators can then be injected into these in any proportion or directly into the amplifier further down the signal path.

Turning first to the analogue filters, you'll find the expected cutoff frequency and resonance controls for each of these on the top panel together with buttons labelled Type and Mode. The four filter Types are 12dB/oct and 24dB/oct low-pass filters with or without a light overdrive. The actions of the eight filter Modes — Single, Boost, Twin Peaks, Escaping, Opposition, Endless, Independent and Linked — are less obvious, and there isn't space to discuss each of them here, but it should be obvious that there are lots of possibilities. The filters will oscillate at high resonance, but when I tried to create a patch that used them as sound sources I discovered that they were all tuned to different pitches, which made it impossible to play them polyphonically. Filter calibration was introduced in OS v1.2.2, so hopefully this will sort things out. Unfortunately, the output from the analogue filter block is single-channel even when you feed it stereo signals from the oscillators, so it also includes a stereo simulator. It's not the same as retaining the panorama of the oscillators that you feed into it, but it can provide a pleasing spread.

Despite having just three controls on the control panel, the Digital Former offers 23 processing Types, including comb filters, a decimator, a wide selection of 12dB/oct and 24dB/oct low-pass, high-pass, band-pass and band-reject filters, overdrive and gain. Unfortunately, it's merely a single device operating upon the whole of the signal rather than on a per-note basis so, while you can do some interesting things with it, I now understand why Waldorf called it a Digital Former rather than a digital filter section.

The outputs from these blocks and any oscillators that bypass them then pass to the amplifier section. There are no physical controls for this, although one of the contour generators is labelled Amp and is hardwired to it even though it can be directed elsewhere if desired.

The signal is then passed to the five effects blocks. These lie in series and each offers the same selection of effects: phaser, chorus, flanger, delay, reverb, a four-band EQ, an overdrive and a compressor. Strangely, you can only have one instance of any given effect inserted at any given time. You can control the amount and the depth of the first three effects units using the physical controls on the top panel, while deeper programming can be undertaken using the screen. The range of parameters available for each effect type can be impressive, and you can design and save your own preset effects if desired. I was able to create effects chains that I would be happy to use and, to make them sound a bit more vintage, I could always add a bit of noise from an unused oscillator (if one was available) to emulate the swirly noise of analogue stompboxes. Interestingly, this was where I found my only bug in the Quantum's GUI; the Bypass and Presets buttons didn't appear on screen in the flanger, although if I tapped where they should have been, things worked correctly.

Finally, the signal reaches the output section, which boasts a global compressor plus a volume control that affects the levels at the main audio outputs and the headphones output, but not the auxiliary outputs. (See 'Rear Panel' box.) In the routing page, you can determine whether the post-effects sound is directed to the main stereo outputs, the auxiliary outputs or both, but there's no single-channel option; at its outputs, the Quantum is steadfastly stereo.

Modulation

The Quantum's DADSR contour generators (of which three are revealed on the control panel, while another three are accessible via the screen) are velocity sensitive and can be assigned in the modulation matrix. The maximum lengths of the attack, decay and release stages are of the order of one minute, which is excellent, and an EnvelopeVar parameter imitates the inconsistencies of analogue contour generation. Their attack stages can independently take one of three forms — exponential, linear and logarithmic — while each of their decay and release stages can take linear or two types of exponential forms. In

Quantum v2: Kernel Synthesis

The recently announced v2 firmware includes a new oscillator option called Kernel Synthesis. This will offer six kernels within each oscillator, with each kernel freely selectable from virtual analogue, wavetable, sample and noise sources. It will be possible to tune each kernel individually and each will also include a contour and a wide range of control options. It will then be possible to cross-modulate kernels, selecting from AM, FM, phase distortion (PD) and ring modulation options. The possibilities are immense but, unfortunately, this review was completed before the first beta version became available, so I was unable to try any of this.

Updating The Quantum

Despite its MIDI and USB connections, the method for updating the Quantum is a pain in the arse because, once you've downloaded the file from the Waldorf website, you have to copy it to an SD card to perform the update. I use a MacBook Pro which, amongst other terrible design decisions by Apple, has only USB-C slots, so using an SD card means buying an adaptor as well as a suitable card. Why not stream the upgrade from the computer or use USB memory sticks? I have no idea, but I view it as a poor decision by Waldorf.


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» addition, you can loop the contours, and a Single Trig mode allows you to recreate a paraphonic response. Another nice touch is how the contours are displayed on the screen, which shows each note as a dot moving along the curve as you play, although not always accurately! If I have a complaint (well... a suggestion), it would be nice if there were a control to switch the top-panel knobs between envelopes 1, 2 and 3 (ostensibly filter 1, filter 2, and amplifier) and envelopes 4, 5 and 6. But there isn't. Furthermore, I think that the Quantum deserves more powerful shaping capabilities: six-stage contours would at least allow me to create my favourite *sforzando* brass patches.

There are also six LFOs and, again, you can access three of these from the panel, while the full set can be programmed in depth from the screen. Each offers six waveforms, including S&H, and two additional parameters — Warp and Slew — allow you to mould these into a yet wider range of shapes. Their frequency ranges extend from one cycle every four minutes to 100Hz, and you can also synchronise them to clock. Each LFO also has an associated AR amplitude contour, so you can fade it in and out in pleasing ways. In addition, they can be locked together for global effects, or generated individually for each note played, whereupon they can be free-running or key-triggered with user-defined phase.

Next, we come to the Komplex Modulator. This generates two waveforms that you can select from the usual candidates or draw by dragging a finger across the screen. You can then blend them to create yet more complex curves. But that's not the end of it, because you can smooth the curves, warp the results, and even add jitter to extend things further. You can then determine the frequency and the depth of the resulting waveform, and an AR envelope allows you to fade it in and out. Three modes then allow you to apply the modulation globally, on a per-note basis, or as a one-shot curve in which it's nothing less than a 32-step contour generator that you can control in sophisticated ways. Yet more parameters allow you to determine the phase of the waveform, and whether it's key-sync'ed or free-running. I like the Komplex Modulator and was pleased to find that you can save and recall setups in the usual Quantum fashion.

All of which brings us to the modulation matrix. This offers 40 slots that can draw upon 43 sources that include all of the contour generators and LFOs, the Komplex Modulator, the wheels and any connected pedals, various MIDI CCs, the screen when used as an X/Y pad, the sequencer's modulation rows... and more, including polyphonic aftertouch over MIDI. (While the Quantum's keyboard generates channel pressure, you can

The Arpeggiator & Sequencer

The Quantum's arpeggiator offers seven patterns; a range up to four octaves; tempo that can be synchronised to the LFOs, the Komplex Modulator and the delay effects; user-defined gate length; a selection of seven sorting orders to determine the pattern; three velocity interpretations within the arpeggio; and 31 factory programmed rhythm and accent patterns. In addition to this, there's a five-row, 32-step sequencer, with the first row containing the note data and the other four carrying control sequences that you can use as modulation sources. There are also five playback modes and three reset modes, and you can of course adjust the bpm, the clock ratio, the swing and the gate length. This is all good stuff, but it would be nice if Waldorf updated the sequencer so that it could run without outputting notes, allowing it to be used as a complex modulator.

The Quantum is a substantial instrument, measuring 1006 x 401 x 131mm and weighing in at an impressive 17.8kg.

control the sound engine using PolyAT if you have a suitable controller. What's more, I understand that MPE is on the list for a future update, which should make owners of Seaboards and Continuums happy.) Your choice of destinations is even greater, with 158 options, and you have a choice of 43 controllers (again including polyphonic aftertouch) to control the amplitude of the modulation in each slot. If there's a limitation here, it's that there are only two destinations within each of the effects blocks — the wet/dry mix and the amount of the top-panel control parameter. It would be good to see this side of things developed further. Happily, a huge number of parameters will also respond to MIDI CCs, which means that you can automate the Quantum and, in addition to its MIDI Learn feature, there's a page that allows you to assign your own choice of parameters to MIDI CCs 0 and 2 to 120. You can save these maps for later recall, so you can even configure the synth for different projects, studios or whatever.

Layers

So that's it... except that it isn't. Within the limitations of its eight-note polyphony, the Quantum offers two such synthesizers (Layer 1 and Layer 2) that you can split or layer across the keyboard, and the split mode allows you to overlap Layers so that you can have, in effect, three zones. There's also a unison mode available for either Layer so you can allocate, say, five notes for a pad under your left hand while playing a huge three-voice unison lead with your right. More advanced uses allow you to direct the audio from the external signal inputs to either or both of the Layers for granular synthesis of incoming signals, or via the filters and effects if you want to process the audio in more conventional fashion. You can also use the Layers to create true stereo sounds by layering two appropriate patches and panning these left and right. However, this drops the polyphony to just four notes, so may not be worth the effort. Best of all, the two Layers can respond to different MIDI channels, they have independent effects sections, and you can direct them to separate output pairs, which makes the Quantum truly bi-timbral.

In Use

The Quantum is big, heavy, bold... and surprisingly stylish. It offers firm knobs, positive buttons, clear graphics, and a 61-note semi-weighted keyboard that — while not top of the range — is nonetheless pleasant to play. It also boots quickly, which is more important than you »

The Performance Panel

To the left of the keyboard you'll find the performance panel, which is dominated by a pitch-bend wheel that you can assign to each oscillator individually, and a modulation wheel that you can assign to any parameter in the modulation matrix. Behind these, you'll find six buttons. These include octave up/down (which allow you to transpose the keyboard by ±2 octaves), the latch and chord buttons (which can hold the current notes with or without the arpeggiator playing), and the mono button that, depending upon contour settings, can provide single- and multi-triggered operation for monosynth duties.

» might think. Unfortunately, two minor faults were apparent on taking it out of its box: the Effect 1 Amount LED had been pushed a small way into the case (possibly by a previous reviewer or user) and the screen had a flaw that revealed itself as a blotch in the display. Happily, neither of these affected its operation.

The other thing that the Quantum is, is deep: there's been no opportunity here to go into detail (and there's a lot of detail!) and it has myriad other facilities that I've had no room to discuss. For example, there's its capacious internal memory, its definable knob responses and colour schemes, its waveform displays, oscilloscope and spectrum analysers, its multiple editing modes, its multiple tuning scales, and much more. As a consequence of this, I think that I may have spent more time learning the Quantum before starting to type than any other instrument since the Korg OASYS. But, once I had grasped it, everything soon fell to hand. Sure, Waldorf's programmers didn't get everything right — for example, it's far too easy to hit the wrong option in many of the lists, especially when scrolling — but in general I found myself performing all of my programming via the module selection buttons, screen and editing system rather than using the equivalent knobs.

Sound-wise, it's hard to summarise something that combines the souls of a PPG, a virtual analogue synth, a dedicated sampler, a granular synth and a physical modelling synth... but it's not impossible. If I had to characterise the Quantum, I would say that it sounds 'European'; clean and precise whether it's producing sounds that are simple or deep or complex. This may or may not be to your taste, but I like it because it allows the Quantum to sit in a mix without drawing attention to itself when you don't want it to. I also have to compliment Waldorf on the Quantum's lack of aliasing. I had to work hard to try to create any, and it never interfered with any musical sounds that I created. Nonetheless, I did discover some pitch instability at high frequencies, most easily heard when using the ring modulator. I checked that nothing in my patches was creating this, but couldn't find anything to explain it. Would this matter in the real world? Probably not... I was pushing the synth beyond reasonable extremes. It's interesting, but nothing to worry about.

So, where does the Quantum go from here? We already know about the next oscillator model (see 'Kernel Synthesis' box) and Waldorf have admitted that there are a number of other updates in the pipeline. This is good, because I can think of a few things that could be improved. For example, I would like to see the scrolling of long lists improved because you can't flick a long list nor can you use the knobs to scroll quickly. Then there are the six sets of 20 'favourite' patches for live performance. These should be good news, but a given sound can reside in only one location in any given set, which precludes stepping through a set sequentially unless you save multiple copies of any patch that you need more than once. I also dislike the way that the arpeggio, latch and mono buttons interact. Oh yes, and I hope the company will add a screensaver. Waldorf claim that the screen won't burn but a screensaver would be a source of reassurance if nothing else. I also think that this is a synth that would have benefitted from a ribbon controller, but that's not something that can be added in a firmware update.

Unfortunately, the review unit suffered from a problem that I gather has already been identified elsewhere. I connected it to my MacBook Pro via USB and recorded a short MIDI piece. All seemed fine until I tried to replay this and play along with it. After a few seconds the Quantum started glitching and creating very loud high-pitched notes. It then crashed. Pressing the panic buttons did nothing, and even turning the Master Volume control to zero did nothing to abate the noise — only a power cycle restored sanity. Further tests showed that the problem is related to the use of the USB interface and, until Waldorf fix this, the Quantum will be unusable by some potential purchasers. Hopefully, it will have been resolved by the time that you read this.

Conclusions

Very soon we'll have at least three flagship analogue/digital hybrid polysynths from which to choose — the largely analogue Moog One, the largely digital Waldorf Quantum, and the Prophet X, which sits somewhere between the two. All three are mighty powerful beasties that nod toward earlier synthesizers, but I'm not going to try to compare them because, despite superficial similarities, I view them as very different. If you are wealthy and suffer from unrelenting gear lust, you could make an argument for owning all three. If you have to choose just one, and if we assume that the sound quality of each (which is excellent in all three cases) is equally appealing, then your choice will possibly be determined by the balance between immediacy and flexibility. If you lean toward the latter, the Quantum is undoubtedly the winner, and I can't see anyone exhausting its capabilities. And, to answer the question that I posed at the start of this review, it's the most 'finished' flagship polysynth that Waldorf have ever released; there's a delightful absence of messages to tell you of features not yet implemented. When the USB/MIDI problem is fixed it will be a superb sound design and performance tool. Lest you be under any illusions, I like it a lot.

> "If I had to characterise the Quantum, I would say that it sounds 'European'; clean and precise whether it's producing sounds that are simple or deep or complex."

$ $4299
W www.waldorfmusic.com

Steinberg Groove Agent 5

Virtual Drummer Instrument

How do you improve on the ultimate drum studio? Steinberg attempt just that with the latest version of Groove Agent.

JOHN WALDEN

Back in 2015 Steinberg tagged Groove Agent 4 as the 'ultimate drum studio'. GA4 was certainly a very significant change from previous GA-badged virtual instruments. With a very deep feature set, and multiple 'Agents' covering electronic drums, percussion and acoustic drums available within a single instance, it certainly staked a pretty good claim for the 'best all-round drum tool', combining elements found in the likes of virtual drummer products such as Superior Drummer/EZ Drummer and virtual MPC-style drum machines.

So, if GA4 was the 'ultimate drum studio', where do you go from there? With Groove Agent 5 now with us, what have Steinberg done to take their flagship drum tool to an 'even more ultimate' status?

Status Quo

With software as deep as Groove Agent, any review is only likely to scratch the surface of what's possible. The focus here will obviously fall on the new and improved features in the v5 release. However, the core functionality found in v4 — and summarised in the June 2015 *SOS* review — remains intact. In brief, that means you get three different drum Agents — Acoustic, Beat and Percussion — any combination of which can be loaded into the four Agent slots in a single instance of Groove Agent. As well as a substantial collection of sounds, there is an excellent set of 'styles' (collections of preset MIDI patterns) in various musical genres, and these can also be assigned to a pad system for triggering. There is also a very neat Style Player system for easy triggering and performance creation as well as a comprehensive MIDI pattern editing environment. If you want additional content, Steinberg have a series of genre-based expansion packs for sounds,

Steinberg Groove Agent 5 $179

PROS

- An incredibly rich feature set for drum production.
- Some great new content including The Kit.
- New live sampling feature simplifies DIY instrument creation.

CONS

- User interface is pretty intense.
- Initial learning curve to be faced.

SUMMARY

If you are looking for your first serious drum production tool, and want a well-specified, all-in-one solution, Groove Agent 5 makes a great candidate.

As well as acoustic and electronic drums, GA5 also includes percussion instruments via the Percussion Agent.

patterns and styles.

For the Acoustic and Percussion Agents, the user gets plenty of mix options for tweaking the sounds (although not direct access to editing the samples themselves), while the Beat Agent provides very detailed sample editing, support for velocity layering, sample import and excellent options for slicing audio loops. You can, therefore, build your own kits — acoustic or electronic — within Beat Agent. Multiple audio outputs are also supported. In short, for users looking for a single solution for all their drum needs, GA4 was already a good bet and, in retaining that core functionality, GA5 remains so.

The Bigger Picture

The overall look of GA hasn't really changed. That's great for existing users familiar with the interface but, equally, means that it can still be somewhat daunting on first encounter. However, GA5 brings two significant UI changes that, in different ways, help on this front. First, the UI is now fully resizable, whether used stand-alone or as a plug-in. While this doesn't do much to counter the 'compact' nature of some of the controls and buttons, it does mean that you can, for example, make more room for detailed sample editing or for the pattern editor.

Second, the UI now offers an integrated browser (called the Load Panel) which can be docked or floated. It does take a little while to get your head around what's possible, but the new Load Panel is both flexible and powerful. The ease with which you can audition, for example, a number of different kick or snare sounds in place, while your project is in playback, is incredibly useful.

Groovy New Content

Of course, the first thing most potential new users or upgraders might look for is new sonic content. GA5 certainly ticks that box and the highlight is a new acoustic drum kit simply called The Kit. This is based around a combination of Pearl and Yamaha drums and predominantly Zildjian cymbals. It was sampled with high-end equipment in an equally high-end room. GA5's increased velocity layer limit is made good use of (some sounds have up to 20 velocity layers) and a good collection of presets configure GA5's mixing options to deliver a wide range of acoustic drum styles. It sounds very good indeed; if you want a great-sounding acoustic kit with a minimum of fuss, The Kit is just the job.

That's not it, of course. There are also some 30 new sample sets for the Beat Agent, many supplied by up-and-coming electronic music producers, and a range of new pattern collections. There is plenty here for budding EDM producers to get their teeth into and many of the kits include loops, bass or melody parts plus various sound effects and hits.

Beats From (Under) The Hood

GA5 also brings some new features under the hood. The number of available stereo outputs has been increased from 16 to 32. Users who regularly operate multiple Agents will appreciate the extra flexibility when routing their GA sounds into their host DAW's mixer. Perhaps of more general significance is the increase in the number of supported velocity layers from 8 to 32. No, it might not match the technical specification found in the very top-end of the virtual drummer world but, as demonstrated by The Kit, it is going to be more than enough to satisfy all but the geekiest of drum sound geeks.

The Sample Editor tab itself has some useful new options. When auditioning a multi-layered pad, you can now solo a single layer. In addition, when editing the pitch, filter and amplitude envelopes, you can now toggle on a waveform display for the sample being edited underneath the envelope itself. As you make edits to the envelope, the waveform display automatically adjusts, providing some very useful visual feedback.

It remains true that all the detailed sample editing is only available within Beat Agent and you still can't edit Acoustic Agent or Percussion Agent kits at the sample level. Both of these Agents do have very useful mixer and performance »

Groove Agent's excellent Style Player can now be used with Beat Agent as well as the Acoustic Agent.

options, though, so, whatever selection of Steinberg's sample content you have available for these Agents, you can still coax a huge range of sounds from them.

And talking of the mixer options, for the Acoustic Agent and Percussion Agent, Cubase users now get the option to export their GA5 mixer settings, with or without all the plug-in settings, to equivalent channels in the Cubase MixConsole. If you prefer to keep all your mixing tasks in one place, this is very useful and, in my own testing, worked a treat. Unfortunately, this export feature does not, as yet, apply to Beat Agent. Here's hoping that is something that Steinberg have on their 'to do' list. However, one other very welcome change is that the excellent Style Player — previously only available with the Acoustic Agent — now also works with the Beat Agent including, of course, with all the new Style content.

Splitting Snares (And Other Drums)

Beat Agent's Instrument Edit page's sample editing has two further interesting new features. The Decompose tab allows you to split samples into their pitched and noise components. The Sensitivity, Cutoff and Duration knobs provide you with some control over how the two components are separated and, if you both enable the Pre-listen button and use the Solo buttons for the Tonal and Noise components, you can audition the two elements of the sound as you adjust these controls.

Clicking the Apply button will then replace the original sample with two separate samples or, if you also engage the Mix button, a single sample with whatever volume balance is set. The Options button gives you control over where any new samples are stored. If you create two samples — a tonal and a noise one — you can then engage the SEL button (located top-right above the waveform display) and that lets you apply further edits to just the selected sample. This means you could, for example, apply a pitch envelope to the tonal component but leave the noise component unaffected. By default, the two samples are mapped ready for velocity-based triggering but you can, of course, switch a pad's triggering to Layer mode (via the Main tab accessed via the button top-left above the waveform display), and that mode simply plays all of the samples mapped to a pad when the pad is triggered, regardless of their velocity mapping.

If you are into sculpting your own drum sounds, or into more general, sample-based, sound design, there is a lot of potential here. Indeed, there is also scope for creating some cool 'playable' pitched instruments by copying tonal and noise samples to multiple pads and simply adjusting the pitch of the tonal elements (leaving the noise element unchanged) on each pad. Or simply creating some downright weird noises... the choice is yours.

Beat Agent's Sample Editor now provides a waveform display while editing the pitch, filter and amp envelopes.

Roll Your Own

And talking of creating your own instruments, the second new element within the Instrument Edit page's options is the Recorder tab. This essentially allows you to record samples live into GA5 and automatically assign those samples to pads in various ways. This is derived from the same feature set in Steinberg's dedicated sample instrument, Halion.

The sampling in GA5 is surprisingly comprehensive, though, and there are two particularly cool elements. First, you can live sample from either an audio input or from a sound source within your DAW. The latter can be an audio track or another virtual instrument and this is made possible via GA5's side-chain audio input.

Second, the Recorder tab provides enough features to automate the sample recording process so that capturing multiple samples to build a complete instrument — whether various pitch or velocity layers — is a streamlined experience. For example, I fed the audio output from Superior Drummer 3 into GA5's side-chain input, set the recording process to use a threshold level for starting and stopping the recording of each sample, and set the mapping mode to Fixed (all samples end up velocity mapped on the same pad; a chromatic mapping option is also included for pitched instruments). I then programmed a series of 30 kick hits in SD3, starting at low velocities and gradually increasing. When I enabled recording in GA5, and set my programmed hits to playback, GA5 simply captured each hit and velocity mapped them for me to the chosen GA5 pad. You could, of course, do exactly the same thing with any virtual instrument, recorded sample, or from a 'real' instrument such as an acoustic drum kit or synth. This new feature is super-simple to use and very effective.

Conclusion

Groove Agent is undoubtedly a very powerful drum production environment, and this latest release simply adds to what was already a very rich feature set. For electronic drum production, GA5 is competitive with the obvious alternatives (for example, NI's Battery 4) and, with The Kit, can also provide high-quality acoustic drums for those that are not yet ready for something like Superior Drummer 3.

Existing users, providing they can justify the price of the upgrade, will undoubtedly find much to like in all the new highlight features. That said, Groove Agent does remain a complex beast and, while it is competitively priced, new users should go in with their eyes open; there is a learning curve to be tackled. However, for those searching for a powerful, comprehensive, all-in-one drum solution, GA5 is certainly a step worth taking. And if you want to dip your toe into the GA5 waters, Steinberg have very sensibly made a free 30-day trial version available for download. Well worth a look provided you are prepared to invest some time mastering the comprehensive suite of options.

$ Groove Agent 5 $179.99; upgrades from $99.99.
W www.steinberg.net

JONES-SCANLON
STUDIO MONITORS

~ Krish Sharma (Rolling Stones Recording Engineer)

"I first used these speakers with the Rolling Stones, there is no more demanding a situation for a speaker than reproducing a Stones tracking session at volume. I usually would only do a tracking playback for these players on the large studio main speakers, but I happened to have the Jones-Scanlon's on as the band and producer Don Was walked in and as they liked the sound I continued playback. We were so pleased that when we moved to another studio we kidnapped the speakers and brought them with us!"

"The Jones-Scanlon Studio Monitors are hands down my new favorite monitors. They are unicorns in that they not only are accurate both for frequency and transient response but they have punch and headroom to communicate raw takes with power. While being amazingly accurate, they do not sacrifice musicality."

"Mixes come together very easily on the Jones-Scanlon Studio Monitors. I find myself needing to switch between speakers much less and I am able to balance very quickly because the speakers are accurate and musical at all volume levels. Working with the Rolling Stones, I need a speaker that can reproduce dense arrangements of stringed instruments accurately and without coloration and these achieve that goal effortlessly."

"The Jones-Scanlon Studio Monitors have the detail and accuracy for low volume mixing and also the power to rock the house with musicality. They are a great studio solution and accomplish multiple roles without compromise."

"After hearing these speakers quite by accident, I knew I had to have them. I tracked down the manufacturer unsolicited."

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

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Arturia KeyLab MkII

Controller Keyboard

Arturia take their KeyLab range to the next level.

SIMON SHERBOURNE

KeyLab is the family name of Arturia's range of master keyboards that combine stand-alone MIDI controller functionality with DAW control. The keyboards all also serve as an integrated hardware front-end for Arturia's Analog Lab: the company's soft synth that consolidates the mighty V Collection into a workstation instrument.

Back in February 2018 we reviewed the KeyLab Essential. The KeyLab MkII comes in as pro or flagship big brother to the Essential, bringing the whole range up to date. The weighted 88-key version is still available with the older first-gen design.

Key Features

There's a strong family resemblance between the KeyLab MkII and Essential, but the MkII is significantly deeper (in fact deeper than most MIDI keyboards) to allow for a 4x4 pad grid and longer faders. The MkII is a much sturdier and heavier construction, with a metal main case. The buttons are solid and clicky instead of soft and wobbly. The higher-spec metallic wheels appear above or next to the keys depending on whether you have the 49- or 61-key version.

The keybed is the same 'Pro-Feel' Arturia mechanism with aftertouch as found on the MatrixBrute and MiniBrute 2. It has a light, fairly shallow action, like the Essential, but with less wobbliness. I like the low-profile pads, whose multicoloured backlights indicate pressure with brightness. The main data knob and screen area appears much the same as on the Essential, but is enhanced by three dedicated buttons for switching between Analog Lab, DAW and User modes. The right-hand zone sports nine strips of sliders, rotary encoders and buttons.

Connectivity is a strong point. Power is via the USB port, although there's an optional DC adaptor. There are no fewer than five pedal/expression control inputs on quarter-inch jacks. Traditional MIDI is connected via a pair of full-size DIN ports. There's also CV connectivity comprising a single CV input, and pitch, gate, and dual mod outputs.

Quick Start

Unlike a number of its high-end peers, the KeyLab can function without needing a computer to think for it, but in most cases it's probably going to find itself as part of a DAW-centred studio. It's a class-compliant USB device, so should start making itself useful as soon as it's plugged in. However, to take full advantage of the KeyLab MkII you'll want to install the software that comes with it, and get it set up correctly with your DAW of choice.

Software download, install and licensing is all handled smoothly by the dedicated Arturia Software Center application. With the KeyLab MkII registered on your Arturia account you'll get access to Analog Lab 3, Piano V, and the MIDI Control Center. Analog Lab installs as a stand-alone app as well as a plug-in in the AU, AAX, and VST formats.

No special codecs or packages are required for DAW setup. Up-to-date versions of Ableton Live have the necessary script built in, and all other DAWs communicate with the KeyLab via Mackie Control or HUI. You'll need to select the correct DAW map in the MIDI Control Center utility and follow the instructions for setting the prefs in your DAW.

DAW Control

With everything set up the MkII can be easily re-focused between controlling

On the back panel we find CV outputs, MIDI I/O, three quarter-inch aux control ports, expression and sustain pedal inputs, a CV input, and a USB B port.

Analog Lab, your DAW, and your own custom assignments. DAW mode is where the MkII behaves most like a generic MIDI keyboard, ready to play any instrument plug-in or MIDI synth attached to your DAW tracks.

DAW mode also provides transport controls, and buttons for common functions like Save, Loop mode and Click, etc. In this mode the sliders map to faders and pan in your mixer. The buttons below the faders act as direct track selectors, or you can use the left and right cursors on either side of the main data knob to nudge the focus between tracks.

Selecting a track targets it for the five Track Control buttons in the centre of the panel. These access Solo, Mute, Rec Arm, and automation mode controls. For the most part this is a better use of space than providing per-track buttons or banked layers for these functions as on many controllers. It does mean, though, that you can't quickly mute/unmute multiple tracks.

In most DAWs selecting a track also targets it for MIDI input. In Ableton Live, however, Rec Arm does not follow Select, meaning that to move MIDI input focus between tracks is a two-stage process of selecting the track, then hitting the Rec button in the Track Controls. This is a common issue with Live controllers (it was the same with the Novation SL MkIII we looked at recently). It is possible to have linked controls, as evidenced by Push and the Native Instruments controllers, but it seems many manufacturers overlook this in their scripts.

The Live integration would also be improved if the transport record was mapped to Session record instead of Arrange record (at least as an option). A way to bank the pads up and down in Drum Racks would be welcome too; as it stands you can only access the first 16 slots.

Live enjoys some extra functionality in the Session view, with a clip-launching mode available on the pads. The Main data wheel scrolls the Scene focus (although you can't launch a whole Scene), and the horizontal position can be banked with buttons, or by selecting a track. I did find the layout confusing, though: the pads show two clip rows from eight tracks, but stacked into two chunks because of the 4x4 pad grid.

Actual Sounds & Voices

In Analog Lab mode the KeyLab MkII melds with Arturia's VI master plug-in to become a hybrid workstation. Analog Lab acts as a unified front-end for the entire V Collection. It provides a tagged patch library, a shell for creating dual-layer splits, and a way to store patch playlists that can be used when playing live.

The beauty of Analog Lab is that it gives you access to the sounds of the V Collection, without actually buying it. However, if you do buy the Collection, or any of the individual instruments, it unlocks the full GUIs within Analog Lab. Apparently browsing and control maps are also implemented for Arturia's new Pigments synth.

The central display and data wheel can be used for patch browsing and selection. The single Category button tabs you through the hierarchy of the library, starting with the top-level groupings of Synth, Piano, Organs, and Multis, then drilling down to Instrument Types, Styles, then source banks. When you get to the directory you want, you can tap the Preset button to redirect the wheel to browsing actual patches.

As noted with the KeyLab Essential, it's a pretty limited set of controls and visual feedback to work with, but does work fairly well. It helps of course to have the plug-in open to reference on screen. For live work, if you keep a single Playlist open to choose from you'll be fine. A notable advantage over the Essential is the row of buttons below the faders, which take »

Arturia KeyLab MkII
From $449

PROS
- Simple and fast to use.
- Tight integration with the included Analog Lab 3.
- DAW controls.
- Lots of connectivity.
- Flexible stand-alone capabilities.

CONS
- Just one small display.
- No Scale features.

SUMMARY
The KeyLab MkII is a solid, customisable master keyboard and DAW controller, as well as the perfect companion to the V Collection plug-ins.

The KeyLab MkII can browse and control the V Collection sounds in the included Analog Lab plug-in.

» you instantly between the Instrument Type categories. Unfortunately these immediately load the first patch in these categories; I'd prefer that it just called up the list on the data wheel.

Each patch in Analog Lab has two pages of pre-assigned controls for the encoders/sliders section, plus a third where you can build layered macros to control the other assignments and the internal Part mixer. The rotaries are continuous, so can pick up control from any point. While the sliders can be switched between Jump or Pick-up modes for DAW control, they can also Scale in Analog Lab for smooth takeovers. Their current positions are shown as ghost faders on the plug-in when the values are not in sync.

Alternatives

The MIDI controller keyboard market is probably the most competitive space in music tech hardware. There is a ton of controllers out there with buttons, knobs, pads and faders. The KeyLab MkII's distinguishes itself from cheaper, basic controllers with its build quality, endless encoders, excellent programmability and of course the inclusion of and integration with Analog Lab. Close by in the bang-per-buck curve are **Akai**'s **MPK** and **Nektar**'s **Panaroma**. The KeyLab's list price does put it within £100 of **Novation**'s new **SL MkIII**, which offers multiple parts, sequencing, scales and displays, although is more complex and doesn't have an equivalent to Analog Lab. And of course you can never dismiss the **NI** keyboards which, like the KeyLab, offer a companion sound bank and excellent DAW integration.

There's a consistent mapping system used, to the extent that the most common assignments are hard printed on the panel. For example, the encoder mapping generally starts: Filter Cutoff, Resonance, LFO Depth, LFO Rate. The first bank of slider assignments is usually dedicated to envelope controls. This helps a lot given that there are no displays on the hardware to guide you.

Custom MIDI & CV

The KeyLab MkII provides 10 slots for saving and recalling your own MIDI control assignments. These are set up and managed in the MIDI Control Center software, or you can edit them directly on the hardware. There's a sophisticated degree of choice about how controls are configured. Buttons can be toggled or momentary, rotaries can be absolute or relative and work with different acceleration/gearing. You can limit CC control ranges. The pads can be colour-coded, and in addition to note, CC, or Program Change messages they can be set to recall your other user templates.

The Control Center is also where you configure the various foot controller inputs and the CV connections. The three Aux pedal inputs can be assigned to any CC value in either continuous or switched modes. They can also be used for Program Changes.

CV integration in MIDI controllers is becoming more common. Arturia certainly have good form here with the KeyStep, and the KeyLab MkII benefits from pro options like variable voltage ranges, and the option to choose which MIDI note maps to 0V. The note source can be taken from either keyboard part in a split config.

The Mod sources are directly assigned to specific buttons, faders, rotaries or wheels, or can be derived from velocity, aftertouch, or any of the pedal inputs. This keeps things simple and can be set up from the keyboard. Currently the CV outputs can't be accessed from your DAW, but apparently this will come later.

Conclusion

The MkII KeyLab is a solid design and layout update, and adds dedicated DAW control functionality and CV connectivity. It's refreshingly straightforward to set up and use, both as a general controller and as a sound source with Analog Lab. Analog Lab is of course one of the major selling points, offering a wealth of sounds, albeit completely focused on synths and keys.

Having come straight from reviewing the Novation SL MkIII, I can't help thinking that Arturia have missed a trick by not adding some stand-alone sequencing functionality from the KeyStep, especially as there are CV connections. I guess that there will be a 'KeyStep Pro' at some point to scratch that itch. However, as a versatile master keyboard for both studio and stage with elements of both a traditional controller and hybrid instrument, the KeyLab MkII has a lot going for it. ■■■

Chords Back In Fashion

The KeyLab has several chord assist features. The most simple is accessed by holding the Chord button while playing a chord. This stores the chord in memory, and you can then play the chord from single keys. A more sophisticated option is to activate the Chord Transpose mode on the trigger pads. This stores chord shapes on the pads, which are also played from single keys on the keyboard. Finally there's the Chord Memory mode on the pads, which triggers playback of specific (non-transposable) chords from the pads, leaving the main keys free to play over the top. All the modes are useful. What the KeyLab does lack compared to some of its peers is a Scale constraining feature. This means that chords always transpose to chromatic intervals and won't automatically stay in a particular key.

$ KeyLab MkII 49-note $449, 61-note $499.
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MATTY MOON

The last decade has seen a boom in boutique microphone manufacturing, which is quite exciting for us recording engineers. Online shopping, an abundance of social media platforms and cheaper production costs have enabled many talented electrical engineers and designers to create products and easily sell them to the masses. Never before has there been such a wealth of different microphone options available at all price levels.

Founded in 2015, in Nashville, Tennessee, Luke Audio are fairly new kids on the block. Allen Luke and his team began by creating emulations of classic microphones. The company initially manufactured the AL-X751, based on the Telefunken ELAM 251; the AL-X767, emulating the Neumann U67; the AL-X747, based on the Neumann U47; and the AL-X712, based on the AKG C12. Their latest offering is a large-diaphragm valve capacitor model that tries to combine these sonic flavours into one microphone by including three interchangeable capsules, at a highly affordable price.

What's In The Box?

The AL-Y56 comes nicely flightcased with everything you need, including the power supply, a good-quality cradle, foam windshield, IEC lead and seven-pin braided microphone cable. The microphone and additional capsules are enclosed in elegant dark wood boxes with sliding lids, and everything fits neatly into carved slots within the flightcase. You definitely can't dispute that you get a lot for your money here. It's maybe slightly over-engineered, with its boxes within boxes, but everything feels well manufactured.

The electronics are hand-wired and the mic body is fitted with an EHX 6027A tube and an 11.5:1 output transformer. Depending on which capsule is attached, the microphone is claimed to accommodate a maximum SPL of 136-140 dB, presumably with the optional 10dB attenuator switched in. This, for some reason, is located on the circuit board within the microphone casing. The base of the microphone has to be unscrewed and the microphone disassembled to access the switch, which is not at all ideal. That said, whilst testing the mic I found no need to engage this pad even on relatively high-SPL sources.

If The Capsule Fits

The three interchangeable 'pop-top' capsules have fixed cardioid pickup patterns and are custom, in-house

Luke Audio AL-Y56 **$999**

PROS

- You get a lot for your money.
- Great-sounding mic with three distinct-sounding capsules.
- Quick and easy to change capsules.
- Very reasonably priced.

CONS

- Output level varies by 2-3 dB across different capsules.
- 10dB attenuator switch located inside the microphone casing.

SUMMARY

A well-built, affordable and great-sounding valve mic with a modular capsule system that gives you three distinct sonic flavours.

Luke Audio AL-Y56

Valve Microphone & Capsule Set

Luke Audio's new valve microphone provides three classic capsule options at a highly affordable price. Is it too good to miss, or too good to be true?

reproductions of the CK12, K67 and K47, as found in vintage microphones such as the AKG C12/ELAM 251, Neumann U67 and U47 respectively. However, there is an interesting difference between Luke Audio's capsules and the originals, in that these all use a 1-micron, platinum-sputtered diaphragm rather than the more conventional 3- to 6-micron gold-sputtered affair.

The overall design and concept looks somewhat similar to the Blue Bottle Rocket, which also has interchangeable capsules. Blue offer a larger range of 10 different capsules, but the AL-Y56 and its three capsules come in at about a quarter of the price of Blue's flagship model.

In Use

During recording sessions, I always strive to fully explore the microphone options available. In reality, though, auditioning multiple mics on a source can be time-consuming. Time is always tight and it's tempting to go with what you know. This is where the AL-Y56's 'hot-swapping' capability comes into its own. The capsules are fixed to the microphone body using a bayonet-style mount; a light push and twist quickly releases the capsule, allowing for swift change-overs. The ability to put up a single microphone for a vocal, for instance, and quickly record a verse with each capsule, is fantastic.

Luke Audio briefly describe the sound character of each capsule thus: the A1 (CK12) is said to have "a bright, airy, open feel", the A2 (K67) "a more mid-focused and articulate sound" and the A3 (K47) "a full-bodied, rounded and warm sound with a true vintage feel".

Trying the mic over a few recording sessions, I discovered that the ability to quickly change the capsule was great for keeping things flowing. The ease with which you can go back and forth meant that it never felt laboured, or that the session was being held up. I also found myself enjoying auditioning each capsule. They are quite different-sounding: the A1 has a nice 'roundness' in the low end, with a slight boost at about 100Hz. It is also bright, with a presence lift in the upper mids and highs, and captures transients well. The capsule does have an 'airy' quality and I found the slight upper lift was great for bringing a source forward in a track without any extra EQ. The A2 has a softer, slightly mellower and more natural sound compared to the A1. The low end is also a touch tighter. The A3 is even less bright again; it seems much more mid-range focused, with an emphasis around 200-500 Hz, a slight attenuation from 100Hz down and a dip in the upper mids and highs compared to the other two capsules. One thing to note here is the slight variation in output level of each capsule. The A1 and A3 seemed fairly matched, but the A2 was around 2-3 dB quieter, and whilst this isn't a major issue, it can mean you need to perform minor preamp adjustments when switching between capsules.

During a drum session I chose to put the AL-Y56 up as a mono room mic. After listening to the drums in the room I placed the mic three or four feet in front of the kit, slightly off to the left (floor tom) side. It was about a foot from the floor and slightly angled down to get a little less direct cymbal sound and to pick up »

The AL-Y56 ships in a large flightcase, along with a range of quality accessories.

some reflections from the wooden floor. I wanted to capture a full kit sound with lots of low-frequency energy that would most probably be heavily compressed or saturated in the mix to add energy and bring out the character of the room. I popped on the A1 capsule first and straight away I was struck by how great the mic sounded. However, although it was crisp and bright, it had a little too much transient information for this purpose, and the slightly pronounced upper mids also meant the hi-hats and cymbals would quickly dominate the sound when heavy compression was applied. In the same role, the A2 capsule sounded lovely. There was a nice bloom to the bottom end, but without making the drums sound too tubby, and this capsule's mellower, less bright sound meant the cymbals were pushed back slightly. The A3 sounded almost dull in comparison. It was very mid-range focused, which can be a great characteristic for drum rooms, but it sounded muddy here and would have needed some low mids notched out to make it sit in a mix.

On the same session I thought I'd see how the AL-Y56 fared on bass guitar, setting it up about a foot back from an Ampeg 8x10 cab, pointing at the edge of one of the speaker cones. I was eager to see how the three flavours of capsule could be used to sculpt the bass tone and how each sat with the kick drum. The A1 sounded incredible. The low end was full and the broad boost from around 2kHz up really helped the note articulation. This capsule would be great if you wanted the bass right up front, driving the track. The A2 was really pleasing to my ears for this particular track: the low end instantly sat well with the kick drum and there was still just enough 'zing' to help the upper harmonics cut through without dominating the mids. The A3 definitely had a vintage vibe here. It gave the bass a throaty, mid-rangey sound, which would sit well if you had a lot of low end on the kick drum and needed the bass to sit just above it.

I also took the opportunity to try the mic on a rhythm acoustic guitar part that needed to sit in a busy track, under the lead vocal. Positioning the AL-Y56 about a foot away from the guitar with the capsule pointing between the 12th fret and the soundhole, the A1 sounded full and bright with a nice attack. The low end was also tight and controlled. The A2 gave the acoustic a lovely soft tone and made it sit back in the mix.

On electric, I think the A2 would be of great use when stacking thick, double-tracked distorted rhythm guitars, where you have a lead melody guitar that needs to sit on top and cut through. The A3 had a 'chewy' character that I have to admit I didn't like.

Finally, I was particularly keen to try this microphone on a vocal session. When tracking vocals, especially when working with a vocalist for the first time, I will set up a few different microphone options to find something that complements their voice. It can be a slightly lengthy process getting three different — and often large — microphones to play nicely together behind a pop shield, as well as patching each mic into the same preamplifier/compressor chain and then level-matching them. The AL-Y56 made this easy, and within a matter of minutes I had the microphone up and we were soundchecking the A1 capsule. The bright, transient characteristics really suited this particular vocalist and I found that it took compression well. The lift in the upper mids and highs helped the vocal sit up-front in the track, without sounding overly sibilant or harsh. I found that the A2 was perfect for backing vocals here; its having less presence meant the vocal sat behind the lead without me having to EQ. The A3 sounded a little too dark for this track. However, it saturates nicely and does give that '60s vocal sound, which would work great on the right song and production.

Hear For Yourself

You can download audio examples of all three capsules in action, on acoustic guitar, electric bass, drum kit and vocals, at the online version of this article: http://sosm.ag/lukeaudio-0419.htm.

Alternatives

The **Blue Bottle Mic Locker** is probably the closest alternative, and it ships with four capsules rather than three, but is several times more expensive.

Conclusions

This is a well-manufactured and quality-sounding microphone that is great value for money. Sonically it stands up well against far more expensive microphones. The 'pop-top' design and ease with which capsules are changed makes auditioning really quick, and the variation in character of each capsule makes for a highly versatile microphone with a broad sonic palette. I found that the A1 and A2 capsules sounded most instantly pleasing in tests, but I have no doubt that if I'd had the time to get more intimately acquainted with it, the A3 would find itself in use a lot too.

$ $999
T Wave Distro +1 973 728 2425
E sales@wavedistro.com
W www.wavedistro.com
W www.lukeaudiollc.com

JST Billy Decker Bus Glue

Plug-in Suite

Billy Decker's signature plug-in collection is aimed at those who'd rather get things done than get into the details.

SAM INGLIS

Billy Decker is a Nashville-based mix specialist who's equally at home balancing modern country music and extreme metal. What's most remarkable about Decker, though, is the speed at which he works. As he explained in *SOS* January 2018 (www.soundonsound.com/techniques/inside-track-chris-young-losing-sleep), it typically takes him an astonishing 45 minutes to turn a raw tracking session into a radio-friendly hit single.

This incredible efficiency is made possible by working entirely in the box and, in particular, by the extensive use of templates. Decker has a huge Pro Tools session ready to roll, containing the plug-in chains and drum samples he's honed over many years at the top of his game. So refined is his system that he can get close to a finished mix simply by bringing the target audio into this session — and now Decker has teamed up with Joey Sturgis Tones to make some aspects of his template available to the world at large. These are not simply snapshots or models of the existing plug-ins that Decker uses,

Joey Sturgis Tones
Billy Decker Bus Glue
$199

PROS

- Very simple to use.
- Targeted feature set helps you to get polished results fast.

CONS

- Huge installer for such an apparently simple product.
- Some controls are gain-compensated but others aren't.

SUMMARY

A suite of instant 'make it sound better' plug-ins targeted at the master bus and instrumental submixes, Billy Decker Bus Glue is simple and effective.

but new algorithms created to capture the style of processing he typically applies.

Are You On Glue?

Billy Decker Bus Glue is a suite of plug-ins that follows in the footsteps of the 'celebrity engineer signature sound' model established by the likes of Waves. In other words, the user only sees two or three very stylised controls, often with friendly and non-technical names, but under the hood, these are adjusting multiple parameters. (At least one would hope there's more going on than meets the eye, given that the Mac installer is well over a gigabyte in size!)

The Bus Glue suite is authorised using an iLok account and contains seven plug-ins. As the name suggests, these are designed to sit across the master bus and various subgroups within a typical rock, pop, country or metal mix. The six instrument-specific Bus Glue plug-ins target bass guitar, acoustic guitar, electric guitar, keyboards, drums and vocals. Although there are tonal aspects involved, the primary action of each plug-in is dynamic, and if you think of them as preset compressors set up for specific applications, you won't go far wrong.

The parameter sets vary slightly from plug-in to plug-in, but nearly all of them feature a 'one-knob' compression dial — labelled, variously, Clamp, Squeeze, Staput or, boringly, Compress — and another control simply titled Deckerate. There's an output level control too, as well as a hidden input trim which is accessed by clicking on the virtual VU meter. Some of the plug-ins also have wet/dry mix controls to permit parallel processing.

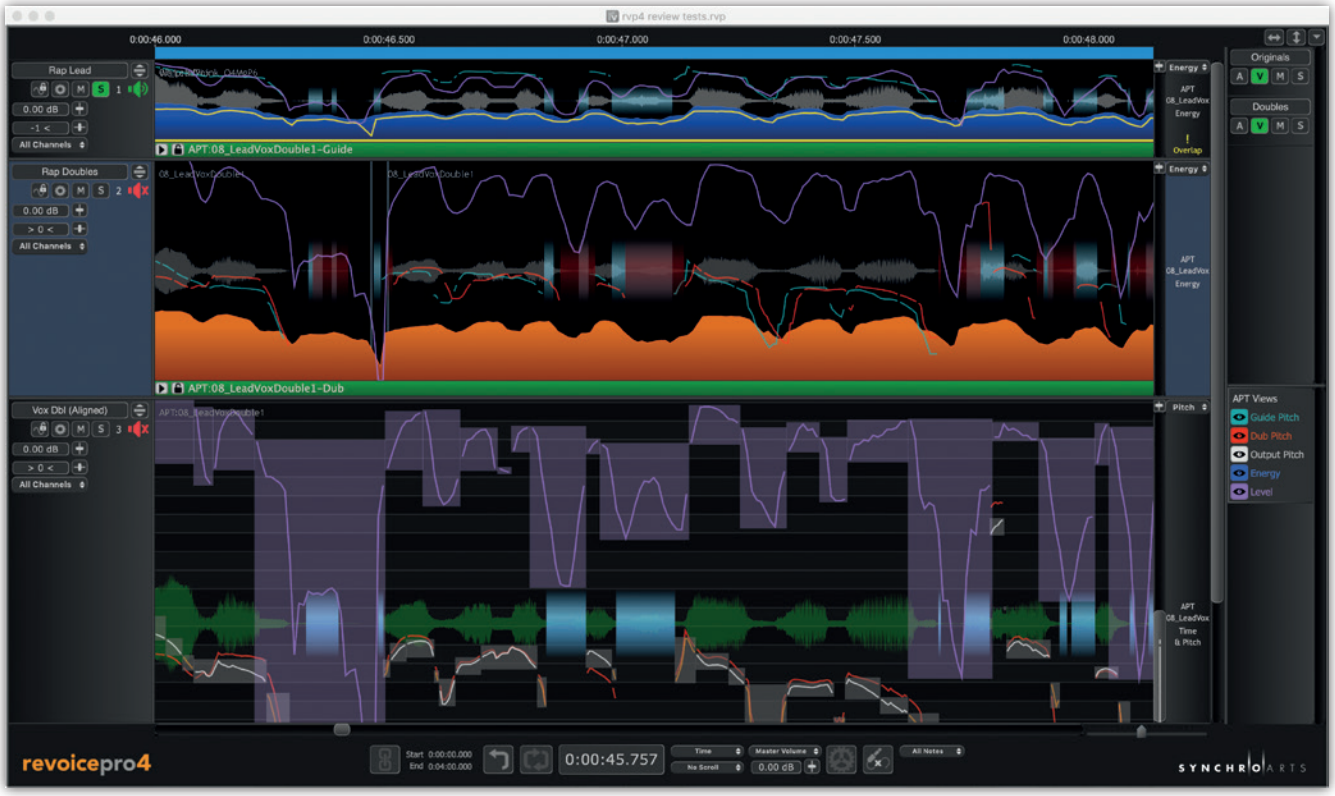
As you'd expect, the compression characteristics are very appropriate to the sources involved, at least within the musical contexts in which this plug-in is designed to work. It's easy to go from a gentle tickle of the meter to full-on crushing, and make-up gain is automatically applied as you do so. Oddly, however, this is not the case with the Deckerate control, which typically makes things both louder and brighter as you turn it up, and needs to be balanced by judicious use of the output level trim. Deckerate does different things in each plug-in, but typically involves upper-mid boost and either limiting or soft clipping, at least in most of the seven plug-ins.

The one instance where Deckerate doesn't change the perceived level is in the Master plug-in, where it's joined by a separate control labelled Finish, which absolutely does make things louder and brighter. A sort of re-Deckerating function, if you like?

A Full Deck

As you'd expect, these plug-ins are mostly simplicity itself to use. I do wish there was the option to compensate for the gain introduced by the Deckerate controls, but presumably their behaviour mimics faithfully the way Decker's own template works, in which case it's understandable. As long as the signals you feed into these plug-ins are somewhere in the ballpark sonically and stylistically, you can absolutely get together a pretty polished-sounding stem mix in no time at all. Which begs the question of what to do with the other seven hours, 15 minutes of the working day...

$ $199
W www.joeysturgistones.com



Synchro Arts Revoice Pro 4

Automatic Pitch & Time Alignment Software

The latest version of Synchro Arts' Revoice Pro brings many new features, including improved algorithms and deeper DAW integration.

MATT HOUGHTON

Revoice Pro is one of those rare applications that's genuinely unique: no other software or hardware is capable of doing everything it can do. There are plenty of pitch- and time-correction processors, of course, but only a handful that can generate convincing fake double-tracked vocals, and even fewer programs that can impose the timing of one vocal part on to one or more others. Cakewalk has had a VocalSync function for a while, and a vocal time-alignment facility was added in Steinberg's Cubase Pro 10; however, although the latter is impressive for a bundled DAW feature, I've not typically managed to achieve the same quality of results using that as I've obtained using Revoice Pro.

Of course, there's nothing in Revoice Pro that can't be accomplished manually in any half-decent DAW, but doing so takes time and effort. While the asking price of Revoice Pro is not exactly trivial, it could earn its keep very quickly in a professional music-production or film/TV post-production setting, whereas serious hobbyists might find that it leaves them with less 'left-brain work' to do, and more time to focus on the music.

What's New?

Until now, the biggest leaps in Revoice Pro's evolution occurred during versions 3.0 to 3.3, and if you want to find out about how this software works more generally, check out Sam Inglis' review of Revoice Pro 3 (www.soundonsound.com/reviews/synchro-arts-revoice-pro-3) and my series of Revoice Pro 3.2 workshops (www.soundonsound.com/techniques/revoice-pro-32-masterclass-part-1). Revoice Pro 4 is the most significant update in ages, and I'll focus in this review on what's new. The version I evaluated for this review was v4.0.0.26, and I'll refer to it from hereon as 'RVP4'.

Headline improvements include an ARA2 plug-in, which enables deeper integration with any DAW that supports that format. At the time of writing, this includes Studio One, Logic Pro and Cakewalk, and it's already in beta for Reaper. Steinberg have also announced plans to implement it in an update of Cubase Pro 10 (and presumably in Nuendo too, at some point).

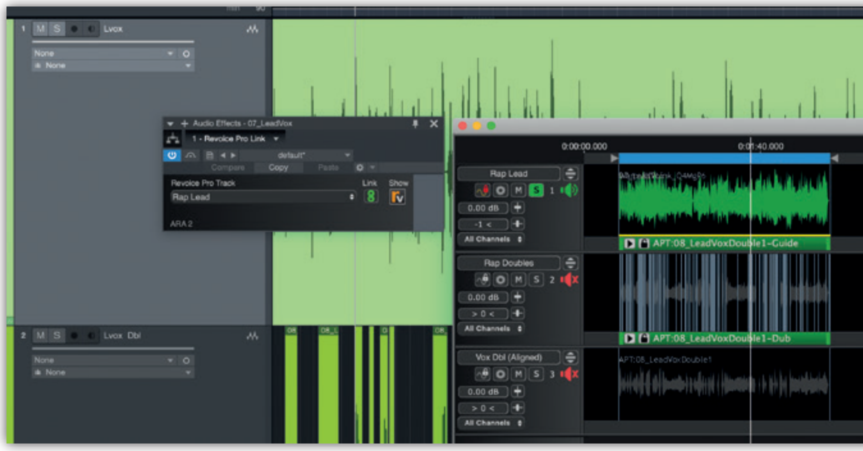
There have also been improvements to the process Synchro Arts call Audio

Performance Transfer, whereby RVP imposes the pitch, time and level of one part on to another. These include the provision of new APT algorithms specifically for musical material (the software has its origins in dialogue replacement for film and TV), and the ability to map the timing characteristics from a guide part (as before) but later pitch-correct the output from that APT process relative to a reference scale rather than the guide. The Warp Region function, which is used when you wish to manipulate the pitch, time and level of notes/phrases in an audio clip manually, has also received some useful upgrades.

Less obvious workflow improvements include the faster redrawing of some graphical elements, new shortcuts that allow you to perform existing actions in fewer clicks, and new facilities, such as being able to save a duplicate project for archival, complete with copies of the audio files. A nice touch is the introduction of drag-and-drop grouping of tracks, with the resulting groups allowing you to show, hide, solo and mute bunches of tracks in one click. For instance, with suitable grouping, you could solo the results from all your processes, but mute and hide the originals, all in three clicks.

There are various other small but useful tweaks, such as the ability to resize some windows and panes you couldn't previously — most notably the left pane, which contains the track names and main controls, and the right pane, which contains the groups and various controls that determine what's displayed in the central pane. New APT view options allow you to decide what 'traces' are displayed on the waveform. There are literally dozens of such small changes — far too many to list here, but full details are in the online manual's Release Notes: www.synchroarts.com/AppSupport/RevoiceProV4.0/Manual/ReleaseNotes.html.

Despite all this change, users of previous versions should find everything reassuringly familiar. The GUI has the same look and feel, with no confusing new windows, no new workflows being forced on the user, and all the old default key commands still working as you'd expect. Speaking of key commands, these can now be customised — a welcome improvement — though do be aware that you're not warned you if your chosen key command is already assigned to an OS function (in hindsight, Command+V was not the best choice for creating Vibrato Warp markers!). RVP4 uses a new project file type which, due to the new functionality, previous versions can't open, but RVP4 can open projects created in previous versions. All in all, things just feel generally a bit slicker and that little bit better; it's a product that is finally reaching maturity — though I'm sure there's better still to come!



Revoice Pro 4 being used via the ARA2 plug-in in PreSonus Studio One.

Installation & Basic Operation

On installing RVP4, I was informed that there was no licence on my iLok 2 dongle, yet the iLok License Manager utility told me otherwise. A quick look at the FAQs on Synchro Arts' website suggested I update the iLok License Manager. It would have been nice if the pop-up had warned me of that, rather than leave me to research it, but with the update done, RVP4 opened. Another pop-up then told me I was missing a couple of required plug-ins, but clicking OK installed them and all was well. It's worth noting that while it's possible to install RVP4 over previous versions, it might not work properly if you do, because the plug-ins may open the wrong version of the application — so if you're demo'ing RVP4, it's recommended to uninstall previous versions and keep your installers handy for later reinstallation.

RVP4 has several related elements. At heart, it's a stand-alone application, and you can still choose to use it as such, simply dragging and dropping audio clips from your DAW, processing them in RVP4, and dragging the results back. The audio is automatically 'spotted' on the RVP4 timeline, and most DAWs have a command for spotting time-stamped files too, though they use different terminology (for example, Cubase and Studio One call it 'Move To Origin').

Though it's much easier to use the stand-alone version than you might think, and I was happy enough using RVP3 like this, Synchro Arts have created a range of plug-ins to make integration with your DAW more seamless. Audiosuite plug-ins for Pro Tools allow you to apply presets such as creating a fake double-track, or time-aligning one clip to another without ever leaving the DAW; you only have to open RVP4 if you need to change the presets you wish Pro Tools to apply. The VST and Audio Units plug-ins perform different functions, allowing you to capture audio to RVP4 in real time, sync RVP4 to your DAW's transport controls, and stream audio from RVP4 to a track in your DAW.

ARA2

I've explored the stand-alone operation and the Audiosuite, VST and AU plug-ins »

Synchro Arts Revoice Pro 4 $599

PROS

- Unmatched time-alignment algorithms.
- Great-sounding pitch processing.
- Still creates convincing fake doubles.
- ARA2 plug-in works well.
- Can now identify esses and breaths.
- Various GUI and workflow improvements.

CONS

- No distinction between esses and breaths.
- Would be nice if the Revoice Pro project's settings could be automatically matched to those of the DAW when using ARA2.

SUMMARY

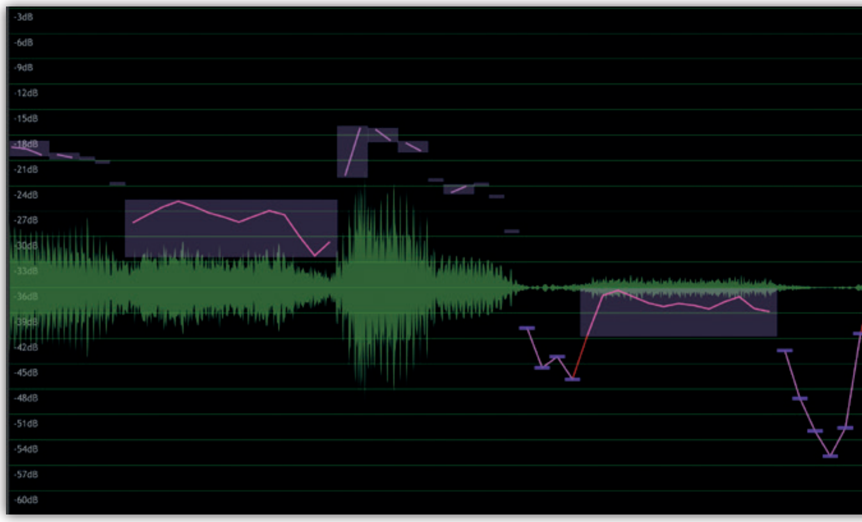
The previous version of Revoice Pro was already a highly effective pitch- and time-alignment processor. Now it's even better, and boasts deeper integration with many DAWs.

Alternatives

As explained in the main text, nothing else can do everything that Revoice Pro can. When it comes to time alignment, **Synchro Arts**' own **Vocalign** can do the same thing, and both **Cakewalk** and **Cubase 10** build in tools that take slightly different approaches to the same problem. Cubase's **VariAudio** can also create good fake double tracks, as can **Celemony Melodyne,** which is also compatible with ARA2 hosts, and offers a few tricks that Revoice Pro doesn't — though not the time-alignment functions.

» in these pages before, so I was particularly keen to discover what the new ARA2 plug-in would bring to the party. To test this, I used RVP4 with PreSonus Studio One 4.1, on a 2018 MacBook Pro running Mac OS 10.14.1 (Mojave). Despite a little teething trouble (borne largely of user error!), I have to say that it's a vast improvement. To get started, you simply right-click on a clip in Studio One, and select Audio / Edit With Revoice Pro. The ARA2 plug-in window opens, and includes a button to launch RVP4 and to sync the two applications. From a menu, you assign the currently selected clip to a track in RVP4, and it is automatically spotted on RVP4's timeline on the desired track, making the whole process of getting audio into RVP much easier. When you've done your processing in RVP4, hit Play in Studio One and you'll hear your results. The RVP4 project will be saved in and recalled by your Studio One project, even if you don't save the RVP4 project itself. And there's more...

So far, the audio you hear on playback resides in RVP4; the waveform in Studio One doesn't reflect any changes you've made and can't be edited. If you select the clips/tracks in Studio One and execute the DAW's Render function, though, all the RVP4 processing is brought into the DAW, and the corresponding part is removed from RVP4. It really is very slick. Some processes require additional tracks, of course, and you can return these to new tracks in the DAW in a number of ways



Revoice Pro can now identify and highlight unpitched vocal elements such as esses and breaths, and allows you to click and drag to attenuate them — the default pink line can be changed to a more obvious colour if you prefer.

— I tended to use the Shift+Option-drag (drag-and-drop) function.

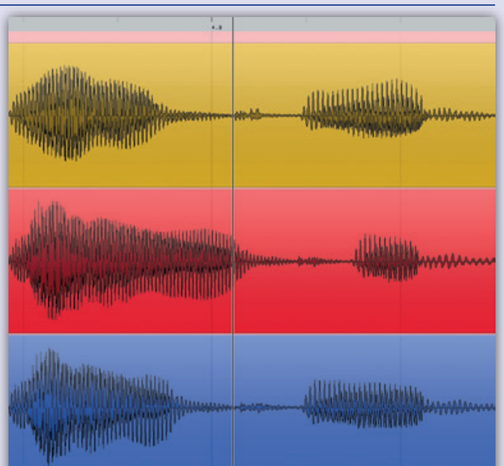
There are just a couple of niggles. First, while the ARA2 plug-in can launch RVP4 and create a new project, you still have to go in to RVP4's settings and manually update the tempo, frame rate and timeline offset to match you DAW song/project. There are good reasons these things need to be matched, but it would have been nicer if the plug-in could create an RVP project with those settings already populated — though perhaps that's a limitation of the ARA2 standard? Second, the dynamic link with the DAW means it's possible to wander inadvertently into 'loops of confusion'. For example, if you close the RVP4 project or delete a clip that the DAW's ARA2 plug-in generated in RVP4, the DAW will automatically relaunch/recreate it; that can leave you scratching your head until you realise what's happening! But these are minor issues, and for the most part the ARA2 integration is most welcome.

Warp: Vibrato, Esses & Breaths

The biggest new feature that comes under the 'Warp' heading is called Vibrato Warp. It's really impressive, in part because it's innovative but primarily because it's effective. Used just like the existing warp markers, Vibrato Warp markers give you control over a singer's vibrato rate separately from any time stretching/compression. Normally, of course, the vibrato rate increases as you »

Time Alignment: Revoice Pro 4 vs Cubase Pro 10

As a long-time Cubase user, I was keen to discover how well its new time-alignment algorithm compared with the ones in Revoice Pro 4. Each system has its advantages. Cubase makes use of its existing audio warp facility, and for the most part this works well. The changes you make are reflected in the waveform in the DAW, and the warp markers used in the automatic time alignment remain available for manual tweaking in your project. It wins hands-down in the DAW-integration wars, as you'd expect. But Revoice Pro delivers better results: its algorithms seem to be able to match timings that bit better. It's not that Cubase usually does a bad job — far from it — but RVP can do an outstanding one. By way of example, check out the screenshot, which shows a 'guide' part (top/yellow), the result of time-aligning a 'dub' using Cubase 10 Pro (middle/red), and the result of time-aligning the same dub using Revoice Pro 4 (blue). The part is a rap and this section features the lyrics "take it". Both applications have aligned the start and the end of the phrase, but notice the position of the small 'k' sound in between the higher-level parts of the waveforms, and of the start of the word 'it'. In several tests, Revoice Pro exhibited greater timing accuracy than Cubase with such low-level details.



time-compress a note and decreases as you stretch it, which can make it easier to spot that a vocal has been processed. This new function is easy to use and the results sound good.

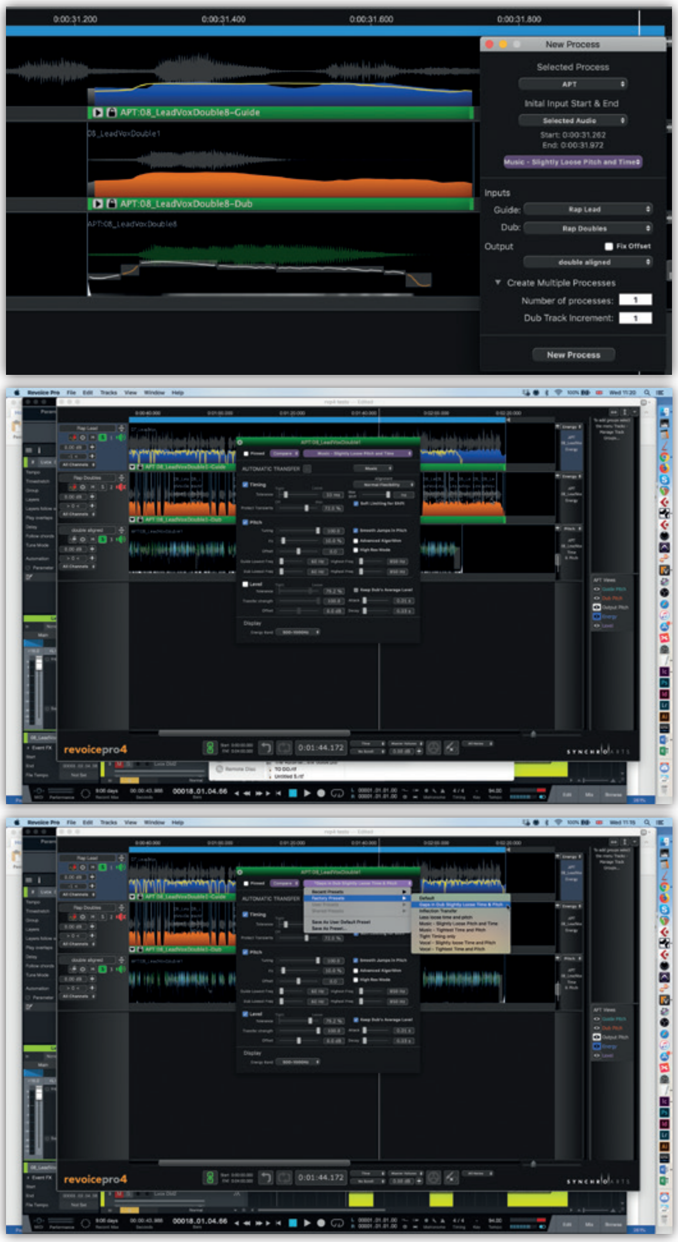
When first exploring RVP3.2, I suggested to Synchro Arts that it would be good if it could host plug-ins, or at least include a de-esser, de-popper and de-breather. This would allow the user to perform all his or her vocal editing/cleaning in RVP, instead of performing some operations in the DAW before rendering them to new clips for export to RVP. It seems Synchro Arts were listening, because RVP4 can identify breaths and esses in a Warp Region. Once you've created a Warp Region, hit the 'S' key and they'll be highlighted with pink traces. You can manually drag to adjust the level of each ess/breath, or Option-drag to boost/attenuate in 3dB steps. This is a welcome addition. I've always found editing and adjusting clip gains to be an effective approach to de-essing and editing breaths. RVP4 makes this easy, and I can do it in the same 'pass' as any manual pitch correction, making it less onerous a job than you'd imagine. Even if you don't achieve perfect results, a de-esser plug-in will have a less challenging job later on.

I have only a couple of small criticisms. First, the pink line of the ess/breath 'notes' perhaps isn't the most obvious contrast to regular notes' purple lines; thankfully, you can change the colour in Preferences. Second, RVP4 doesn't distinguish between esses, breaths and other non-pitched sounds; you have to listen to each one and decide if changes are required. Presumably that's why I could find no way to select all these 'notes' and adjust their level collectively. Intelligently identifying esses, breaths and so on separately is a rather bigger ask of an algorithm, but perhaps it's something that can be explored in the future?

Musical APT

I mentioned that there are some new APT options for musical material. For example, where before we had 'Slightly Loose Time & Pitch' now we have an option of 'Music Slightly Loose Time & Pitch'. To be honest, the old algorithm usually worked well enough on most sources, but the new one does too. Certainly, I could discern no audible artifacts when using it on suitable material. For aligning the pitch and time of doubles and backing vocals singing the same part as the lead, for example, it generally worked superbly, and using it to tighten just the timing of simple backing vocal harmonies (by unticking the preset's pitch section in the APT pop-up) yielded similarly pleasing results.

Note, though, that as with most time-stretching software, to achieve the best results, you should audition different



These screens demonstrate the importance of choosing or refining the algorithm to suit the material. The example comprises a rap lead and a double, with the double only emphasising certain words and syllables so there are some gaps. The first two screens show the result of applying the 'Music - Slightly Loose Pitch and Time' APT preset: notice how the 'aligned' part is much shorter overall than the original. The close-up shows where RVP has attempted to align one syllable of the dub to two syllables of the guide. The third screen, though, shows an almost perfectly aligned double — the result of the 'Gaps in Dub - Slightly Loose Time & Pitch' preset.

presets or tweak the chosen preset. For instance, when trying to align a rap lead vocal with a double that emphasised only certain words and syllables, most presets gave me very strange results. It was a while before I noticed the 'Gaps in Dub Slightly Loose Time & Pitch' preset, which worked flawlessly. Comparing this preset with others revealed that the Max Shift setting was the offending variable in the other presets. Making such comparisons between presets is a great way for RVP virgins to learn how to refine the results of APT settings.

Verdict

There is much more that's worth investigating in Revoice Pro 4, but I hope I've been able to convey a sense of what's on offer here. If you work in ADR or do lots of music work with stacked vocal parts (or, in fact, other instrument doubles — it works very well on bass, on guitar and various other sources) this tool could prove indispensable. It remains among the best-sounding pitch processors out there, and when it comes to time alignment, it's in a class of its own. The many small — and not-so-small — improvements should easily justify the upgrade price for most users, and most definitely will for anyone with a DAW that supports ARA2.

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WHAT IF YOU MADE A PRODUCT AND NOBODY NOTICED?
AV10-MHF
AV10-MLF
AV10-MLF
AV10-MHF
Germano Studios in New York City was the first beta tester of our AV10-MLF & AV10-MHF components. Housed in their classic NS10Ms since January 2018, they have had in top talent such as The Rolling Stones, Josh Groban, Sheryl Crow, Big Sean, J Balvin, Frank Ocean & countless mixers such as Kevin Shirley, Dave O'Donnell, & Conrad Hensel. All used the classic NS10M Monitors with our components. Nobody noticed. We wouldn't want it any other way.
AV10-MLF
THE FAMILY HAS EXPANDED.
AV10-MHF
NY USA AVANTONE PRO
CLA-10
Passive Studio Monitor System
CLA-10A
Active Studio Monitor System
CLA-200
Studio Reference Amplifier
AVANTONEPRO.COM

PLUG-IN FOLDER

Eventide Instant Phaser MkII

Formats: Mac & PC VST & AAX; Mac AU

Eventide's rackmount Instant Phaser was launched in 1972 with the aim of recreating tape flanging — which, as I recall, we used to call tape phasing back in the day. Electronic phasing didn't have the necessary delay time to produce a true tape flanging effect, but it got close and soon became an effect in its own right. Eventide have long offered a plug-in version of that original hardware, and now they've revisited it to add both new capabilities and the ability to 'age' the circuitry by introducing modelled component value drift. They've also made the behaviour of the emulation more faithful to the original hardware. If you already own Eventide's Anthology bundle, you get the new Instant Phaser MkII free of charge.

The hardware employed analogue multiple phase-shift stages based around all-pass filter networks that used FETs and RC networks wrapped around 741 op-amps, which gave it a smooth, musical sound. Modulation came from a 'not quite triangular 'LFO or from an envelope follower. As I recall, the 741 wasn't the quietest of chips but Eventide clearly got the best out of it. The sonic performance of that original analogue circuitry has been replicated in software complete with all the imperfect behaviours that circuitry produced, including the LFO shape. However, Eventide decided that a few additions would be worthwhile to make this phaser of the past more useful in studios of the present — and I suspect that includes ditching any unnecessary circuit noise.

This MkII version of the plug-in features a simple Age knob, which allows you to simulate the required degree of component value drift in the circuitry making up the phase-shift networks and the LFO. The range of this control goes beyond that of reasonable ageing to the point of 'completely knackered', though some interesting sounds can be produced by invoking excessive ageing. Once past 80 percent the effects become very apparent.

Instant Phasers old (top) and new (above)!

A newly added Mode switch provides three different phasing characteristics denoted as Shallow, Deep and Wide. In Wide mode, a different amount of phase shift is applied to the two channels, enhancing the sense of stereo spaciousness. Feedback adds resonance to the effect in the expected way and is a practical if obvious addition.

While a typical phaser pedal is driven only from an LFO modulation source, the Instant Phaser still has four modulation options: Oscillator, Manual (using the large phase knob), Envelope and Remote (MIDI). A newly added side-chain function allows the envelope follower source to be taken from a different DAW track, and the LFO can now be sync'ed to host DAW tempo with a choice of multiples. A new, automatable momentary Reset switch forces the LFO back to the start of its cycle. Depth sets the mix of dry and effected signal, but only goes from a 50/50 mix to all wet, so there's no way to fade the effect fully in and out; I did contact Eventide on this matter and while they couldn't commit to anything definite, I think it is safe to say that the matter is at the 'we'll think about it' stage for a future update.

Sonically, the MkII sounds a little more organic than the original plug-in, especially when you want to add subtle movement without the phasing effect becoming too obvious. The Wide mode creates a useful sense of stereo spread and movement, while increasing the feedback amount produces the familiar and far more obvious resonant phasing effect. That ageing control is also effective in changing the character of the effect in an organic way; things get very warbly and weird when you get past 80 percent, but sometimes warbly and weird is just what you need. A large choice of factory and artist presets shows off the versatility of the effect.

In all, then, this is a very significant update to what was already a great-sounding phaser plug-in, and if anything can rekindle your love affair with phasing, the Instant Phaser MkII is it. *Paul White*

$129 if purchased separately.
www.eventide.com

FabFilter Pro-Q 3

Formats: Mac & PC VST & AAX; Mac AU

I'm not sure if there is such a thing as a desert island EQ plug-in, but if I was stranded on a Pacific atoll with only one tool to tackle problematic audio files, FabFilter's Pro-Q would be that tool. It sounds good, it's amazingly flexible, and best of all, its graphical interface makes it a joy to use.

The Pro-Q user interface has been widely imitated since its debut in 2011, but FabFilter haven't stood still. The v2 update in 2014 added a swathe of new capabilities; the one I use all the time is the 'tilt shelf' curve, which is perfect for transparently rebalancing the tonality of a source or a mix, while other v2 highlights included 'match' EQ, a freely resizable window and greatly improved analysis features.

Launched late in 2018, the third major update to Pro-Q is likewise packed with functionality, and the headline feature this time around is dynamic equalisation. FabFilter fans will, of course, be well aware that the company already make the superb Pro-MB, which provides versatile frequency-dependent dynamics processing, so the question on many lips was how they would manage to introduce this into Pro-Q without either duplicating Pro-MB functionality or creating a pale imitation of it. And FabFilter fans won't be surprised to learn that they have indeed found a way to do this.

What you don't get in Pro-Q 3's dynamic bands is much control over the actual dynamics parameters. Attack, release and ratio are fully automatic, as is the threshold by default, though this can be switched to manual control. In use, though, this really isn't an issue, because the automated settings respond very intelligently to any input signal you throw at them, and you can easily set a limit to the amount of gain boost or attenuation that gets applied; the 'direction' of the dynamic action is also easily controllable, so you can have a boost that attenuates dynamically or vice versa. What you do get here that isn't on offer in Pro-MB is the ability to create very narrow dynamic bands, allowing you to dynamically dip out resonances, feedback whines and so on. Both plug-ins handle straightforward duties such as taming proximity effect on a close-miked vocal with aplomb; for more advanced work, and especially for controlling the tonality of complex signals such as a full mix, it is definitely worth investing in Pro-MB too!

There have also been further improvements to the analysis features. Previously, functions such as EQ matching relied on the user setting up side-chain routing, but now, instances of Pro-Q can communicate directly. In the pop-up control panel for the spectrum analyser, you'll see a list of all the other instances in your project. Selecting one will display the signal it's seeing overlaid with that of the open instance. This makes it so much easier to set up EQ matching, and also enables a new »

feature called Show Collisions, which warns you when two sources might be fighting for the same bit of spectral real estate.

Other new features include a 'brick wall' filter slope and the neat 'flat tilt' shape, which simply rotates the entire frequency spectrum about a fixed point, applying more and more boost or cut the further you get from that point. Finally, Pro-Q 3 now works in all major surround formats, and a drop-down menu for each band allows you to specify exactly which channels that band should affect — in stereo, the choices are left, right, Mid, Sides or both channels. Oh, and it'll even open your presets from Pro-Q and Pro-Q 2. I make no apologies for being a FabFilter fan myself, and updates like this are the reason why! *Sam Inglis*

$134

www.fabfilter.com

Antares Auto-Tune Access

Formats: Mac & PC VST & AAX; Mac AU

Antares' Auto-Tune needs no introduction, and the company have now launched a super-simple version that is more affordable and doesn't need an iLok to authorise. The control set has been pared down to a minimum, but still provides the tools you need to implement either gentle and natural pitch correction or super hard-tune 'Auto-Tune as an effect'-type processing. Auto-Tune Access can track according to a pre-selected scale, a scale specified by the user, or to a chromatic scale; clicking on the keys displayed at the bottom of the screen adds or removes notes from the current scale. This keyboard has plenty of space to display the note names and to highlight the currently identified note. However, it doesn't appear to be possible to change the frequency of A, which can be a bit limiting, especially if you're working with a piano that's not in concert pitch.

In normal use there are only two controls to adjust — what looks like a large knob in the centre of the screen is actually a display that shows the currently detected note and the amount of pitch correction being applied. However, there is also a Hold button that can lock the current note, which can be automated in most DAWs. Retune Speed offers slow, medium or fast options: fast gives the familiar hard-tune effect, and you can't select in-between positions. Likewise, Humanize just presents three options. Of these, Max allows more of the natural pitch variations to come through, helping to preserve a natural vocal sound, especially at faster correction speeds — but this should be turned off for the most dramatic hard-tune effect.

On the tech side, Auto-Tune Access is light on CPU load and has very low latency, making it suitable for live use. It is also compatible with Auto-Key, the optional Antares plug-in that detects the key and scale of a piece of audio, then sends that information to Auto-Tune Access or Pro.

In use Auto-Tune Access worked flawlessly within the limits that any fully automatic pitch-correction plug-in can. Where it can come unstuck is when the singer's pitching is so bad that they stray more than halfway between the intended note and the next note in the scale, in which case you can end up with a perfectly tuned wrong note. When applied to an already reasonably in-tune vocal, the results sound quite natural as long as a slow or medium retune speed is selected and one of the Humanize options dialled in. It also delivers the familiar robotic vocal sound with Speed set to Max and Humanize set to Off.

If your DAW already has auto-pitch-correction capability, such as Logic Pro's Pitch Correction plug-in, then Auto-Tune Access may not bring anything new to the party, but if not and you don't need all the fancy Auto-Tune Pro features, Auto-Tune Access is a very practical proposition for adding polish to vocals that are already 'nearly there'. *Paul White*

$99

www.antarestech.com

Leapwing Audio StageOne

Formats: Mac & PC VST & AAX, Mac AU

Leapwing Audio have produced a series of innovative plug-ins that appear to be prompted above all by demand from

mastering engineers, but which have applications in mixing too. I recently reviewed their intriguing CenterOne v2, which provides tools for mono extraction and stereo rebalancing; StageOne is also a plug-in designed to manipulate stereo imaging, but in a very different way. The basic concept seems to be to extract a signal that corresponds to what we hear as the 'phantom centre' and process this independently of the rest.

Stage One contains three basic processing modules called Width, Depth and Mono Spread, each of which can be enabled or disabled independently. Width is perhaps the most conventional in terms of its application — it's a stereo widener — but the techniques it uses to achieve this goal are obviously a bit more sophisticated than the usual 'fling an M-S matrix on and ramp up the Sides level' approach. In essence, it appears to be boosting the level of the 'not phantom centre' component above a user-specified frequency threshold; this is not the same as boosting the Sides signal, and sounds quite different. Used in moderation, it's impressive, reasonably mono-compatible, and has a solid feel to it that doesn't hollow out the mix or sound weird on headphones in the way that exaggerating the Sides can. That said, the available range is rather higher than I can imagine using in a mastering context, and high Width values introduce obvious pitch modulation into the signal.

Next up is Depth, which adds early reflections that are "directionally optimised to create an enhanced sense of depth in the sound field". As you might guess, this is not a conventional reverb, and the only controls available are Depth and a Color slider, which manipulates the frequency balance of the added reflections. In use, it's quite a mysterious processor, and it can be hard to predict when it will be effective. I didn't like it on everything, but to my surprise, I loved it on a mix that had been recorded live and was already quite reverberant, where it seemed to glue everything together in a way that I hadn't realised needed doing.

Finally, the Mono Spread module is more versatile than its name suggests. Fed a mono signal, it performs something that is conceptually akin to the old 'electronically reprocessed for stereo' trick that blighted so many albums in the late '60s — but which actually sounds remarkably good. That's progress, and the Mono Spread feature is equally at home with stereo sources too. Raising the Mono Spread slider increases the width of that 'phantom centre' component, but what's most useful is the Center Gravity control. This allows you to shift the apparent focal point of a stereo signal to the left or right, either to centralise an unbalanced stereo image or to push a stereo source to one side of a mix without making it mono.

Mono Spread is probably my favourite of the three modules, and is capable of surprisingly transparent results, but all three of StageOne's, er, stages have the potential to be very useful in both corrective and enhancement roles. What this plug-in does might appear familiar, but the way that it does it definitely isn't! *Sam Inglis*

$199

www.leapwingaudio.com

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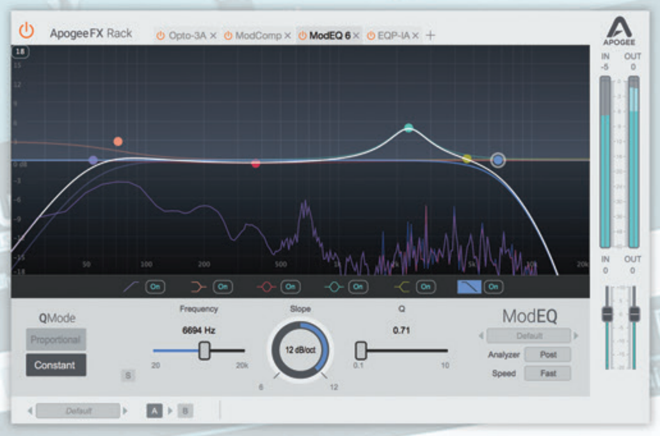

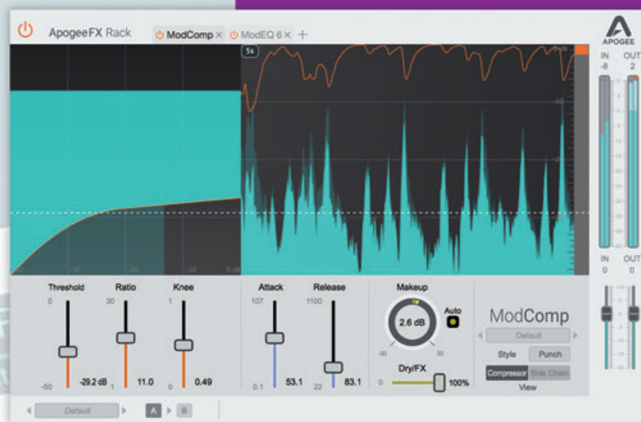

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APOGEE

PreSonus Atom

Pad Controller

The PreSonus Atom offers hands-on control and tight integration with Studio One.

ROBIN VINCENT

The mightily compact Atom turned up a little unexpectedly. PreSonus have been pushing the seriousness of their DAW and the professional level of their hardware for some time now, and a colourful box of pads seems slightly at odds with that. But delve beneath the colourful exterior and you'll discover an engaging workflow that has a lot more in common with the FaderPort. It would be a mistake to see it as a toyish beat controller; the Atom has a lot more going for it than pretty light-up pads.

The Atom is a 4x4 MPC-style pad MIDI controller, but it's the tight integration with Studio One that really sells it. You could see it as a hardware projection of the Impact XT software sampled drum instrument. Playing percussion is always far more satisfying on a pad controller than it is on a MIDI keyboard. But there are also buttons and modes that let you build up tracks without as much as a glance at your mouse. And it's in that workflow that the Atom really works its magic.

First Impressions

But first, the basics. Load up a kit into Impact XT and the Atom lights up in response with all the same colours as the GUI. You can then get to work on the nicely responsive velocity-sensitive and pressure-sensitive pads. Each pad turns white as it's triggered, making for quite a disco dancefloor in response to your finger-tapping skills. You don't have to strike the pads with much force to get them to work, in fact you can trigger them with just a slight push. The pads feel right, a little smaller and wider spaced than those on NI's Maschine, but good and chunky beneath the fingers.

The build of the Atom is solid and unyielding with a nicely low profile, and feet sticky enough to keep it from moving around. I did find a little bit of 'crosstalk' between pads; if I pulled down on the bottom half some of them had a tendency to trigger the pad above. But that's more of an anomaly when trying to find fault with a device rather than something you'd hit upon when playing.

At the top are four tapered encoders that feel solid, although they are the only parts of the Atom not lit and so tend to be difficult to see, but with encoders you're working more with feel than with accurate positioning. In Impact XT these are pre-mapped to the amp gain, panning, pitch and decay, and work on the currently selected sample or last triggered pad.

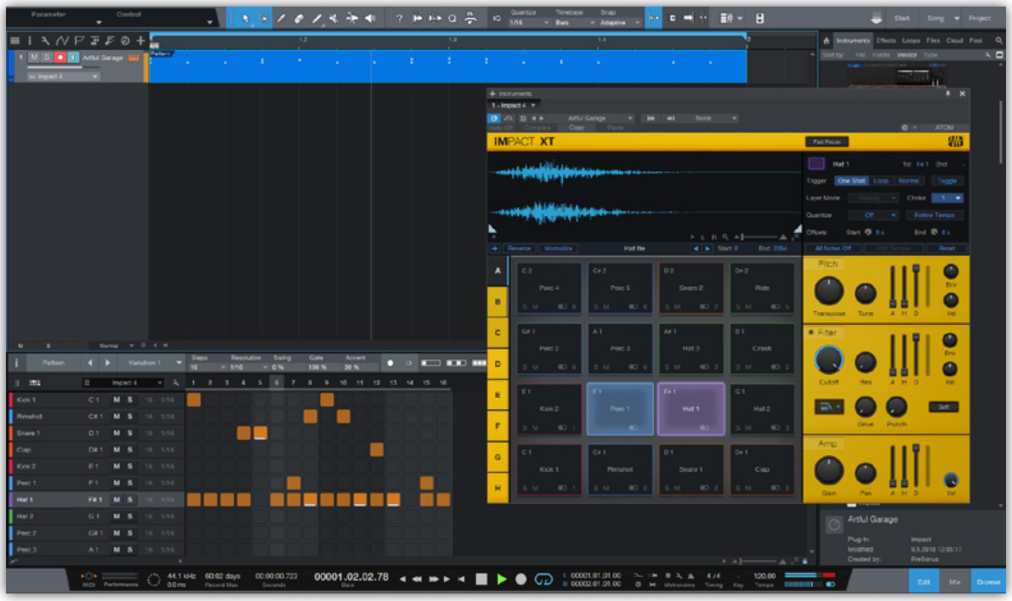
There are a couple of performance control buttons on the left. The first one enables Full Level mode so the samples are played at maximum velocity however hard you play. The second is Note Repeat, which turns the bottom two rows into ratcheting options for the last played or selected sample. They make for easy inputting of rolls and fills into patterns and have some live performance potential if you have fast fingers and really know what you're doing.

It's at this point that I start to lose time playing with the kits, like you do with any well-crafted integration between software sounds and a controller.

Workflow

What keeps you focused on the Atom are the function and navigation buttons. The Atom is not trying to be a comprehensive DAW controller, instead you have just enough to keep your hands off that mouse while putting together multiple tracks of beats and patterns.

All the pads have additional functions printed on them, and these become active depending on what buttons are pressed or held. Starting with a blank Song, press the top-left Setup button and tap pad 13 to show the Browser. On the right are some navigation buttons, so you can tap Down to find Impact XT



The Atom works well with Studio One's Impact XT instrument.

in the browser instrument list, tap Right to open the presets and then Down/Up again to find the kit you're after. Hit Select to load Impact and the kit onto a new track. The GUI comes up and you're off finger tapping. You can step through the preset kits by holding the Preset button and making use of the navigation buttons — no need to refer back to the Browser. Holding the Bank button illuminates the bottom two rows that represent banks A-H, and you select the Impact drum kit bank with a tap.

When you are ready to record you can either insert a pattern, which we'll come to in a minute, or simply hit Record and do your thing. There are four transport controls in a column on the right, each with a dual function that you access by holding the Shift button. At the top in blue you have Click enable and Shift to enable a Count In. Next you have Record in red with Save under the Shift function. Green is for Play and Shift enables loop playback. Lastly, Stop is yellow, and also has the enormously useful Undo function when used with the Shift button.

Once you've finished recording, press the Event Editor button to open the drum editor and now you can use the navigation buttons to step through the notes and move them about. In this mode only pads 9-12 are lit, offering Duplicate, Delete, and Velocity + and - for the currently selected note. A Nudge button helps you push and pull notes, and using Shift will snap the note to the quantise setting.

While the functions are fairly basic there are just enough buttons to enable you to record and edit a drum part without having to take your hands off the Atom.

Playing With Patterns

The Atom is perfectly suited to Studio One's Pattern Sequencer. Pressing Setup + pad 5 inserts a pattern at the current song position. Pressing Shift + Set Loop puts the loop markers around the pattern and pressing Editor brings up the pattern editor and enables sequencer mode. Instead of a colourful drum kit you now see each drum lane available as a 16-step pattern spread over the 16 pads. Turn the steps on and off, add accents with the use of the Shift button, use the navigation buttons to change lanes or groups of 16 steps. It's a very smooth pattern-building experience.

What isn't served by the Atom/pattern experience is the automation lanes. As you can't select individual notes in a pattern (they turn on or off) there's no way to direct a controller to step parameters such as Repeat and Probability. You can't, for instance, hold a pad to enable the step of a snare and then turn a knob to alter the probability. Those sorts of things are still in the domain of the mouse. You can do regular track automation with the four knobs, but only on regular track and instrument parameters — you can't map them to those two key pattern parameters.

Controlling Instruments

It doesn't have to be all about Impact XT. Atom will happily work as a clumsy pad-based piano keyboard. It works well enough to write melodic parts and chords. You can load an instrument such as from Mai Tai using the Atom, just as you did for Impact XT (with browsing), and it generates a new track ready to go. The first 13 pads become notes of a scale, with white notes in yellow and black notes in blue. Pads 15 and 16 become transpose up/down buttons. Knobs 1 and 2 are automatically assigned to cutoff and resonance, and you can map all four to whatever you want.

It was at this point that I ran into a bit of trouble. The mapping of the knobs didn't seem to stick. After a bit of fiddling I discovered that the Atom's knobs would only control parameters when the instrument GUI was in focus. This was true of Impact, Mai Tai or any other instrument I was using. I probed PreSonus about it and they said it's because the knobs function differently in different control layers. For example, knob 2 controls Tempo in the Song Setup layer. So, in order to prevent conflict they decided to have them active with whatever was in focus. This means you can have the knobs mapped to something globally that is different to how they are mapped when the instrument is in focus, which brings in some unexpected versatility while still being a little annoying. »

PreSonus Atom $149

PROS

- Small footprint but solid build.
- Impact XT integration is fabulous.
- Pattern sequencing.
- Enough buttons to let you ignore the mouse.
- Note Repeat.
- Decent pad controller for other DAWs.

CONS

- Could go further with Pattern Sequencer.
- Needs GUI in focus to respond to knobs.
- Very colourful.

SUMMARY

The Atom is a usefully compact pad controller for Impact XT and the Pattern Sequencer in Studio One, and can also hold its own in other DAWs and with other instruments.

The Atom's Melodic Step Record mode in Studio One.

» Timeline recording is as you'd expect, and when you press the Editor button it now brings up the regular piano roll view. You can navigate the editor and edit notes and velocity, just as you did in the drum editor. Moving notes around, quantising and simple actions are all very easy. When you close the Editor you can then use the same buttons to navigate in the timeline and can use the same Duplicate and Delete pads to do a bit of arranging. You can step up and down through tracks, duplicate clips and insert patterns and select what you want to edit next. The Atom is no replacement for all the tools and possibilities of your regular mouse and keyboard combination, but you can nip around and do some basic editing all from the controller.

Melodic Patterns

You can use the Pattern Sequencer with non-Impact instruments, but it's not quite so fully formed as it is with drum sequencing. When you bring up the pattern editor on an instrument track the pads all go grey and unlit, which doesn't really inspire confidence. If you disengage the Editor button it doesn't close the pattern editor, it just puts the pads back into play mode. If you then enable Step Record with your mouse you can use the pads to enter a pattern of notes step-by-step. What's missing is a button to enable Step Record and also a button to enter a Rest. But this is where your secret user functions come into play. They're not really secret, but they feel like a discovery when you first stumble upon them.

User Functions

Pressing the Song Setup button turns the pads into buttons to access things such as tempo and the Browser. The bottom two rows of pads (1-8) are actually user definable function buttons. By default pads 5-8 are set to insert patterns and create variations. Pads 1-4 cover Duplicate, Select All, Mute and Solo. You can access and edit these via the External Devices window in Studio One. This is also where you can map the knobs to things. So, prompted by the lack of dedicated Step Record and Rest buttons I set these up here in order to give me a fuller melodic pattern experience. Right-click a pad and choose Assign, search for 'step' and under Step Record you can assign 'Enable' to pad 3 and 'Insert Rest' to pad 4. You can change the pad colour if you wish, and you're ready to go. I'd very much like to see those two functions incorporated into the main interface somewhere, but the User Functions are very handy and give you the space to create the shortcuts to the tools and actions you need in your workflow.

Venturing Outside Studio One

The Atom is also a regular MIDI controller. The pads put out MIDI messages and all the knobs and buttons are similarly assigned. There's no particular cleverness about it, no automatic mapping, or Komplete Kontrol or VIP style plug-in wrapper, and it doesn't even use Mackie Control for the transport. But it still plays really nicely in other DAWs and with other virtual instruments. There are a few modes and adjustments you can make to smooth out your non-Studio One experience. Under the only non-light-up button on the Atom are the background controller settings. Here you can swap between drum pad layout and keyboard layout, and you can adjust the pads' velocity curve, pressure mode and responsiveness.

I found that, when using the Atom with kits in Bitwig Studio Drum Machine, it needed a softer velocity profile, which I could access in the Setup. Note Repeat still worked well and velocity was automatically mapped to pad pressure. Mapping the knobs was all very easy; this doesn't have to be all about Studio One.

Final Thoughts

The Atom is a little powerhouse of pad-oriented control. Being made by a DAW company gives it that deep integration with Studio One that's very enticing. It creates an easy workflow and the secure and uncommon feeling that it just works. The compact size is a definite advantage — you can whip it out when you want to use it and it's not going to take up an annoying amount of space on your desk. And yet they've managed to squeeze in enough functionality to keep your hands off the mouse and your fingers on the pads. It's not going to fulfil the role of a DAW controller, but then that's not its job. There could be some deeper control in the Pattern Sequencer (accessing those Repeat and Probability functions would be nice) but PreSonus say these are things they are looking at, so there could be more around the corner. Outside of Studio One, the Atom works remarkably well as a simple pad controller, so while it's a no-brainer for beat makers in Studio One, users of other DAWs are not getting a raw deal, just a simpler one.

$ $149.95
W www.presonus.com

Roland VT-4 Voice Transformer

Vocal Processor

From the classic vocoders of the '70s through to the hyper pitch-corrected pop vocals of the last decade, Roland's VT-4 can do it all.

RORY DOW

The VT-4 is the latest version of Roland's popular Vocal Transformer line, designed to give you all manner of popular vocal effects in a handy and inexpensive desktop box. The original VT-1 was a Boss product first released in the later half of the 1990s. The last version, the VT-3, was reviewed in *Sound On Sound* in May 2014 by Paul Nagle. Back then, Paul praised the VT-3 for its versatility, but noted that the pitch correction had reliability issues. Let's see what the latest incarnation has to offer.

Front & Back

The VT-4 is a similar format to its predecessor. At 174 x 133 x 58mm, it's slightly smaller, but maintains the four faders for principal controls, plus the large knob in the middle. The green livery of the Aria range is present, but much more subtly than in the VT-3, being confined to the LED-backlit buttons.

There's no traditional power supply socket; power comes via the USB 2 port or via four AA batteries if portability is required. The USB port also allows the VT-4 to be used as a USB audio interface — not class-compliant, sadly; drivers are required. Around the back is a 'combi' XLR/TRS input with optional phantom power for your microphone, and a stereo

pair of line outputs. There’s also a MIDI input port, more on that shortly.

On the front of the unit we find a 3.5mm headphone output and another microphone input. This input is to be used with more inexpensive plug-in-power compatible microphones. You’ll also find switches for mono/stereo output and a phantom power switch for the rear XLR input.

Simplicity has always been a key factor with the Vocal Transformers and the VT-4 is no exception. Plug in a microphone, adjust the mic sensitivity control and off you go. There are four sliders on the front panel, which always have an immediate effect. The Balance fader is your wet/dry control, mixing the processed voice with your natural voice.

“The robot and vocoder modes are hellishly good fun and the other effects combine to make a well thought-out vocal processor.”

The Reverb fader applies a simple reverb or delay effect, chosen from eight different presets. These effects aren’t really the VT-4’s *raison d’être*, but they do add a little icing to the cake.

The other two faders on the front panel are the Pitch and Formant controls. These are fundamental to the Vocal Transformer and work in all modes. Pitch transposes your voice, up to an octave either way. Formant adjusts formant filters to achieve gender-bending results. With these two controls, one can summon anything from Sauron himself to Alvin the chipmunk on helium, and everything in-between.

Bag Of Tricks

In the middle of the VT-4 is a big knob labelled Auto Pitch. It can be used in any mode to apply automatic pitch correction to your voice. Fully counter-clockwise, there is no pitch correction. As you move the control clockwise the pitch correction becomes more pronounced until, when fully clockwise, your voice becomes hard-quantised — an effect beloved of saccharine pop music. You can choose to correct your voice to any major or minor scale using the Key knob. You can select any of the 12 notes in a chromatic scale and that will give you the major scale in that key. If you want a minor scale, you simply turn the knob three clicks to the right, which will be the relative minor scale. Other scale types are not possible.

In his review of the VT-3, Paul found that the pitch tracking left something to be desired. I’m happy to report that the VT-4 did an exemplary job of tracking and correcting my voice. There were no odd glitches or obvious tracking problems.

Having dialled in the perfect amount of pitch correction, you might want to hit that Harmony button to add sympathetic harmonies to your voice. You can choose from up to eight harmony presets using the Effect Variation buttons (with a modifier to select presets 5-8). They cover the most common vocal harmony intervals with up to three voices, but sadly there’s no way to create custom harmonies (or is there? see the ‘SysEx’ box).

With Robot mode engaged, your voice is stripped of any expression and the pitch remains utterly constant at a note set by the Key control. The Pitch and Formant sliders will still do their thing. Once again, you can select from one of eight different Robot variations using the Effect Variation buttons. They tend to simply change the base octave, but there are a couple of variations that utilise feedback to create a sort of ’70s sci-fi comb-filter robot sound, which I was rather fond of.

The next effect is the Vocoder mode, which when engaged will vocode your pitch-corrected, robotized vocal using an internal carrier signal at the pitch set by the Key and Pitch controls. Again eight different flavours of vocoder are available, accessed via the Effect Variation buttons. These cover a range of styles such as classic Cylon vocoding, Stephen Hawking speech synthesis, Talk Boxes, and there’s even a lo-fi Speak & Spell-inspired preset. They’re all great fun.

Here is where that MIDI input comes in handy. With a MIDI controller connected, you can control the Robot or Vocoder carrier pitch using MIDI notes. And if you »

SysEx

Looking through the MIDI Implementation manual available on Roland’s website, it seems there are quite a few parameters exposed via SysEx that are not accessible any other way. For example, it appears possible to edit your own harmonies by choosing different intervals for each harmony voice. It also appears possible to change the gender of each harmony voice. Further investigation revealed SysEx control over reverb, megaphone, equaliser and vocoder parameters which don’t exist anywhere else. Sadly, I didn’t get time to try these out, but intrepid Internet explorers are always one step ahead, and it seems that these SysEx parameter do indeed work. My guess is it won’t be long before some enterprising soul makes an editor that exposes all this. Most interesting!

Roland VT-4 Voice Transformer **$229**

PROS

- More vocal tricks than you can shake a stick at.
- Excellent pitch tracking and correction.
- MIDI pitch control with harmonies up to four voices.
- A fully fledged vocoder.

CONS

- None at this price.

SUMMARY

Over the years, Roland’s Vocal Transformer line has ripened into a solid box of tricks. Whether you just need gentle pitch correction or want to turn yourself into a robot barbershop quartet, the VT-4 has you covered.

» press the Harmony button, you can play up to four voices. It works beautifully and really opens up the possibilities of both Robot and Vocoder modes.

The last effect you can apply on top of everything previously mentioned is the Megaphone. This supplies you with a selection of distortion or modulation type effects different to the Reverb effects applied later in the chain. Like the other mode switches, the Megaphone has eight variations which go beyond a simple megaphone into effects like FM radio, choruses and vibrato. Again, these effects are basic and there are no parameters to tweak (at least from the front panel — see 'SysEx' box again), they're either on or off. Nonetheless, they provide some fun extra character.

Finding My Voice

In order to help you get the perfect sound, there's a low-pass filter and noise gate that can be applied to the incoming signal. There's also an enhancer effect which can be enabled on the processed signal. Controls are limited to using a combination of keys and the four Effect Variation buttons to choose one of four strengths for each element. It might seem limited, but it's usually enough to sort out any problems with unwanted noise, mic rumbles etc.

Once you've crafted the perfect hell-demon, '70s robot or perfectly pitched pop harmony, you're going to want to save a preset. Roland call them 'scene memories'. There are just eight slots to save in, two banks of four. Saving is as simple as pressing one of the four memory buttons (plus a modifier for slots 5-8) for a few seconds. Eight slots might not seem overly generous, but given the simplicity of the VT-4, it doesn't seem like a troublesome limitation.

The VT-4 can also double up as an audio interface. Connect a USB 2 cable to your computer and install the drivers from Roland's website and you're away. The audio interface opens up some interesting options. Audio played back from the computer can be routed straight to the VT-4's outputs, but also to the mic input and the Vocoder carrier input. The VT-4 sends audio back to the computer in one of three stereo streams: mixed, wet or dry.

This means you can, for example, pitch-correct pre-recorded vocals without leaving the digital domain, straight in and out of the DAW. Not only that, it gives you the option to use your own carrier signal in the vocoder, which is not possible any other way and turns the VT-4 into a fully fledged vocoder. Record your vocals dry in the DAW and send them to the mic input whilst you send a pad or synth bass line to the carrier input. Dial in just the right sound and record both the wet and dry signals simultaneously back to your DAW for further processing.

Conclusion

The VT-4 is way more fun than it has any right to be. The pitch correction was, in my experience, fast and accurate. The Robot and Vocoder modes are hellishly good fun and the other effects combine to make a well thought-out vocal processor. The ability to use MIDI for pitch or supply your own vocoder carrier via the USB 2 interface actually qualifies the VT-4 as a vocoder, and eight variations of every effect mean you'll be finding new combinations for a long time.

The Vocal Transformer line of products started out as a simple effect, but has matured into a flexible and impressive vocal toolbox. It could be used for everything from subtle pitch correction, harmonising and vocoding to Hollywood sound design of demons, fairy-folk or spy-thriller vocal disguises. Quite simply, I imagine anyone who records vocals will find a use for it. ■■■

VT-3 Vs VT-4

If you're thinking of upgrading from the VT-3, here's a run down of the main changes. Improvements include reliable pitch detection, multiple effects at once, MIDI pitch input of up to four voices, eight variations of every effect and the vastly improved vocoder. There are also eight memory slots instead of three on the VT-3. The VT-4 is slightly smaller so will fit into the same desktop space with some room to spare.

VT-3 features which didn't make it to the VT-4 are the Synth modes, which play synth sounds with your voice, and the foot-pedal input, which was used for hands-free bypass. Roland's Scatter effect, ubiquitous on the Aria line of instruments, also didn't make the grade.

On the VT-4's back panel we find a power switch, a USB port, a MIDI input, a 'combi' jack/XLR socket and a pair of quarter-inch audio outputs.

Around the front is a 3.5mm headphone socket, a 3.5mm mic input socket, a switch to select mono or stereo line out, and another switch to turn the XLR input's phantom power on or off.

Alternatives

There may be nothing with the unique combination of effects the VT-4 offers, but there is a fair bit of competition in the pitch-correction and vocoder markets. For a dedicated vocoder, Roland's own **VP-03** is worth a look (reviewed by Gordon Reid in July 2017). For a similar suite of effects in a guitar pedal, the **Electro-Harmonix Voice Box** offers harmonising and vocoding plus a few other tricks, but without pitch correction. Finally, the **TC-Helicon VoiceTone Harmony-G XT** (reviewed by Paul White in August 2011) offers harmonising and effects but no vocoder.

$ $229.99
W www.roland.com

Genelec S360

Active Monitors

Genelec's latest monitor is a flexible beast, designed for the large multi-speaker setups that are needed in VR and immersive audio. But does this versatility come at the expense of its usefulness for conventional music mixing?

PHIL WARD

It's seemed to me for a while that some of the traditional dividing lines between different types of speaker are becoming blurred. Of course, there have always been examples of speakers with feet in multiple camps, the Yamaha NS-10 and the BBC-designed LS3/5A being obvious examples. But in our increasingly multi-format, multi-channel, multi-platform audio world where, for example, a video game soundtrack that's largely choral was recorded at AIR Studios (*Everybody's Gone To The Rapture*), or live music is broadcast as an immersive audio stream (see Moods Jazz Club: www.moods.digital/en/), the traditional roles of speakers are changing.

The nearfield/midfield monitoring speaker niche that I most often write about in these pages is a perfect example of a market sector where change is afoot, and the subject of this review, the Genelec S360, shows how one manufacturer is responding to the change. But how, you might reasonably ask, are things changing? Well, it's predominantly about channel count and scale. Traditional music mixing brings to mind a pair of small(ish) nearfield speakers located maybe a metre from the listener, a pair of midfield speakers perhaps two metres or more away, or a pair of soffit-mounted main monitors either side of the control-room window. But those traditional paradigms can't really accommodate the increasing demand for multi-format and immersive multi-channel material.

Genelec's expectations for the way in which the S360 is likely to be used illustrate perfectly why 'we're not in Kansas any more' when it comes to monitoring. They see the S360 not as a nearfield or even a midfield monitor, but as a speaker to be used flexibly in multi-channel monitoring, for TV, movies, games and VR applications,

where perhaps 16 or more speakers, each potentially at a significant distance from the listening position, might be required to create an 'object-oriented' immersive soundfield capable of reaching genuinely high volume levels. So the fundamental design brief for the S360 reads: compact, flat and wide bandwidth, tightly controlled dispersion, and ability to play loud. It's a brief that has as much 'PA' about it as it does 'monitor'.

Tech Specs

Genelec are relatively unusual among monitor manufacturers in taking their responsibility to publish believable technical specifications seriously, and a quick scan through the data in the S360 user manual shows there's little doubt that those design brief parameters have been met. The S360's frequency response is flat on axis from 39Hz to 19kHz, within ±2dB limits. Its horizontal dispersion falls in a gentle and linear way with frequency, and in the vertical axis, shows only a narrow discontinuity over the crossover region. And a single S360 can achieve a short-term sound pressure level of up to 112dB at two metres in a room with a 0.4s reverb time. That's a genuinely high sound pressure level for a relatively small monitor.

Without self-spoiling my review, Genelec's success in satisfying their design brief is notable, and the S360's performance is in some respects an advance on the kind of monitoring we're all mostly used to. However, the S360 is still at heart a traditional moving-coil speaker that would look familiar to Rice and Kellogg who first (well, some say that's debatable) came up with the concept in the 1920s. In fact, and somewhat ironically, despite its intended application in genuinely cutting-edge audio production, the S360 is in some ways more conventional than Genelec's other recent monitor launch, the One series.

Having written that the S360 is reasonably conventional in terms of appearance, I'll move on to describing it more fully. In terms of overall dimensions, the S360's 36 x 36 x 53cm isn't all that much bigger than the kind of size that just about fits into 'nearfield' expectations, although it more comfortably fits the 'midfield' bracket. The S360 enclosure is constructed of black or white finished birch ply. The front edges are rounded to reduce diffraction and the horn waveguide of the tweeter is machined directly into the wooden carcass. On each side of the enclosure are tapped bosses intended for the connection of mounting hardware, and the underside of the enclosure is similarly equipped for the attachment of speaker stands. The underside of the enclosure also incorporates a plinth that both positions the down-firing reflex ports an appropriate distance from any mounting surface, and provides some mechanical decoupling. The plinth can be removed when the S360 is stand-mounted.

The S360 is a two-way active system, with the usual arrangement of amplifier and connections around the back. The amplification employs Class-D technology and is rated at 250 Watts for the bass/mid driver and 100 Watts for the tweeter. Very unusually, and potentially very usefully in some installations, the S360 amplifier module can be removed and rackmounted remotely. Connection from amplifiers to speaker in the case of remote amplification is achieved through four-way Speakon connectors. I'll move on to describe the S360's configuration and control facilities more fully a few paragraphs down the page, because I first want to write a little about its drivers.

Drivers

Said drivers are perhaps the most obvious sign that the S360 is not a typical contemporary studio monitor. Firstly, the tweeter is a deep horn-loaded, titanium-diaphragm compression driver. Compression drivers are most often found on speakers designed for live-sound applications, where efficiency, very high volume capability and reliability are priorities — sometimes, it has to be said, at the expense of sound quality. Having 'dissed' compression drivers, however, my experience of the JBL 7 Series monitors, reviewed in the February 2018 issue, showed that these days they can definitely be made to compete on sound-quality terms with direct-radiating drivers. The operating principle of a compression driver is not, in reality, hugely different from that of a direct-radiating driver: there's still a voice coil suspended in a magnetic field and attached to a diaphragm. The difference is that the diaphragm radiates into a small volume, connected to a horn (often these days known as a waveguide) by an aperture that's significantly smaller than the diaphragm. The compression that then occurs as the diaphragm radiates significantly improves coupling to the air, and consequently lifts the radiation efficiency. The aperture forward of the diaphragm creates the mouth of the horn, which then also progressively helps match the diaphragm impedance to the air, while at the same time defining the overall directivity.

It's not only the S360 tweeter that appears inspired by a PA driver technology. The S360 bass/mid driver also incorporates »

Genelec S360 $9190

PROS

• Seductive combination of volume, bandwidth, clarity and tonal accuracy.

CONS

• None.

SUMMARY

The S360 seems something genuinely different from Genelec. Its design was driven by a contemporary demand for compact, multi-channel immersive monitoring, but the result is a brilliantly capable monitor that would be able to succeed in almost any monitoring role.

Diagram 1. S360 connection and configuration

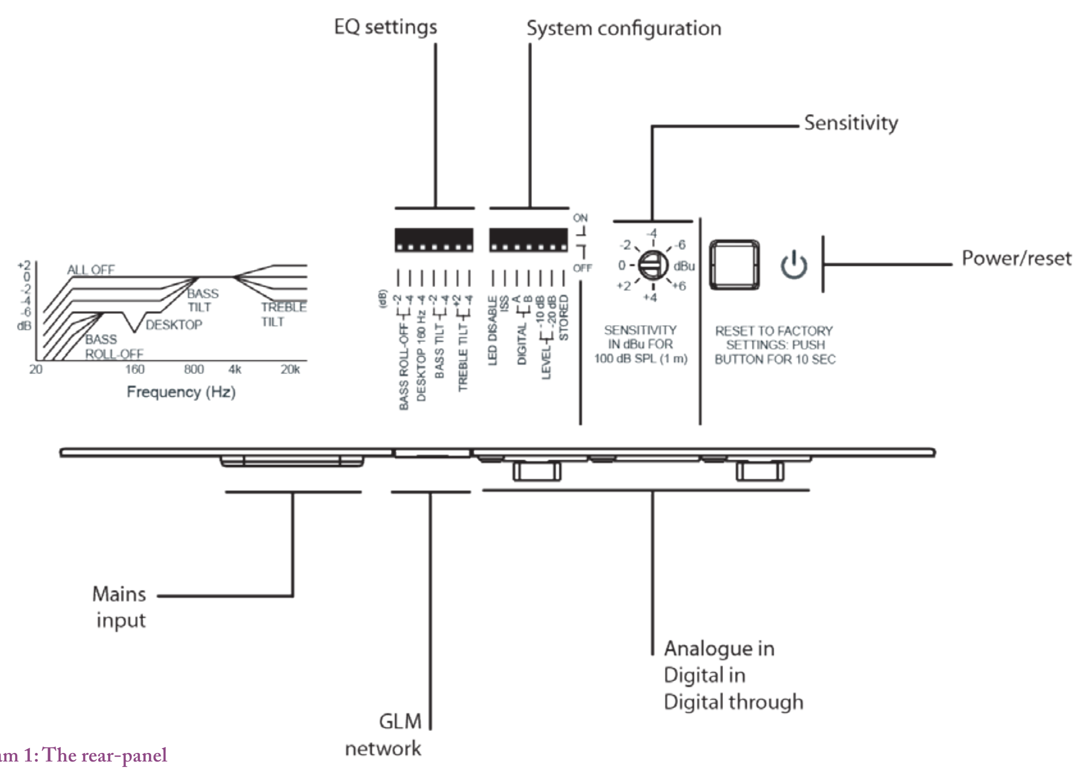


Diagram 1: The rear-panel controls and connectors.

» features more usually seen in speakers designed for live sound. Firstly, its 250mm nominal diameter is large for a unit required to reach significantly up into the mid-range, and secondly, its coated-paper diaphragm and pleated roll surround are features traditionally aimed towards maximising sensitivity and level, sometimes at the expense of response linearity and coloration. The use of a PA-style bass/mid driver in the S360 is the second time recently that I've seen such a thing in a speaker system designed for low coloration and high tonal accuracy, the other example being the Finkteam Borg hi-fi speaker: www.finkteam.com/borg. I wonder if advances in computer modelling of diaphragm behaviour are enabling diaphragm materials and profiles that were previously considered as only good for less demanding PA applications to become viable for high-accuracy monitoring?

Even with an extremely high-performance 250mm bass/mid driver, however, "You cannae change the laws of physics, Jim," and a fundamental consequence of the driver's size is that it will become noticeably directional at a relatively low frequency. So if the system of bass/mid driver and tweeter is not to have a significant off-axis dip in its frequency response, which wouldn't be a good thing, the crossover frequency between the two needs to be set at a frequency below that at which the dispersion of the bass/mid driver narrows. The crossover frequency on the S360 is therefore down at 1400Hz, which is an octave or more below that of more traditional two-way speakers — and which, with a conventional direct-radiating tweeter, wouldn't be feasible. A low crossover frequency is well within the realm of the S360's horn-loaded compression tweeter, however.

Another useful consequence of the S360 tweeter's waveguide loading is that its widest dispersion is defined largely by the dimensions of the mouth, and it's no coincidence that said mouth is around the same size as the bass/mid driver diaphragm — which mean that, around the crossover frequency, the dispersion of the tweeter and bass/mid driver will be similar. All the pieces of the jigsaw fit together to give the S360 both the high volume-level capability and the well-controlled and relatively narrow dispersion it needs to meet its intended applications.

Pass The Port

I've touched on the techniques employed in the S360 to manage its dispersion and to provide high volume-level capabilities, but written nothing so far about bandwidth. When 'bandwidth' is mentioned in the context of speakers it's almost always in reference to low-frequency extension, because that's fundamentally defined by some rather noticeable electro-mechanical parameters — things like enclosure size, bass driver diameter and amplifier power. But low-frequency bandwidth is also partly defined by the bass loading technique chosen for the system: reflex, closed-box, and transmission line being the three most common. For the S360, Genelec have chosen the reflex-loaded option, with suitably profiled ports exiting on the underside of the enclosure. Bearing in mind the intended applications envisaged for the monitor, and the need to keep it relatively small and affordable, reflex was, I suspect, the only viable option. »

Subwoofer

The 7382 is Genelec's largest and most powerful subwoofer. It employs three 380mm (15-inch) drivers, driven by an external amplifier rated at 2.5kW, in a reflex-loaded enclosure of around 400 litres internal volume. The -6dB low-frequency cutoff is at 15Hz and the system can achieve 129dB SPL at 1m between 30Hz and 85Hz. The 7382's group delay exceeds 50ms at 20Hz; however, where I think it matters most (in the kick drum/bass guitar band above, say 40Hz), it's well below 10ms. The 7382 weighs a trifling 145kg.

The 7382 subwoofer uses three 15-inch drivers and is driven by an external 2.5kW amp.

» I've written about the inherent time-domain characteristics of reflex-loaded speakers many times in these pages, and have argued that, for nearfield mix monitoring duties in particular, the minimal inherent low-frequency group delay (the frequency-dependent latency inherent to an analogue filter) offered by closed-box loading is a significant benefit. To put some numbers on this, closed-box nearfield monitors typically have group delay of around 5ms below 100Hz. In contrast, the S360's published group delay (Genelec are to be applauded for publishing group delay numbers) is around 10ms at 80Hz, rising to around 23ms at 50Hz.

Clearly, given that I'm not in the position of designing monitoring systems that need to work in their intended environment and application, it would be easy for me to write that every monitor should be employ closed-box loading to minimise group delay, or perhaps incorporate DSP-based time-domain correction in order to compensate for it. But a conversation with Genelec's Technical Director, Aki Mäkivirta, about the S360 and reflex loading illustrated perfectly why a closed-box or even a group-delay-compensated S360 wouldn't really work. Firstly, the S360's combination of bandwidth, enclosure size, amplifier power and cost target simply doesn't fall within closed-box territory. One or more of those parameters would have to be significantly modified, and then the S360 would no longer meet its design brief. And secondly, despite the S360 being very much a digital product that brims with DSP power, equalising in the time domain to compensate for LF group delay would unavoidably increase the wide-band latency of the entire system to somewhere around 30ms, which, in a multi-channel and quite possibly audio-visual monitoring environment, isn't workable. In a genuinely practical sense, the S360 is toe-to-toe with the laws of physics, which, as I mentioned earlier, "you cannae change".

One of the things you can change on the S360, however, is its configuration. In common with other Genelec models, the S360 rear panel is well populated with connection and control sockets, and DIP-switch configuration options. This is illustrated in Diagram 1.

Signal connection to the S360 is by either balanced analogue XLR or digital AES-3 XLR. A digital 'thru' output socket is fitted to enable signal daisy-chaining. Alongside the signal inputs is an RJ45 socket for GLM (Genelec Loudspeaker Manager) network connection, and finally, next door to that is an IEC mains input socket. Located just above the sockets are two banks of seven DIP switches. The left-hand bank offers EQ configuration including 0dB, -2dB and -4dB low-frequency roll-off options, similar low-frequency shelf options, a desk-mount EQ contour, and finally +2dB, 0dB, -2dB and -4dB high-frequency shelf options. The right-hand bank of DIP switches offers LED illumination options, automatic power saving, digital channel selection, level (0dB, -10dB, -20dB, -30dB) and finally an option either to select the rear-panel DIP switch configuration for the monitor or, if it's connected to a GLM network, to load the stored configuration and room-compensation EQ. Alongside the right-hand DIP switch is a rotary knob offering ±6dB fine control of input sensitivity, and a power/reset button.

London Calling

In light of the 'midfield' size of the S360 and its intended high-level monitoring duties, my usual practice of listening to review monitors on either side of my DAW in my home studio room was not really appropriate. So Genelec kindly gave me the opportunity to hear the S360, along with the Genelec 7382 subwoofer (see box), in the Studio C control room at London's Metropolis Studios. Of course, listening in an unfamiliar environment without access to known monitoring references is not the same as listening in one's own space, especially as my usual practice with review monitors is to combine concentrated listening with

Genelec Loudspeaker Manager

All Genelec's SAM (Smart Active Monitor) products, including the S360, are able to benefit from optimisation and system configuration provided by the GLM (Genelec Loudspeaker Manager) application. There are two elements to GLM. First there's the 'management' side of things, probably most useful on larger multi-channel installations where monitoring configurations for different user preferences or listening positions need to be setup, stored and quickly recalled. GLM can manage systems of up to 45 individual SAM monitors and subwoofers. The second element of GLM is room compensation, which, in conjunction with a custom measurement mic and USB interface, uses response measurements taken at a range of positions to calculate room-correction coefficients for each monitor in the system. GLM then sends the coefficients to each monitor as appropriate, where the monitor's internal DSP engine translates the coefficients into EQ curves for each driver. GLM, like the similarly conceived Sonarworks, IK Multimedia ARC and Trinnov systems has, to my mind, its pros and cons. The top-line pro is that, at the listening position, the subjective frequency response can be made more linear (or made to hit a specific response profile). The con is that the monitor frequency response away from the optimised listening position can become bent noticeably out of shape, especially if the room is acoustically poor to begin with. There are also potential issues of monitor overload, especially at or below the port tuning frequency, where a room EQ curve may demand levels from a monitor that it is simply unable to deliver.

Alternatives

For slightly differing reasons, the S360 and 7382 subwoofer stand somewhat out on their own: the subwoofer because pretty much nobody else makes a sub quite so ambitious, and the S360 because it's really aimed at a new niche. However, if you were to consider the S360 in a nearfield/midfield monitoring role (in which I think it would excel), three similarly ambitious monitors you'd also probably want to hear are the **Kii Three**, the **Neumann KH420** and the **Unity Audio Boulder**.

continual use for background listening, but I think it's still possible to make value judgements about the sound and to form an opinion on a monitor's success or otherwise.

I spent around three hours listening at Metropolis to a range of familiar material from CDs and Pro Tools sessions, sometimes with the S360s alone and sometimes in conjunction with the 7382 subwoofer. In addition to hearing the S360 with and without the subwoofer, I was also able to hear the system with and without Genelec's GLM monitor control and calibration software (see box).

My first impressions were of an extremely revealing monitor that holds absolutely nothing back in terms of detail and clarity. With the subwoofer operating, as perhaps is to be expected considering its size, the system's low-frequency bandwidth was to all practical purposes unlimited — and that's how it sounded. There was an utter lack of any sense of low-frequency mix elements missing or being curtailed. With the S360 alone, things returned to a more normal listening experience but still with no great lack of low frequencies, and still with that same sense of clarity and ability to offer great insight into the architecture of a mix and the character of each mix element. The S360 is not really intended as a traditional stereo mix monitor, but it undoubtedly has the necessary abilities.

Conclusions

Monitors that provide great clarity, like the S360, can sometimes also seem tonally cold and unrelenting, and part of me was expecting to hear that sort of character from the S360 — especially in the context of its paper bass/mid driver diaphragm and compression tweeter. But that turned out not to be the case at all. One favourite piece on CD that I took along to Metropolis, and which perfectly illustrated the S360's tonal character, is an old Nimbus recording of Benjamin Britten's 'Variations on a Theme of Frank Bridge'.

> "The strings were rich and warm, yet still extravagantly detailed, and perhaps even more remarkably, their seductive character didn't waver as the volume rose."

It's a fabulous recording of a fascinating piece, but sometimes it can verge towards the uncomfortable in its string tonality. Heard over the S360s, however, the strings were rich and warm, yet still extravagantly detailed, and perhaps even more remarkably, their seductive character didn't waver as the volume rose. Stereo imaging, too, was spectacular and I could have listened to that CD all day. Tonally, the S360 is a remarkably smooth-sounding monitor with very few, if any, rough edges. With the GLM room compensation switched off, a little of the S360s' smoothness seemed to dull, to reveal a little more speaker-borne character. The sound became just a little less precise, but still the fundamental quality of the S360 shone through and, again, the volume level didn't seem to matter. It was the same, whether quiet or really quite loud.

With or without GLM room compensation, the smoothness of the S360 continues up into the high frequencies where there appears absolutely no sign that the tweeter is a compression driver. The tweeter is smooth and detailed, and down at the other end of the band, with the 7382 subwoofer muted, the S360 seemed relatively dry in its bass character, with no obvious tell-tale reflex port effects — although of course it's at low frequencies that an unfamiliar room acoustic is at its most, well, unfamiliar. However, another favourite CD I took out for the day was Joe Jackson's *Body And Soul*, and the S360s did every possible justice to its spectacularly rich and resonant bass. *Body And Soul* is an early digital multitrack recording from the mid-1980s. It's a predominantly live recording with a seriously great band set up in an old Masonic hall in New York that turned out to have, I think, one of the finest acoustics for rock & roll that has ever been recorded. As with the Benjamin Britten CD, *Body And Soul* played right into the strengths of the S360: dynamics, volume, detail, clarity and a fabulous portrayal of that amazing recorded acoustic.

Using the S360 in a traditional stereo monitoring role is of course not really the application that they were primarily developed for. On the basis of the few hours I spent with them in that mode, however, there's little doubt in my mind that they could take almost any role in their hugely long stride. The S360 is very much a hit.

$ S360 $9190 per pair, 7382 subwoofer $11,995.
T Genelec +1 508 652 0900
E genelec.usa@genelec.com
W www.genelec.com

JoeCo Cello

USB Audio Interface

JoeCo's debut desktop interface combines high sound quality, robust engineering and some unusual features.

SAM INGLIS

Former SADiE developer Joe Bull has a knack of spotting gaps in the market, and launched JoeCo to develop multitrack recorders for capturing live shows. Rugged, super-reliable and compact, JoeCo's BlackBox Recorders come with a range of I/O options that allow them to hook into mixer insert points, digital outputs or mic splitters, and have become the *de facto* industry standard in some quarters.

The BlackBox Recorders are stand-alone devices, so it was a logical development to launch the BlueBox Recorder. This too can record and play back 24 tracks of audio to and from local media, but adds, among other things, the ability to serve as a USB audio interface for Mac OS and Windows. I reviewed the BBRW24MP in *SOS* May 2017; considered as an interface for studio use, it makes an intriguing alternative to the familiar choices.

Emboldened by that success, Joe and his team have now created the Cello, a desktop USB interface designed for home and studio use. On paper, its feature set looks much more conventional than that of the BlueBox, boasting two mic preamps, a high-impedance input for guitars, a pair of headphone outputs and monitor control functionality amongst other things; but, as we'll see, there are still respects in which the Cello differs from its peers.

Cello Case

One aspect of the Cello that immediately marks it out as different is its industrial design. It occupies a chunky brushed-metal housing and offers hands-on control courtesy of chunky retro knobs and attractive 'Opal fruit' buttons with integral LEDs. Further visual feedback comes from six-segment LED ladder meters embedded into the top panel, and a two-line text display that looks as though it belongs on a sampler from 1989. Taken as a whole, I think it works well, making a bold visual statement whilst feeling very robust and substantial. My only slight reservation is that the buttons don't feel as good as they look.

The tiny on/off switch on the Cello's rear panel is an unusual momentary affair rather than a conventional toggle, but the unit itself leaves you in no doubt as to whether it's powered up. In theory, the Cello can run on bus power, but an additional 5V power supply is included in case your computer can't supply enough juice. My MacBook Air failed dismally on this front, and I was only able to use the Cello with the external PSU. This is inconvenient, as it's a wall-wart affair that comes with a cable only 1m long.

In this day and age, JoeCo's decision to include five-pin MIDI in and out sockets is unusual (and welcome) in itself. These are on the left-hand side panel, while the audio I/O is divided between the front and rear. The front sports two Neutrik Combo mic/line sockets, plus the quarter-inch instrument input and two quarter-inch headphone jacks. On the rear panel, meanwhile, we find the main stereo monitor outputs, again on quarter-inch jacks, plus the line-level and digital I/O. The latter comprises two optical inputs, catering for up to 16 channels of ADAT-format audio, and a pair of coaxial sockets that can be switched between S/PDIF and word-clock in and out.

The Best Of Both Worlds

Most interesting, and unusual, are the line-level analogue I/O arrangements. There's a pair of balanced quarter-inch inputs, plus a further pair of balanced jacks that act as insert sends from the two front-panel mic channels. Using the inserts deprives you of the separate line inputs, because these become insert returns, but the big plus is that the unprocessed mic signal is still available as a recording source in your DAW. It's thus possible to connect an outboard compressor or EQ as an insert, then to monitor and record this signal through the line inputs, while also recording an uncompressed 'safety' copy of the mic input to a separate track. I like this setup a lot, and I can't immediately think of any other interface that works this way. (It would be even better if the mic signal and insert return were perfectly phase-aligned; as it is, the latter seems to be delayed by a single sample, at base sample rates.)

The sharp-eyed will notice that there are two types of position indicator on the top-panel knobs. The mic preamp and instrument gain controls are conventional analogue potentiometers with a fixed travel, while the headphone level controls, like the main monitor volume, are endless rotary encoders (which begs the question of why they have position indicators at all). Input gain control is actually split between the analogue and digital domains. All the mic-input buttons, including the pad, are reproduced in the JoeCoControl utility; and the line inputs also have a preamplifier which has only digital gain control. Curiously, the variable gain runs from +20 to +40 dB in half-decibel steps, and if you want the line inputs to operate at unity gain, you have to use the lowest setting and engage the -20dB pad.

Levels for the two line inputs can be set separately in software, but there's also a top-panel encoder that adjusts both simultaneously, preserving any offset between them. The same encoder is also pressed and held to activate the built-in talkback (latching mode is available only in software), an arrangement which runs the risk of the engineer accidentally changing the line input gain whilst attempting to communicate with the musicians. The built-in talkback mic is located right next to the encoder, and consequently transmits an unpleasantly loud 'thunk' when this is released. On the plus side, it's available as a source for recording, allowing you to slate takes and so forth.

The two headphone level controls also have a press function, in this case relating to the mic preamps. By default, these have a flat frequency response, but JoeCo's Top+ option lets you add extra sparkle courtesy of a fixed shelving EQ boost turning over at 10, 12, 14, 16 or 18 kHz. Though it's controlled digitally, this operates in the analogue domain, as does the high-pass filter; this is as it should be, since one of the main reasons for putting a high-pass filter in the input chain is to protect the A-D converter from being overloaded by subsonic thumps and rumbles.

My tests did uncover some issues with polarity not being preserved across all the inputs and outputs, but JoeCo said they expect to be able to resolve this in a firmware update which should already have happened by the time you read this. Minor quibbles about the talkback aside — and it's great to have talkback, which is a glaring omission from many interfaces — the whole system is pretty comprehensive and well thought-out, though I can't quite fathom why JoeCo haven't put all of the input gain settings under digital control rather than just some of them.

JoeCo make available comprehensive specifications for the Cello, and these demonstrate a uniformly high level of performance. The mic inputs can apply up to 78dB gain, with EIN quoted as -127dBu unweighted, while dynamic range on the line inputs and outputs is 120dB and 127dB respectively. Most of the analogue circuitry also boasts a frequency response that is said to be flat up to the Nyquist limit — which, since the Cello is one of very few audio interfaces that can record at 354.8 and 384 kHz sample rates, could be as high as 192kHz! Interestingly, lurking in the control panel software are options for 'ADC Setup' and 'DAC Setup'. These change the shape of the anti-aliasing and reconstruction filters, with a choice of options that range from 'Maximally Flat' to 'Musical' and 'Ultra Musical' at the other. Who doesn't want their conversion to be ultra-musical?

Cello Control

An attached computer 'sees' 24 inputs and eight outputs from the Cello. None »

JoeCo Cello $1125

PROS

- Sounds very good, with high-quality preamps.
- Incorporates some unusual features such as 384kHz recording and the ability to choose different filter shapes for the A-D and D-A converters.
- Insert points allow processed and unprocessed signals to be recorded to separate tracks.
- Rugged build quality.
- Built-in talkback, with comprehensive additional monitor control available in software.

CONS

- Unspectacular low-latency performance.
- External PSU cable is too short.
- Talkback button transmits mechanical noise to the talkback mic, and only operates in momentary mode.

SUMMARY

JoeCo have brought fresh thinking and some impressive engineering to the world of USB audio, creating a desktop interface that stands apart from the crowd.

The Cello's back panel houses connections for USB, an external PSU, monitor outputs, S/PDIF, word-clock and ADAT I/O, line inputs/insert returns and insert sends. The unit's full-size MIDI ports are found on the left-hand side.

of the inputs showed up with descriptive names in Pro Tools on my system, but it's easy enough to figure out that 1/2 are the mic inputs, 3 is the instrument jack, 4 is the talkback mic, 5/6 are the line-inputs-cum-insert-returns, 7/8 are the S/PDIF ins and the rest are all ADAT ins. By contrast, physical outputs are not addressed directly: instead, the eight outputs available in software show up in the Cello's internal DSP mixer as four pairs of stereo 'stems'. These, along with direct signals from the inputs, can be mixed in any combination to the main outs, the two headphone outputs and the S/PDIF outputs.

This is achieved using the same JoeCoControl utility that is supplied with the BlueBox Recorder, slightly tweaked to reflect the different purpose and I/O configuration of the Cello. The main difference is that the Live and Mix modes have disappeared, which is sensible enough, since they are not really relevant to typical Cello applications. (You also lose some of the inputs and outputs at 354.8 and 384 kHz, which I think is unlikely to bother anyone much.)

JoeCoControl is likely to see much more intensive use with the Cello than would typically be the case with the BlueBox Recorder, so it was with some relief that I installed the v1.2.0.0 update that became available around Christmas 2018, because this brought with it two features that make a huge difference to its usability. The first is the ability to hide the digital input channels when you're not using them — which, with this sort of desktop interface, is most of the time. The second is the ability to freely resize windows. For some reason this doesn't yet apply to the 'parent' window, which remains minuscule, but as before, all of its components can be floated as separate panes, and these can now be made as large as you like. Another new feature introduced in this update is a switchable Mid-Sides matrix for channels 1-2. This operates in the Input pane, so if you connect your Mid mic to input 1 and your Sides mic to input 2, then engage the matrix, what you get in the rest of the mixer and your DAW will be conventional left-right stereo.

As on the BlueBox Recorder, one of the best aspects of the Cello's built-in mixer is the master section, which includes not only the talkback, dim and monitor control available from the top panel, but additional functions such as mono, independent left and right output muting, and the ability to designate either headphone output as the PFL bus destination in place of the main outs — unusually, there are separate PFL and solo buses. There is potential for further improvement on the usability front — at present there's no way to link adjacent mixer channels, nor any easy way to make a channel's pan control jump to the centre — and JoeCoControl remains cheerfully idiosyncratic in its design, but it is clear, functional and succeeds in fitting an awful lot of control into a compact interface.

Cello, Is It Me You're Looking For?

Although it's not in the premium price bracket occupied by desktop interfaces such as the Prism Sound Lyra 2, the Cello is far from being a budget product. A glance at the specifications might suggest that there are rivals offering similar performance for less: Arturia's AudioFuse springs to mind, for example, as does RME's Babyface Pro. Thanks to its reliance on generic USB drivers, meanwhile, the Cello's low-latency performance can't match that of the custom drivers developed by RME, MOTU and Focusrite.

As a result, there are probably three things that will cause people to buy a Cello rather than one of its competitors. The first is its subjective sound quality, which is excellent all the way from the high-quality mic preamps to the headphone amplifiers. Second is its physical construction, which is both smart and seriously rugged; I would back the Cello to survive life on the road longer than most interfaces in its class. Finally, there are the unique and unusual features of JoeCo's design, such as the choice of converter filter shapes, the distinctive insert arrangements and the ability to capture audio at octo sample rates. This last might not be very relevant for conventional music recording, but it will be of keen interest to those sound designers who like to record ultrasonic sources and pitch them down into the audible frequency spectrum. And I think anyone who has a 500-series rack, or a couple of outboard units, will enjoy the flexibility of being able to split a mic input and record both processed and unprocessed signals. With the Cello, JoeCo are offering a characteristically individual take on the desktop audio interface, and the factors that mean it won't suit everyone will also make it perfect for some.

Latency

The Cello is a class-compliant USB audio device, and on Mac OS, uses Apple's built-in Core Audio driver. As ever, this offers reasonable but unstellar low-latency performance, and the Cello is also burdened by the small amount of delay added by its build-in DSP mixer. At the lowest 32-sample buffer setting, Reaper reported a round-trip latency of 6.7ms; a loopback test showed that the true figure was actually a few samples lower. JoeCo's specs give a detailed list of round-trip latency measurements at different sample rates and buffer sizes relating to operation under Windows, and here, the lowest achievable round-trip latency at 44.1kHz is given as 10ms with a 32-sample buffer. On the plus side, this apparently falls to under 4ms if you record at 384kHz...

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BOB THOMAS

Overstayer 8755DM

Stereo Modular Channel

Overstayer's versatile stereo processor offers endless ways to manipulate — or obliterate — any input signal!

Digital audio virtually eliminated the problem of unwanted distortion, but in doing so, it probably reminded us that distortion can also be musical — and there are now many tools designed specifically to create such distortion, among them Overstayer's 8755DM Stereo Modular Channel.

Staying Over

Founded by musician, engineer and producer Jeff Turzo, Overstayer have created a range of compact, high-quality analogue recording and mixing processors. This latest unit is a stereo mic/line/instrument channel with a unique combination of features and facilities, incorporating some functionality from their existing microphone channel, VCA compressor and analogue saturation and distortion processors.

Apart from the separate left and right input gain controls and polarity switches, the 8755DM's two channels share a single set of controls; it's intended for stereo applications. Nonetheless, the sheer number of switches and vintage-style knobs, and the minimal metering, make it essential that you make an effort to understand its signal flow and constituent modules before you dive in. The first clue that the 8755DM is not your average stereo channel comes when you look at the I/O options on the rear panel, where you'll find XLR connectors for its two transformer-balanced microphone inputs, two sets of balanced line inputs, a set of balanced preamplifier outputs, balanced sends and returns for the left and right channel effects loops, and the left and right channel main outputs. In addition, there are quarter-inch jacks for

the left and right channel instrument-level inputs, external side-chain and control voltage (CV) inputs for the VCA compressor, and separate left and right CV inputs that control the output level of the 8755DM's saturation section, of which more later.

Input Routing

Above the master input level control, you'll find a row of three switches. Selecting which input goes where involves the first two of these, plus another unlabelled, horizontally oriented 'floating' switch at the top of the Bandwidth Control section. The first switch selects which of the first three inputs (Instrument, Mic or Line 1) is routed through the main channel path. The second determines the source entering the mic inputs' balancing transformers, from the microphone input (Mic), the microphone input with a -20dB pad (Mic-20) or the fourth and final input (Line 2). Finally, the 'floating' switch overrides the second switch, routing Line 1 directly to the input of the mic transformer. Since, in this configuration, the line inputs replace the microphone XLRs, the first switch must be in its Mic position to route the selected Line inputs down the main channel path. A serendipitous result of the switching complexity is that the signal passing through the gain controls is always available at the preamp outputs. Another completely logical and very useful situation arises when Line 1 is selected on the first switch and the 'floating' switch is set to the left, at which point you'll find that the preamp outputs are carrying the signal from the microphone inputs, which remain active unless replaced at their transformer inputs by Line 1 or Line 2. The final three switches in the input section activate the filter (Bandwidth Control), the EQ (Frequency Control) and the compressor (Amplitude Control).

Filters & EQ

Under the front-panel heading of Bandwidth Control, you'll find individual high-pass (20Hz to 4.7kHz) and low-pass (220Hz to 22kHz) filters. Each has frequency and peak (resonance) controls, and self-oscillates at higher resonance settings. Like almost all the other rotary controls, these are marked only from 0 to 10 with »

Overstayer 8755DM
$2995

PROS
- Ranges from pristine master-bus subtlety to distortion-laden mayhem.
- Great-sounding mic preamps.
- High- and low-pass resonant filters.
- Musical EQ.
- Unique compression Behaviour control.
- Superb harmonic saturation and hex-inverter distortion.
- Flexible parallel/series processing options.
- Works well with modular synths.

CONS
- Any unit as complex as this presents something of a learning curve!

SUMMARY
Overstayer's impressive Stereo Modular Channel is capable of audio performance ranging from the clarity and subtlety required of a master bus processor to sledgehammer levels of saturation and harmonic distortion, via every shade in between — it's truly a unique and very appealing combination of features and facilities.

The sheer number of analogue sockets on the 8775DM's rear panel tells you just how complex and versatile this device is!

» four-division increments, so your choices of frequency and the amount of resonance will — as with every other continuously variable parameter on the unit — be primarily listening-based. The final control in this section is the horizontally oriented Curve switch. When active, this increases the low-frequency headroom in the harmonic saturation and distortion processes, in order to preserve power and dynamics in the low bass.

The three-band Frequency Control (EQ) section follows, with shelving bass (±18dB) and treble (±15dB) controls offering corner frequencies at 50/150/300 Hz and 5/10/15 kHz, respectively, plus a wide-ranging swept bell-curve Presence band (±12dB 260Hz-12kHz). This has a proportional-Q response that narrows the bandwidth at higher boost or cut levels.

Sitting on either side of the bottom edge of the Frequency Control section (but having nothing to do with it!) are two switches and two green lenses. On the left, the insert switch activates the 8755DM's effect loop, and the lens next to it illuminates when the unit is powered up. The switch on the right controls the relay-switched hard bypass, with the green lens next to it going dark when that bypass is engaged. Sitting on the top edge, directly above these switches, are the 8755DM's only meters: two four-LED ladders that display the amounts of saturation and compression.

Compressor

The Amplitude Control contains a variant of the stereo VCA compressor found in Overstayer's Stereo Voltage Control peak limiter/compressor. Attack and release times are set either manually, using two three-position switches (Fast, Medium, Slow), or by engaging the compressor's RMS detector side-chain (a pull switch on the threshold knob) to obtain programme-dependent timings. The threshold control is scaled from 10 to 0 so, once again, it's your ears that will be used to set the actual threshold. There's no ratio control as such, but Overstayer's unique Behaviour control is an equivalent, if somewhat unconventionally implemented, function that is capable of taking a signal from all transients to all ambience. In essence, this not only manipulates the compression envelope and its 'hardness', but also skews the ratio and timing. The make-up gain control also has a dual function, switching in an internal 220Hz side-chain high-pass filter when pulled out. A three-position switch allows you to set the source of the side-chain to be post-EQ (switch up) or pre-filter (switch down). In its unlabelled middle position, this switch activates the external side-chain input. If no external source is present, this effectively bypasses the VCA; since the make-up gain is active if the compressor is in circuit, you've got another source of gain (boost or cut) to play with.

Harmonics, Saturation & Distortion

Although the Drive knob sits amongst the compressor's controls, it's actually part of the harmonic amplifier, and its role is to set the amount of distortion delivered by the MAS, Sat and Hex stages. These stages are arranged in descending order of headroom and are switched in individually, so you can cascade the three together should you feel the urge to do so (I'm fairly certain you will!). The MAS (Multi Analogue Stages) stage allow you to add both second- and third-order harmonic character, and the peak-rounding limiting characteristic of vintage analogue recording chains to a signal. In *SOS* November 2016, I reviewed Overstayer's MAS 8101 Stereo Analogue Distortion Processor (https://sosm.ag/overstayer-mas-8101) and in that review you'll find more detail on the MAS process featured in this stage. Similarly, the Sat stage is derived from that in the NT-02A Stereo Analogue Saturator (reviewed by Matt Houghton, *SOS* May 2014: http://sosm.ag/overstayer-nt02a). The final Hex stage is an interesting one. It's based around a complementary metal oxide semiconductor (CMOS) hex inverter, which is a chip type that's usually associated with microprocessors rather than distortion amplifier stages. As its name implies, a CMOS hex inverter contains six inverters, each based on FET circuitry. When overdriven it produces a valve-like distortion — a quality that's often exploited by guitar effects-pedal manufacturers.

Mixer

In recent years, parallel processing has become *de rigueur* in both digital and analogue effects processors, but the 8755DM takes this concept further than any other audio processor I know of. It does this by providing a continuously variable mix of three discrete feeds, each with an individual mute switch: Dry, which can be pre filter, post filter, post EQ or post both filter and EQ; Compressor, which is a post-compressor feed that can be driven dry or by the Dry feed's post-filter/EQ

Alternatives

I know of no hardware that's quite like the 8755DM, but units such as the **Black Box Analog Design HG-2**, the **Looptrotter Emperor**, the **Thermionic Culture Vulture**, **Elektron**'s **Analog Heat**, **Amtec**'s 500-series **DST-5A** and, of course, **Overstayer**'s own **MAS 8101** and **Saturator NT-02A** can cover aspects of its performance.

options; and Sat, a post-harmonic amplifier feed that can be driven dry or by the Dry feed's post-filter/EQ options and/or the compressor. The final switched front-panel function engages something called an output Ceiling, which is a limiter that sits in the signal chain in between the three Feed faders and the final Output level control. The Ceiling is set so that, with A-D converters calibrated to -18dBFS, 6 on the output level control gives 7dB of headroom, 7 gives 4dB of headroom, and 8 gives 0.1dB of headroom.

Overall

As I've come to expect of Overstayer devices, the 8755DM delivers world-class sonic sculpting facilities. It sounds great and allows an almost bewildering degree of control, but it's also massively versatile, offering everything from master-bus subtlety to full on dirt and distortion. When you combine its superb-sounding mic preamps, it resonant filters, its powerful, well-configured EQ and its wide-ranging compression, with the harmonics generation, saturation and distortion, and the various series/parallel processing configurations, its qualities and subtleties as a high-end stereo channel strip come together to create something that's most impressive indeed. All the more so, in fact, given what is a very reasonable price, considering the quality and number of processing stages and the fact there are two channels of everything.

Since it can be so subtly effective, I'd have absolutely no hesitation in using the 8755DM across a master bus. When it's not doing that, I'd happily unleash its harmonics generation, filter resonance, EQ and occasionally crazy compression on drums, synths or guitars. Vocals would probably need to be treated with a much lighter touch, but it's capable of that too. And if you're into modular synthesis, or have a CV source, there's endless inspiration to be found in driving the compressor's side-chain from its CV input. Special mention has to go to the Curve function, though, as the reduced harmonic saturation and distortion in the low bass not only means it maintains its power and articulation, but also makes the mid-range and treble distortion appear subjectively clearer and more dynamic — a quality that can be enhanced by the CV-driven virtual fader between the Saturation and the Hex distortion. Finally, if, like me, you're a guitarist, I have to say that the 8755DM is, hands down, the best-sounding, most inspiring and most expensive distortion 'pedal' that I've ever encountered.

Up until now, I hadn't discovered a channel strip quite like the Overstayer 8755DM Stereo Modular Channel, and I certainly don't recall anything else that can act equally well as a subtle mastering EQ/compressor, a source of inspiring musicality and a sonic mangler of epic proportions. The 8755DM isn't going to be to everyone's taste, but for those who get drawn in by its web of flexibility, performance and potential there is, I fear, little chance of escape!

$ $2955.
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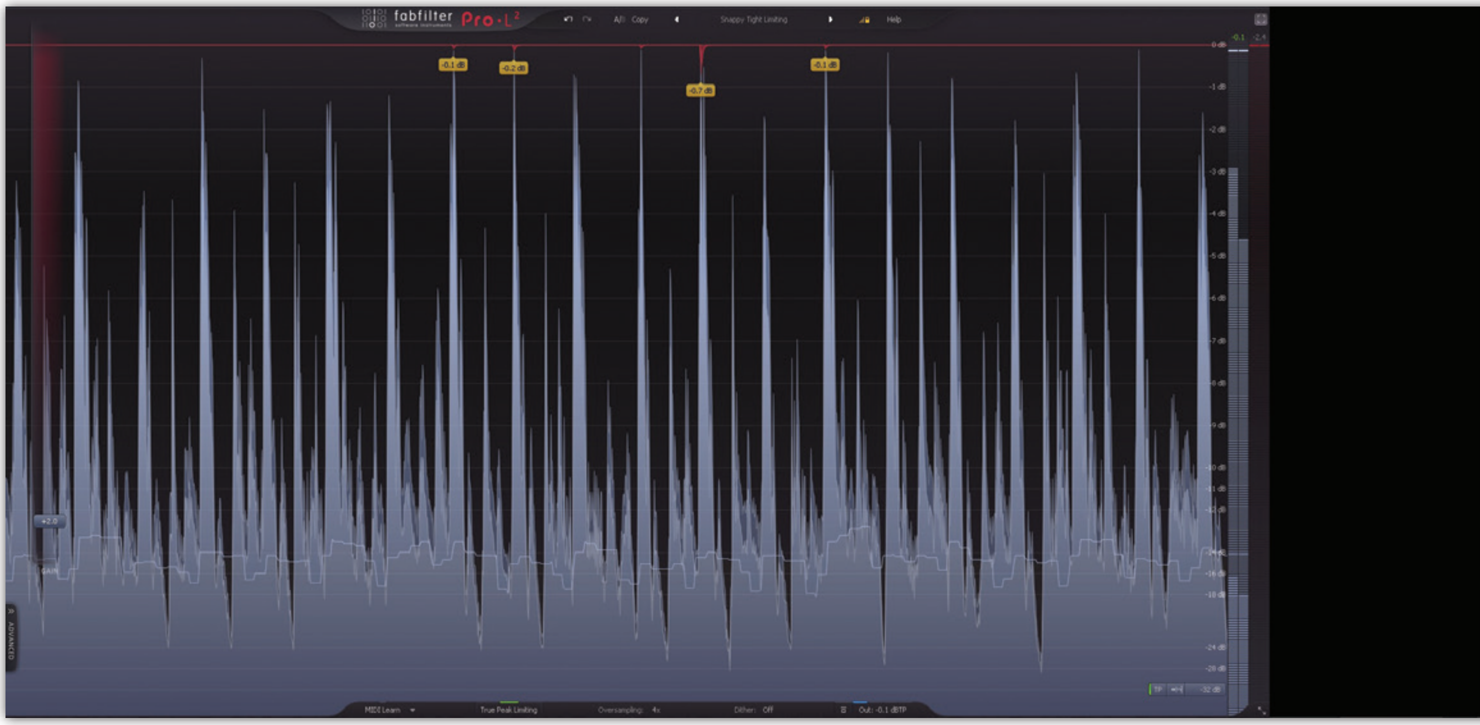


ERIC JAMES

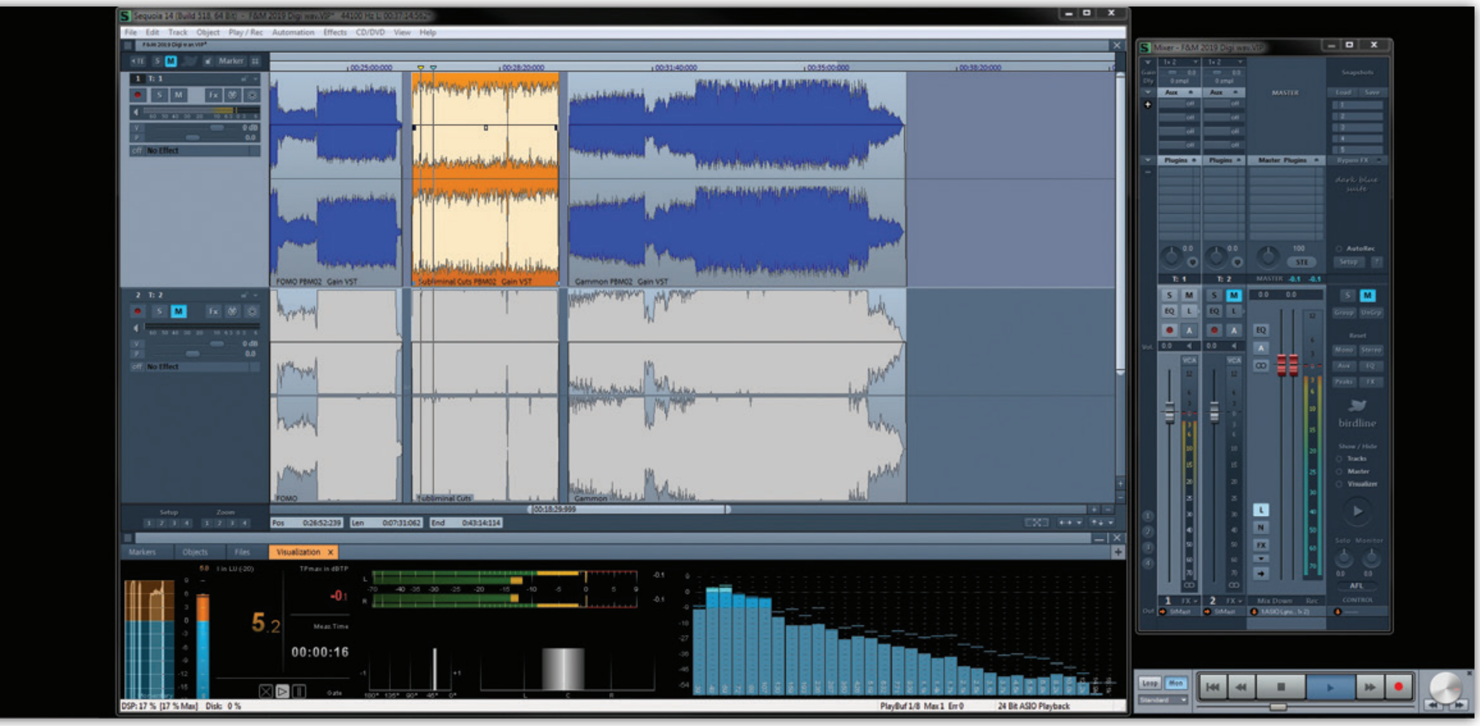
FabFilter Pro-L 2

Mastering Limiter Plug-in

FabFilter have reinvented their Pro-L limiter for the new world of loudness normalisation.

It's a sign of the times that reviews of software mastering limiters just four or five years ago would mainly have judged them on how well they made things loud, and yet today I am reviewing a limiter which I rate highly partly because of how well it allows excellent masters to be made quieter. I would be very wary of saying that the loudness wars are really, truly over and gone — there may well be pockets of resistance in far-flung places of the musical universe — but over the past few years there has been a definite general trend towards moderation, helped enormously by the resurgence of vinyl and by the loudness normalisation enforced by streaming services. In my own mastering facility we are seeing an increasing number of projects where the final delivery format is for streaming and vinyl only, with CD replication not the front-runner that it used to be. This means that the kind of demands made on a modern mastering limiter are no longer such that the loudest wins the day: the criteria and the market have become much more varied. My own short list of limiter desiderata would include: pristine sound, an intelligent ergonomical interface, tweakability when needed, and — combining all three — a set of basic limiting profiles from which to build personal preferences.

Limiting is one of the aspects of mastering that I look forward to the least. Having spent hours listening carefully on a superb monitoring system, adjusting EQ and compression for absolute sonic optimisation, perhaps using state-of-the-art analogue and digital equipment, the mastering engineer then has to sit down with a relatively inexpensive plug-in, and in the service of mere loudness, try not to undo any of the previous good work. Of course that's an exaggeration, and there are some kinds of music that seem more fully alive with the right kind of pumping and others than benefit from some mild limiter-induced tightness or excitement. When I first started mastering, the go-to 'pro' limiter — a digital outboard unit — was better at helping the first kind than the second. This was a shame, as my bread and butter work in those days was almost entirely classical, jazz and acoustic music, and I was surprised and delighted when around seven or eight years ago I discovered a limiter that could add the right sort of liveliness there, too. This was the original FabFilter Pro-L, which had preset a profile called 'Jazzy, Spacious, Dynamic and Loud' and which, with a bit of tweaking and just moderate levels of limiting, could actually give a positive



lilt to the kind of the projects I was then working on. I didn't use it exclusively, and over the past couple of years, more recent offerings have outgunned it in terms of flexibility and raw power — but then a few months ago, Pro-L 2 landed in my DAW.

There are a number of differences between Pro-L and Pro-L 2 which I will not be able to cover in this review for reasons of space, including features such as support for up to 7.1 surround formats, and external side-chain limiting. But what I will go through in use cases are all of those which the FabFilter Pro-L 2 manual describes as improvements "added to enhance the workflow" and the sonic benefits that follow from these improvements. These include new display modes, new algorithms, loudness metering, unity gain and audition limiting, true-peak limiting, higher oversampling and DC offset filtering. And I'll bring them in, roughly in that order.

On Screen

FabFilter have a well deserved reputation for creating user interfaces that are highly informative and bling-free, and in Pro-L 2, almost everything needed to complete the job is contained in a single clear presentation, with the relative proportion of screen real estate indicating the importance FabFilter apportion to the ergonomic need. Thus the real-time action of the limiter in both graphic and metered form takes up the most space, with the limiter parameters on a sliding tray, easily available for adjustment when needed and slid off-screen when not. The one-time choices appear in the upper and lower toolbars or, as the only hidden items, in a small row of tray-boxes at the bottom right of the main display. The rotary controls for parameter adjustment work very smoothly — either vertically or in rotation with the mouse, with precision level adjusted by speed or distance of the drag — with the mouse wheel, and by text entry, which is my own default. There is also a nicely flexible option to resize the whole window. New in Pro-L 2 are useful options to adjust the speed and scope of the the scrolling screen, but the most important new visual elements are the orange 'peak labels', which shout out the more significant peak reduction levels, and, above all else, the new loudness meter.

Pro-L 2 still offers 'normal' metering scales as well as those for the K-System protocol, but it is clear that the new loudness metering is now the main focus, and in the real world this is how it should be. The other scales, whatever their perceived benefits, were nonetheless optional: keeping your levels in the red zone of the K-scales went against recommended practice, and made using them a bit pointless, but that didn't have serious consequences further down the line, and very loud CDs still got made, bought, played and enjoyed. But streaming has changed all of that. ITU-R BS.1770-4 and EBU R128, which is based on it, are standards in the real sense: when they are in operation they are not optional and have to be adhered to, and failing to do so can have serious consequences, from really doinking up your streamed sound to having your masters rejected by production or broadcast companies. The Pro-L 2 manual is generally excellent, and on this topic it is superb: if you need a very brief introduction to these standards, or a quick refresher, have a look at pages 18-19.

For the purposes of the review, and in my real-life mastering work, I used Pro-L 2 »

FabFilter Pro-L 2 $199

PROS

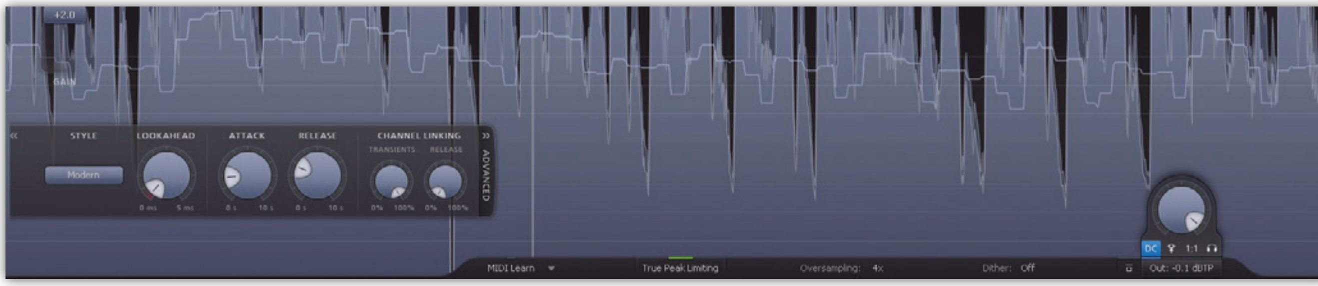
- Great sound.
- Excellent user interface.
- New features make adhering to loudness standards a doddle.
- Better than average manual.

CONS

- None.

SUMMARY

Redesigned with the new landscape of loudness normalisation in mind, FabFilter's Pro-L 2 is a pleasure to use and to listen to.



Advanced parameters are available in 'tray-boxes' that open out when needed.

in a five-stage process that put all of these new features through their paces:

- Establish a basic starting point: set gain and output level, choose meter time scale, engage Lockoutput, switch between presets and listen.
- Refining what I hear through unity gain and audition limiting.
- Refining what I want to hear by adjusting the Lookahead, Attack and Release times, and Channel Linking (transients and release) parameters.
- Ensure compatibility with streaming services by metering and controlling true peak values.
- Check for and eliminate DC offset and set up dithering.

Because the loudness standard requirements can be important, for streaming masters I think it's useful to start by setting the target level (for instance, -14 LUFS for Spotify and iTunes) first. CD levels are still ultimately a matter of taste and opinion, and the FabFilter manual suggests -9 LUFS, which would have been regarded as fairly conservative a few years ago — but it's entirely optional and any level can be chosen. The manual helpfully suggests a similar approach, except that FabFilter recommend choosing a preset first; however, as the target level will affect how the presets react to the music, I prefer to make this choice once the target level is established.

The Output Level setting specifies the maximum final output of the limiter, or 'ceiling' as it's sometimes known, so adding gain to a signal that hits this ceiling will increase the amount of limiting. Depending on the gain structuring during the mastering of the material to be limited, it could be that the streaming level of -14 LUFS will not require much additional gain. My own mastering chain is mainly analogue, but at the end of it, after the conversion back to digital, the final gain stage before capture is the brickwall limiter in a TC System 6000, which is set to an output threshold of 0dB, with only enough gain to control the occasional peaks. For most of the masters I used for this review, it then only took a single dB or so — very light limiting — to reach -14 LUFS. That being the case, it then required roughly 5dB of further gain to reach FabFilter's suggested CD level. This provides a graphic presentation of the differences between mastering for CD and mastering for vinyl, which can only be lightly limited, if at all — and probably explains why vinyl sounds better even when cut from a digital master.

Measuring LUFS gives a different number depending on what loudness time scale is chosen: the Momentary scale shows the current loudness level, but this is not generally as useful to know as the Short Term level, which has a wider time window, and the Integrated, which gives a reading for the entire track. Short Term is what I use for mastering.

Having established a LUFS target, a level of gain which starts to get me roughly in that area, and a loudness time scale, the next step is to audition some presets. Presets in Pro-L 2 are, in essence, different permutations of the Advanced Settings, of which more in a moment; a preset thus comprises one of the basic 'style' algorithms plus different settings of the Lookahead, Attack, Release and Channel Linking parameters. Before comparing them, clicking on the Lock Output button locks all of the limiter settings except these permutations and so enables a more level playing field for the listening comparisons.

Presets & Styles

The v1 presets are still available in Pro-L 2 (and for some reason it is still the original list that is described and illustrated in the manual) but the new version includes a lot more. The v1 presets were categorised according to music style, but the L 2 presets are arranged somewhat differently. A Basic category includes all the available limiting 'styles', while genre-specific and other presets are categorised as Loud, Moderate or Safe. It is worth noting that the preset titles are not prescriptive. I got some very nice results for a country dance band single using an only very slightly tweaked EDM preset!

Having a rough gain level and a preset, the result can be optimised by further tweaking. Pro-L 2 provides two features to help with making decisions on this. The first is Unity Gain, a setting enabled by clicking the 1:1 button in the Output tray. This works automatically to keep the listening level constant by reducing the output level as limiter gain is added, allowing the sonic effect of the settings to be more easily heard (this is actually a refinement of a feature from v1, which did the same but required extra keystrokes). The Audition Limiting setting goes one further, subtracting the original signal from the output to allow you to listen only to what the limiter is doing. This is really very useful; it was something I first heard in the original Tokyo Dawn compressors, but is now found in DMG's Essence and others.

Each preset comes with its own particular setting of the main limiter parameters, but unless you choose

The Manual

I don't normally discuss user manuals as part of a review, but bad experience recently with pretty insipid examples means I'm happy to shout out for credit where it's due. The FabFilter Pro-L2 manual is definitely a Good Thing. It's written to enable the user to operate the advanced controls "using good sense" and it does. It is also very informative about all of the options, with just enough detail to carry the gist of the theory and why best practice is best practice, without becoming a treatise, and at the end it includes a nice Myths and Facts section, which slays a few Internet-spread misinformation beasts.

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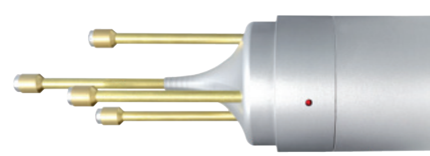

David Hachour (Mastering Engineer)

(Mark Ronson, David Guetta, Kanye West, Avicii)

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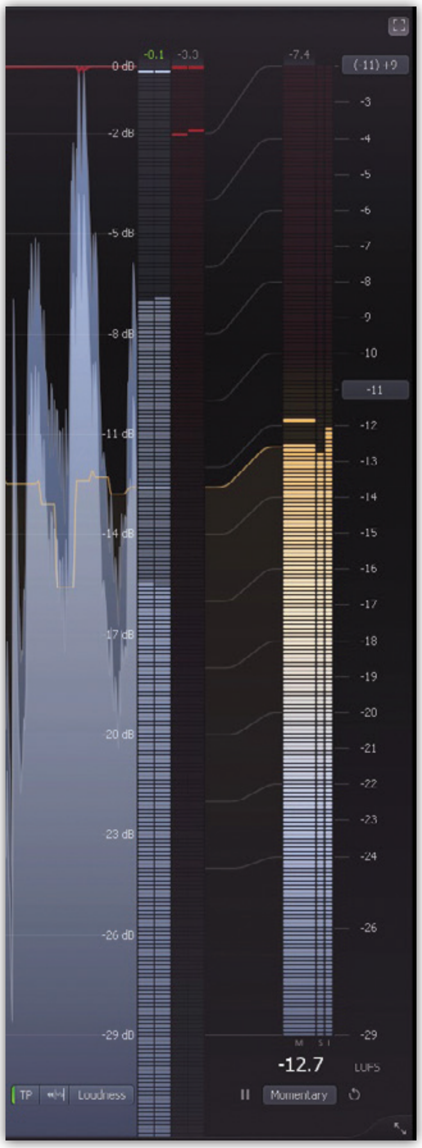

Key to level management for streaming services is the new loudness metering.

» the Safe style, these are completely user-adjustable, and if you don't like what you hear using either of the two above listening features, this is where you make things better. My experience suggests that if you've chosen a preset in the right ballpark, then adjustments here are likely to be minimal — they really are very well thought-out.

Like most modern mastering limiters, Pro-L has two stages of operation. The first, controlled by the Lookahead setting, acts on signal transients and is what keeps them from exceeding the ceiling. Shorter time constants in this stage generally give better results in terms of transient preservation, but setting them too short can create distortion and inter-sample peaks. Longer times are safer in terms of distortion and ISPs, but can smooth things over a little too much. Listening with Audition Limiting on can be a great help in the decision process here.

The second operation works more on the general dynamical profile of the material and on the release stage of the processing: a short attack time here means that the release from the reduction of stage one sets in earlier, and the longer the release time, the more effect the reduction has. In this stage, longer attack times make for more clarity, but the down side is again the possibility of introducing distortion.

The final limiter controls allow you to specify the independence of the two channels in terms of transient and release behaviours. Setting the transient link to a little less than 100 percent can yield increased loudness and clarity at the expense of possible shifts in the stereo image; I usually find that settings of 75 or 80 percent improve things a bit with no ill-effects. The release link I always left at 100 percent: I'm sure there are creative uses of this control, but none that occurred to me during the review period.

Practise Safe Streaming

Aggressive or heavy limiting, especially with very short lookahead times, can create inter-sample peaks — that is, the reconstructed signal can exceed the ceiling even though no individual sample does so. Again, the Pro-L 2 manual has a very good account of what this means. These ISPs can create problems during subsequent D-A conversion, but also in format conversion to MP3 and the like. Because of this, loudness standards specify a maximum true peak value.

Pro-L 2 deals with this problem, first by including a true peak meter to identify when such peaking occurs, and then by offering true peak limiting to eliminate it (even if those peaks were already present in the material prior to limiting). True peak limiting as implemented in L 2 is very effective, as is easily shown by turning on the TP meter, adding larger-than-life levels of gain, and then watching the TP count drastically reduce and disappear when the TP limiter is turned on. Unfortunately, though, asking the true peak limiter to do too much increases latency and at higher levels comes, to my ears, at a sonic cost.

The better way is to adjust the main limiter settings so that you are not generating high levels of inter-sample peaks in the first place, and there are two ways of doing this. One is to make the Lookahead time slightly longer, while the second is to use the Oversampling option. Pro-L version 1 offered up to four times oversampling, but Pro-L 2 supports settings up to 16 times — at the cost, inevitably, of increased latency and higher CPU load. There can also be a sonic cost: the filters used on the upsampled signal can introduce a softness and slight smearing of the sound. Oversampling is not uniformly implemented in digital processing, and some developers do it much better than others. I don't like the sound of upsampling in general and, where possible, turn it off in mastering (even in my monitor D-A converter, which offers upsampling even from 44.1kHz to DSD). However, the four-times option in Pro-L 2 seems to be one of the better implementations, and I found it both sonically invisible and sufficient to deal with almost all inter-sample peak issues.

Alternatives

DMG Audio's **Limitless** also sounds great and is endlessly tweakable and flexible, but by the same token, some might find it a bit daunting!

To The Limit

Using the workflow described in this review, I put Pro-L 2 to work for a couple of months working with a very wide range of material. It performed extremely well on everything I threw at it, and as I got used to working with it I found it to be one of the most efficient limiters in my small limiting arsenal in terms of time spent setting up, adjusting and using it. Only for the very demanding situations where the split-band offerings of a competitor were necessary did I find it in any way lacking. I had fallen out of the habit of using the original v1, but Pro-L 2 has now come back to take a central place in my work practice. The changes from v1 to v2 make it a thoroughly modern limiter: each of the improvements I've mentioned individually adds something to help the limiting process, and together, they make for a very effective and ergonomic workflow which means that the mastering engineer can spend less time limiting and more time on the music.

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Warm Audio WA73-EQ

Microphone Preamplifier & EQ

Find out how this recreation of a classic channel strip compares with the original.

ALAN TUBBS

While the mic preamps that are built into most modern audio interfaces work perfectly well most of the time, there are occasions when you might still want something a little more capable, or perhaps a little more colourful. Some, for example, don't have enough gain on hand to allow passive ribbon or dynamic mics to be used on anything other than drums and amps, or a good recording from a more distant condenser. Others don't respond too well when they're being overdriven — they sound clean all the way up to the point of clipping and then sound... well, awful!

Warm Audio WA73-EQ
$799

PROS
- A convincing 1073 recreation.
- Great price.
- Good build quality.
- Versatile analogue connectivity.

CONS
- Front panel a little crowded, and lettering rather small.

SUMMARY
With its keen price, high-quality components, hardware, and thoughtful design, the WA73-EQ is an excellent mic/line/instrument preamp that will be attractive for use in both home and commercial recording facilities.

If you find yourself in search of both colour *and* more gain, you could do worse than consider a preamp inspired by Rupert Neve's famous 1073. In 1970, Mr Neve developed a new module for the first 24-channel console to be installed in Britain — the Class-A 1073 preamp/EQ. This module helped to define the 'British sound', in part due to the sonic contribution of the Marinair Radar input and output transformers. The size, materials and windings of transformers all have an effect on the electrical signals passing through the metal, and therefore on the sound of an audio signal. As tastes changed, engineers who loved 1073s (and other Neve modules) began stripping them out of consoles and had them racked up with power supplies so they could take them anywhere, and this practice became even more popular as more audio engineers started doing serious work in home and project studios. The supply of sacrificial donor consoles available for mutilation and amputation naturally became increasingly limited, and various companies began issuing recreations of Mr Neve's design to meet the growing demand. But both the rackmounted originals and the boutique clones remained pricey. That's something that, thankfully, has changed in recent years, as a number of more wallet-friendly homages to the 1073 have come on to the market.

Warm Audio specialise in producing budget-friendly versions of iconic recording hardware, cutting as few corners as possible in the process, so perhaps it was inevitable that they'd release their own take on this particular classic. The result is the WA73-EQ, which, as the name suggests, is a recreation of the Neve 1073 mic, line and instrument preamp and features an EQ section too.

Not Just A Pretty Faceplate

Unpacking the review unit was more of a chore than I'd anticipated, largely because the WA73-EQ is so heavy! It comes in the form of a 1U 19-inch rackmount unit, and its reassuring weight is due to a combination of the thick-gauge casing, the large power supply, and the fact that the custom Carnhill transformers are not only fairly expensive models considering the overall asking price, but are hefty ones too. Why does the Carnhill name matter? Well, you'll most commonly read about Marinair-branded transformers being used in the 1073 (and other Neve units) but another company, St Ives Windings, was a supplemental supplier for the input transformers, and theirs had the same specs. Carnhill later bought St Ives.

The WA73-EQ also has plenty of knobs and buttons, which may not add much by way of mass, but certainly contribute to the classy overall feel. Dual-concentric potentiometers are used for the EQ knobs, for example, and frequency knobs are firmly stepped, with the cut/boost knobs offering the ability to fine-tune the

level at the selected frequency. The filter knob and input gain also snap as solidly into position as they should. The small output knob on the far right of front panel means you opt either for a cleaner sound, or to drive the unit quite assertively and still pass on a signal at a sensible level for the next unit in the signal chain. And with a total of 80dB of gain on hand, it's possible to capture soft voices or more distant subjects cleanly. The EQ is inductor-based, again adding to the weight, and inductors can attractively colour the sound too. And Warm Audio have provided some thoughtful features — for example, instead of the fixed 12kHz HF shelf of the Neve 1073 and many clones, we're treated to the greater range of options of the Neve 1084.

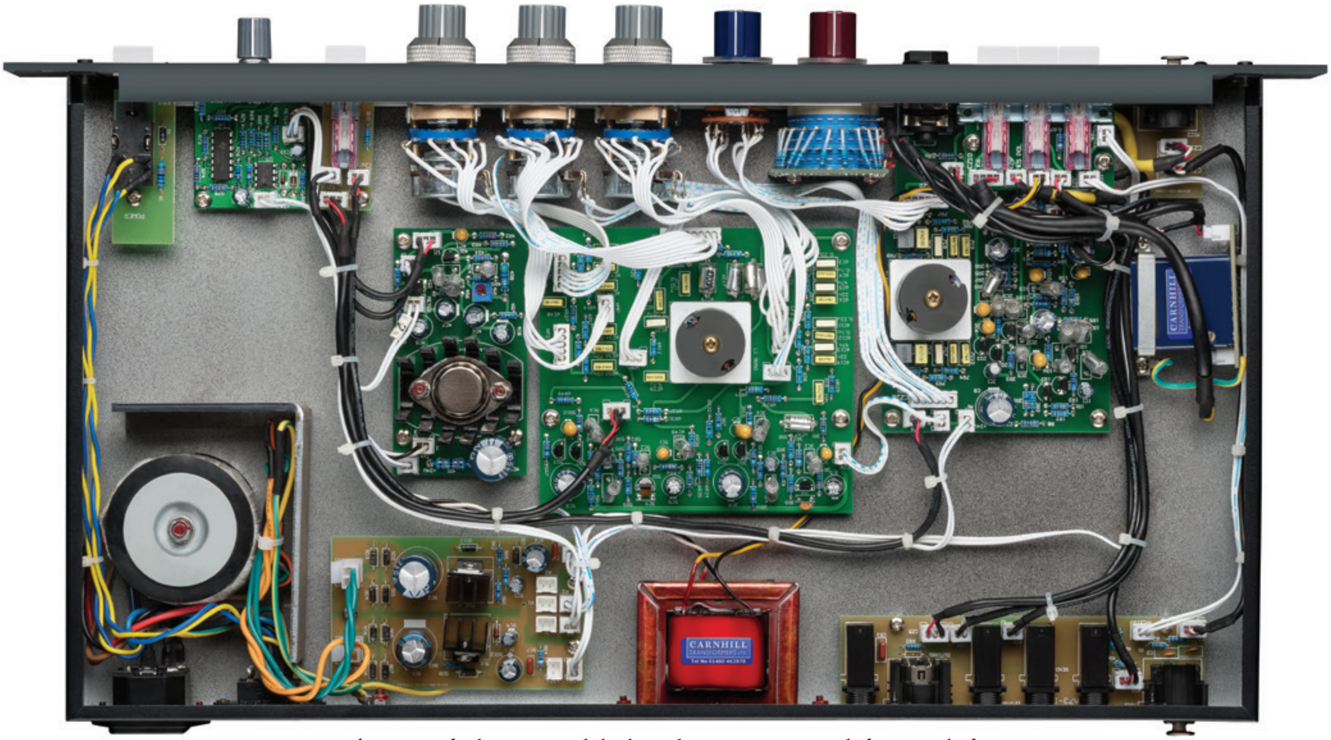
Like the potentiometers, the buttons all feel great and engage firmly and securely. In addition to the expected phantom-power and polarity switches, there's a button for Tone, which changes the impedance of the input transformer — and therefore the tonality of passive ribbons and dynamic mics. Another row of three buttons is used to engage the line and instrument inputs, and switch the insert send/return jacks in/out of the signal path.

My first impressions, then, were very positive — in fact, the only problem was that cramming this number of generously sized controls in a 1U rackmount unit is that it might be considered a little crowded.

Ins & Outs

On the front panel, you'll find an XLR mic input, a TS instrument jack for DI recording, and around the back the professional amenities continue. There's a balanced TRS line input, which passes the signal through both transformers — potentially useful for 'warming up' a live synth or an already recorded track, even when not using the EQ — as well as a duplicate of the XLR mic input. The balanced line out is available both via XLR and quarter-inch jack. Finally, the insert has separate quarter-inch TS jacks for the send and return. So the WA73-EQ caters for any physical connection commonly found in a modern studio — you shouldn't need to order special cables to integrate it into your setup. The internal power supply uses a standard IEC connection, and on the inside this is well separated and shielded from other internal components. A convenient external switch chooses between 115/230 Volt mains power, and there's a ground-lift slide switch.

Finally, removing the top panel reveals a panoply of discrete components inside. The WA73-EQ is obviously hand-assembled and partly hand-wired, with a number »



A peep inside the unit reveals high-quality components, including switched controls, two inductors and a pair of Carnhill transformers.

» of through-hole components populating a number of PCBs. It all seems to be clean, nicely spaced and well-dressed.

On Test

After wrestling the WA73-EQ out of its box, I first tried plugging in an electric guitar, and once I'd set up the gains to my satisfaction there was that low-mid fullness that I associate with Neve preamps — it sounded pleasingly big and thick. One of the things a good transformer does is smooth out transients, which has the effect of 'rounding off' the sound. It can impart a bit of slow majesty to the signal, rather like the French horn in an otherwise brassy ensemble! So, from the get-go, there was that obvious family resemblance between Warm Audio and Neve.

When used on a female vocal, I was able to find an input/output level ratio that put a Goldilocks' dollop of saturation on her voice, adding a touch of excitement to a solid performance. Later, at the Kitchen Studios in Dallas, we recorded typical rock setups through the WA73-EQ — for example, a Fender Twin miked off-centre with an SM57 — and was very happy with the results. However, you don't have to take my taste and opinion as gospel, as we captured some examples for you, and you can find these on the *SOS* website (https://sosm.ag/warm-audio-wa73-eq).

This clone definitely sounds good, then, but some will want to know just how well it emulates the original. Kitchen kindly allowed me to make some recordings through their original Neve 1073. Now, I should be clear that this was not a hugely scientific test, but rather a more practical real-world comparison, recording separate performances while changing the preamp — and we should probably also consider that any 1073 that has been in use for four decades may have had some work done over the years. Though the two devices they were certainly well within the same tonal ballpark, I could hear some small difference in the saturation effect, with slightly different intensities at some frequencies. The biggest difference I could discern between the two, though, was the way the low-frequency control (35Hz) rolled off the lows on bass guitar; the Warm unit seemed a bit punchier when it was cleaned up. So they do sound different from one another under the microscope (again, you can decide for yourself how similar or different they are by listening to the examples) but in my view it's much more important to step back from all the pointillism and appreciate the whole canvas.

Alternatives

There are plenty of Neve-style pre/EQs available at all kinds of price ranges. **Golden Age** make 1073-derived devices in both their **Project** and **Premium** ranges, while **Heritage Audio**'s 500-series version costs a little more and will require a 500-series chassis and PSU. Stepping further up the price ladder, check out the offerings from **BAE**, **Neve** and **Rupert Neve Designs**.

Conclusion

As good as some software emulations are these days, there's still something very special to me about tracking with great hardware and getting the sound right at source; it makes mixing quicker and easier — and thus much more fun! The WA73-EQ is a very capable mic preamp and EQ, which should serve as a great recording front-end, with its transformers and classic inductor EQ being great for gentle sound-shaping on the way in. It can provide a classic, beautiful tone that's a really useful contrast to the modern, more clinical sound of typical audio interface preamps. But don't discount it as a great-sounding line amp/saturator and EQ for adding character when mixing either, because it can be really useful in that role too. All in all then, I think the WA73-EQ is great value for money and should hold plenty of appeal for serious home studio owners.

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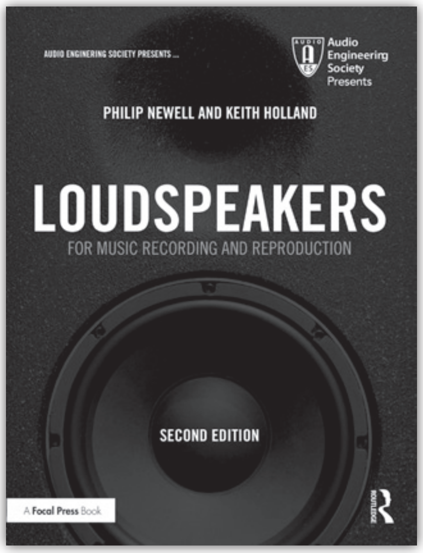
SPITFIRE AUDIO

Loudspeakers For Music Recording And Reproduction (Second Edition) Book Review



If you've read any of my monitor reviews in this magazine you'll probably have gathered that I'm fascinated by electro-acoustics and speaker design. Another thing you may have picked up from my articles is my occasional recommendation of a book on the subject, *Loudspeakers For Music Recording and Reproduction*, by Philip Newell and Keith Holland. It's a book I regularly pick up and dip into, particularly when I'm about to describe something that's technically complex or perhaps potentially contentious. It's not far off playing the role of my Bible. The first edition of *Loudspeakers* was published in 2007 but now, with the much revised and expanded second edition (Focal Press/Routledge, ISBN: 9781138554825), we have reason to give it the formal *SOS* review treatment.

Philip Newell has enjoyed a long and distinguished career in both monitor and recording studio design, and in music production and engineering — his CV includes work with Queen, the Who and Mike Oldfield, to name just three. He has published countless books and technical papers in the field of speaker and studio design and is a member of the Audio Engineering Society, the Institute of Acoustics and the Society of Motion Picture and Television Engineers. Dr Keith Holland is an Associate Professor at Southampton University's Institute of Sound and Vibration Research, where he has worked, first as an undergraduate student, since 1984. Since 1990 Keith has taught electro-acoustics at undergraduate and postgraduate level at ISVR and also on the Tonmeister course at the University of Surrey. Like Philip, Keith has published numerous research papers in the field of acoustics, and electro-acoustics.

If the names of Newell and Holland ring any bells for you it may well be because I've occasionally referred in monitor reviews to an Institute of Acoustics paper they jointly wrote in 2001. That paper, investigating both the longevity of the Yamaha NS10 and the monitoring significance of speaker time-domain characteristics, not only inspired the development of Acoustic Energy's AE22 monitor speaker, but has since become something of a totem for those, like me, who believe that how a monitor performs in the time domain (how quickly it starts and stops in response to a signal) is just as important as its frequency domain performance (how wide and flat is its frequency response).

The kind of real-world and relevant analysis that characterised the Newell and Holland NS10 paper is repeated throughout the entire *Loudspeakers* book. Although there are some slightly more academic sections that may well require a degree of re-reading and thinking about to appreciate fully, the style and content is so clear and firmly routed in the practical that it is equally valuable to read from either an academic or pragmatic perspective on the subject. Potentially difficult concepts are clearly and elegantly explained rather than hidden behind obscure mathematics, and there are few formulae to try and decipher.

If you're familiar with the first edition of *Loudspeakers* you'll perhaps be wondering what's new in the second edition. Well, apart from the design and layout receiving a polish, and the page size increasing, every section of the text has been revised and there are around 100 pages of new material, including, for example, a section on Bass Transmission Index (BTI), a fascinating new technique, based on principles similar to those of speech intelligibility analysis, for objectively quantifying the low-frequency performance of speakers.

Of course, you may not find *Loudspeakers*, the book, as rewarding and useful as I do — but if electro-acoustics happens to be a subject you wish you understood more, or if before a speaker purchase you'd like to be better informed, there is, as far as I'm aware, absolutely no better place to start. *Phil Ward*

$ Hardback $107.43. Paperback $65.07. Kindle $65.65.

W www.routledge.com

Phoenix Audio Nimbus Active DI Box

The transformerless, Class-A DI input circuit from Phoenix Audio's DRS2 two-channel mic preamp (reviewed by Hugh Robjohns in *SOS* February 2003) can pretty safely be described as very successful — not only has it been a fixture in Phoenix Audio's mic preamps ever since, but it has also spawned a range of dedicated active DI products, each with a slightly different feature set. The company's first stand-alone DI was the entirely Class-A, half-rack, two-channel Nice DI that featured a 10MΩ input impedance, the company's DSOP-2 discrete Class-A output amplifiers, and custom-wound DB694 output transformers. With only the two pairs of input and thru quarter-inch jacks and two output level controls on its front panel, the Nice DI's purist, minimalist appeal has endured down the years, and it has even found a niche as a stereo mix bus 'colour' processor. Next came a 500-series version in the form of the Pivot Tone Channel. Reviewed in *SOS* April 2017, this variant added a 10-LED ladder output level meter and a 'tilt' EQ control to balance the basic tonality of the source. The eight-channel N-8, reviewed by Hugh in *SOS* October 2018, was the next DI derivative to appear, swapping out the Pivot Tone's tilt control and LED ladder for illuminated buttons controlling a -15dB pad, polarity reverse and earth-lift switching on each channel. This potted history of Phoenix Audio's DI development brings us neatly to the latest in the line — the single-channel Nimbus DI, which combines the expected 10MΩ input impedance, Class-A transformerless input stage, DSOP-2 output amplifier and DB694 transformer with the polarity and earth-lift buttons of the N-8, along with the Pivot Tone Channel's 10-LED ladder meter.

Since Hugh's detailed dissertation on the technical aspects of a single channel of the N-8 will apply to the Nimbus DI, I'm going to refer you to that review (https://sosm.ag/phoenix-audio-n-8) if you wish to explore that area further and, instead, focus on the more practical aspects of the new unit's performance. On the physical side, the Nimbus' external, universal-voltage, 24V power supply is, as in the N-8, equipped with both an IEC mains connector and — to my personal delight — a locking connector that mates with a multi-pin connector mounted on the rear panel. Other rear-panel connectors are the transformer-balanced output XLR and an unbalanced quarter-inch jack output. The front panel carries a small power LED, the horizontally-oriented LED meter, a strangely-elongated, red-anodised, Neve-ish output level knob, the blue-illuminated 'phase' (polarity) button, the green-lit earth-lift button and two quarter-inch jacks for the unbalanced input and its thru output.

One minor anomaly I immediately noticed is that the first green LED in the ladder meter is permanently lit, and it is identical in colour to the green power LED. Personally, I find that confusing for a couple of reasons: I want to see a different colour LED for power indication; and I don't want to see a meter segment lit when there's no signal present, unless that is the only front-panel indication that the unit is receiving power.

As Hugh found in his tests and I did in mine — both with the Nimbus and the Ascent Two EQ I reviewed recently — the performance of the Phoenix Audio DI circuit is absolutely excellent. Phoenix Audio's approach of using the output amplifier, rather than the output transformer, to provide saturation and harmonic colour gives their DI devices the ability to go from pristine clarity to saturation, making them more than ideal partners not only for electric instruments, but also, with that 10MΩ input impedance, for piezo-equipped acoustic instruments. I'd also happily add a couple

of Phoenix Audio Nimbus active DI boxes to my collection for those occasions when an instrument or mix needs that bit more character to make it stand out from the rest. The Nimbus isn't exactly cheap for a DI, but when you need it, it will be well worth the price. *Bob Thomas*

$ $399.99.

W http://phoenixaudio.net

Nobels ODR-Mini Overdrive Pedal

Nobels may not be the first name that springs to mind when you think of effects pedals, but their ODR-1 overdrive is by all accounts extremely popular with Nashville's musicians. Though it's designed in Germany, the pedal is built in China — and while it isn't expensive, it delivers a distinctly boutique-like tonality. The ODR-Mini is essentially a slightly tweaked version of that pedal, with true bypass and a tightened low end, but squeezed into a mini-format case. This means there's no battery-power option, of course, bit it is fitted with a standard centre-negative barrel connector at the top end of the case, and can be run at any DC voltage between 9-18 Volts (the higher the voltage, the greater the headroom).

At first glance, the control layout looks pretty standard, with Volume, Drive and Spectrum (Tone), but the way Spectrum affects the tone is a little different from the norm. Rather than act as a simple top-cut filter, this adds a low/low-mid boost when positioned clockwise of centre, and what sounds like high- and low-cut filters when left of centre, giving more exposure to the mids. Its detented centre position is about as close as the pedal comes to a transparent sound (with the drive set low), though there's always a little added warmth in the low mids, combined with some dialling-back of the extreme highs. Personally I'd probably have preferred to get some of those highs back at lower drive levels, but that's a personal-taste thing, and wouldn't be a deal breaker for me even then.

The drive range of this pedal is impressively wide, going from virtually clean to a classic rock-style overdrive. Using the Spectrum control in the second half of its clockwise travel delivers the most overdrive, as the EQ boost comes pre the drive circuit. However, it doesn't strip away all your dynamics, as some super-saturated pedals do, and even at higher gain settings both chords and single-note lines retain definition. The circuit responds extremely well to picking dynamics, and if you apply some mid boost by turning the Spectrum control below its half-way position, the effect is not unlike a smooth version of a Tube Screamer — but without the loss of low end. Set the Spectrum control clockwise of centre to add weight to the low mids and restore more of the upper mids, and that can be really useful for fattening single-coil pickups.

While I have to say that this isn't one of those pedals that wows you when you first plug-in, after using it for a while its versatility really does start to shine through. Furthermore, as the drive can be dialled back to almost clean, the ODR-Mini works well in conjunction with other drive pedals, or amplifiers that are set in that responsive region between clean and dirty. I like the way it retains note definition across the drive range and I can see how the Spectrum control could be particularly handy in a studio session, when you need to switch between guitars with different styles of pickup. Because the pedal smooths out the highs, it may be best suited to single-coil pickup instruments rather than already fat-sounding guitars with humbuckers, but if you like your drive on the more subtle side, this is definitely one to check out. *Paul White*

$ $75.

W https://nobels.de

Rode RodeCaster Pro

Podcast Studio Console & Audio Interface

The RodeCaster Pro is a one-stop shop for creating high-quality podcasts.

PAUL WHITE

Make no mistake, podcasting is getting to be a very serious business. And for those who want to create radio show-style productions that include music, cues, jingles or sound effects as well as up to four miked voices, the RodeCaster Pro looks to be a very user-friendly and technically capable option. It incorporates on-board recording and playback facilities via a Micro SD card, and there are four microphone inputs with the provision of phantom power, plus further inputs for USB audio, a hardware music player or phone and Bluetooth audio. The RodeCaster Pro can also function as a two-channel, bi-directional audio interface.

Pod One Out

Each of the RodeCaster Pro's seven input sources is controlled by its own fader, with a separate fader to control the overall level of its eight playback pads. These allow for triggering sound effects, cues, music and so on, and the pads can be 'loaded' via direct recording or via a simple drag-and-drop routine when connected to your computer.

There are four separate headphone outputs on the rear panel of the RodeCaster Pro console, each with its own rotary level control, plus a pair of balanced TRS jack speaker feeds suitable for direct connection to powered monitors. Only the first headphone output follows the channel solo operation, so the operator can check on sources without interfering with what the other three listeners hear. During recording, a timer shows how long your recording has been running for. A large red Record button starts and stops recording and a new file is created each time recording is started. If a phone is connected using a TRRS cable or Bluetooth, telephone interviews can be included in the mix — and an Auto Mix Minus facility eliminates echoes in the monitor feed when using a phone.

Sounds for the playback pads are stored on an internal 512MB flash memory, which has more than enough capacity for intro and outro music plus short cues or jingles. Software for loading these pads by copying WAV files from a connected computer comes as a free download, though you can also record material directly into the pads in stand-alone mode. The RodeCaster Pro software simply shows eight pads onto which files are dragged as required, and their coloured backlights can be set to different colours if you like.

The RodeCaster Pro console is equipped with Class-A mic preamps, and further voice enhancement can be added via the Aphex-on-board circuitry that provides both an Exciter (for adding harmonics to brighten the sound) and Aphex's low-end Big Bottom enhancement. Aphex invented the Exciter so this is the real deal. The RodeCaster Pro also incorporates compression, limiting, de-essing and noise-gating as part of various types of voice-processing characters from which you can choose, so there are all the tools you need to create a radio-style voiceover. An Advanced mode allows these processors to be turned on or off individually, but does not cater for parameter adjustment — the design ethos is to keep things simple for the operator.

Stand & Deliver

Used without a computer, the RodeCaster Pro works in stand-alone mode to allow real-time radio broadcast-style recording

Rode RodeCaster Pro
$599

PROS
- Spacious and uncluttered control surface.
- Four mic inputs and four headphone outputs.
- Pads for triggering cues.
- Bluetooth and TRRS jack phone connections.
- Dead simple to use.
- Clean and quite in operation.
- Firmware update will add multitrack recording capability.

CONS
- Currently the Micro SD card has to be removed and placed in a card reader to move files to a computer.

SUMMARY
For podcasters who don't like to get too technical but who demand high-quality results, the RodeCaster Pro does an excellent job. It isn't the solution for those looking for multi-channel USB operation or

to a Micro SD memory card (the internal memory is just for storing the pad sounds), but you can also hook it up via its rear-panel USB port to a computer, where it appears as a class-compliant stereo audio interface that works with your DAW — a good option if you need to edit what you've recorded (there are nearly always 'ums', 'ers' and fluffed lines to edit out... at least, there are when I'm doing it!). Using the USB output allows a recording to be made on the computer at the same time as to the Micro SD card. Note that if you want to erase recordings or transfer them to a computer for further editing, the Micro SD card has to be removed and then hooked up to the computer via a card reader, so capturing your recording to a suitable DAW on your computer as you go along will save time. I'm hoping that a future firmware update will make file transfer possible without having to remove the card, as having to do this adds an inconvenient step to the otherwise silky-smooth workflow.

The RodeCaster Pro weighs a little under 2kg and measures 350 x 275 x 82mm.

In Use

At the time of doing this review I hadn't managed to find a manual either in the box or online — and it turns out that there isn't one. That's because any necessary instruction comes from online video tutorials, but even without them you can explore pretty much everything in a purely intuitive way as operation has been kept as simple as possible. The touchscreen 'cogwheel' icon takes you to settings for recording and for playing back podcasts; it's here that you choose the best processing preset to suit your voice, tell the RodeCaster what type of mic you have connected (specific Rode models and generic dynamic or capacitor types), pair Bluetooth devices and set your recording levels. That's pretty much all you need to do — there's nothing geeky to get in the way.

If you take any notice of what is said on forums, the most common gripe is the current lack of a multi-channel recording facility. If you assemble your podcasts in the style of a live radio programme and need a stereo file as the finished product, the RodeCaster Pro does the job exceedingly well. Having said that, news reached me just as I was finishing this review that a firmware upgrade will add multitrack recording when in Advanced mode. I'm told this will allow up to 14 tracks to be recorded including separate tracks for the four mic inputs and stereo tracks for the USB, 3.5mm TRRS, Bluetooth and sound pad channels. Hopefully the update will be available by the time you read this.

The other recurring forum whinge is the lack of tweakability in the effects/processing department, but again the RodeCaster Pro has been designed to help non-technical people create a polished end product. So, being able to turn things on or off in Advanced mode or having the decisions made for you when you select from one of three voice types and one of three processing intensities is probably all that's needed. Certainly I was very happy with the voice timbre I achieved.

In use, those long faders on the front panel control the playback level very smoothly, making it easy to balance multiple voices against any pad-triggered music, phone-ins or sound effects. All eight pads come pre-loaded with some fun sounds such as applause and laughter, but it is easy to record your own. As a test I used the software to drag and drop some of the *SOS* podcast cues on to the pads, then I put together a short podcast, which was incredibly easy. Music and cues stay in stereo while the voice is panned centre.

Music or interviews can also be streamed in from a phone, either with the TRRS jack or via Bluetooth (you could bring in two devices at the same time by employing both), so in all essential respects what you have in front of you is a mini radio production system complete with phone-in capability. The sound quality is excellent and being able to trigger music and effects from the pads makes creating a podcast extremely fast and intuitive.

As well as podcasting, the RodeCaster Pro is an ideal platform for recording talking books that require sound effects, or radio-style plays with up to four performers. I can't stress too highly how comfortable the RodeCaster Pro is to work with, and the only things that I hope Rode will add in a future update is the ability to delete recordings directly from the front panel, and to make it possible to transfer recordings to a computer without first having to remove the Micro SD card and then connect it to the computer using a card reader. In all, though, a very enjoyable user experience.

$ $599
W www.rode.com

Mackie DRM speakers in various configurations, including the DRM12A line-array element (shown left on a pole with the optional DRM18S sub, and tripod-mounted centre-right) and the DRM212 (shown right on a pole with a sub, and tripod-mounted and floor-standing centre left).

Mackie DRM: new live speaker range

Mackie's new flagship DRM PA speaker range, consisting of three stand-alone active or passive speakers, one modular line-array element and a subwoofer, with built-in DSP control panels on the active models to aid setup, is aimed at everyone from gigging musicians right up to venues and touring outfits.

The larger members of the range, with their optional flying hardware and features designed for usage in installations and large-scale stage productions, probably go beyond the requirements of anyone reading this. However, the smallest in the range, the 1600W, 12-inch DRM212, can be pole- or floor-mounted for use in a band PA or as an on-stage monitor and, in its Class-D active version, is a neat all-in-one stage speaker capable of producing peak levels of up to 134dB SPL. The 2000W DRM12A line-array element also contains easy-to-use setup options in its DSP control panel for use in one- and -two element speaker configurations, which could play a useful role if pole-mounted as part of a small mobile band PA.

All the speakers in the DRM range are constructed to Mackie's usual rugged 'road-ready' standards, feature built-in power factor correction and universal PSUs (100-240V), and offer easy access to their DSP control options via full-colour LCD control panels driven by a simple, one-knob 'turn to adjust and push to confirm' control interface. Priced at $999 for each active DRM212, $2099.99 for each DRM12A element, and $1499.99 per 18-inch 2000W DRM18S sub, the range will be available later in Spring.

www.mackie.com

Earthworks debut stainless-steel live vocal mic

Although traditionally associated with measurement and instrument microphones, Earthworks also like to shake things up once in a while by producing studio vocal mics, and they recently surprised us at NAMM by producing a *live* vocal condenser mic. What's more, the new SR314 is made of shiny, stainless steel — so it certainly stands out. In true Earthworks fashion, the mic offers a very consistent polar pattern (tight cardioid in this case), with excellent rejection at the rear of the mic when we tried it out on the booth at NAMM, which bodes well for resistance to feedback.

Earthworks claim an extended frequency response of 20Hz-30kHz and that the SR314 can handle SPLs up to 145dB, which will also stand it in good stead live. Shipping later in Spring with a supplied MC4 mic clip and a padded protective bag, the SR314 will retail for $874 in the US. A wireless version is planned for later in the year.

www.earthworksaudio.com

Akai MPK 88-note controller hits the road

The MPK Road 88 is a new full-size hammer-action keyboard controller from Akai aimed at stage use. Designed to integrate with existing sound modules or virtual-instrument sound sources, the 88-note keyboard has both five-pin MIDI in and out ports, plus an integrated class-compliant four-output USB audio interface with two aux and two main outputs on quarter-inch jacks. The latter allows users to trigger laptop-based plug-in instruments from the keyboard over MIDI live and route the resulting audio back from the computer to the front-of-house mixer and monitoring. A fair amount of thought has obviously gone into what keyboard players need most during live performance: dedicated top-panel volume, transpose, keyboard split and preset selection controls are provided for instant access mid-gig, alongside the usual pitch and mod wheels, and there are no fewer than three pedal inputs to add further foot-controlled expression or MIDI control.

The MPK Road 88 weighs 30kg, and ships with a rugged travel case with a handle for extra portability. The whole package retails for $899.

www.akaipro.com

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PAUL WHITE

Xvive U3

Wireless System

If you're looking to add wireless functionality to your favourite mic, Xvive's U3 could be just what you need.

If you like the idea of a radio mic but don't want the hassle of wiring up base stations, plugging in PSUs and screwing on antennae, the Xvive U3 might appeal, as it is well suited to smaller gigs and is extremely straightforward in operation. Both the transmitter and receiver are built into what are effectively oversized XLR connectors, so you simply plug the transmitter directly into the back of your mic and the receiver directly into the XLR input of your mixer. It is even safe to plug it into a phantom-powered mixer input, though you can't charge or power it from phantom power. A switch on the transmitter can be used to select line-level input sensitivity for use with electronic instruments, and Xvive also make a similar system with a swivel jack at the transmitter end and a suitably high input impedance, specifically for connecting electric guitars and basses. Power for both transmitter and receiver comes from internal, rechargeable lithium-ion batteries, giving around five hours of continuous run time from a full charge. You will need a USB power source to charge the units; none is included, but a Y cable is provided so you can charge both pieces at once from a single outlet.

Circular displays surrounding a step button on each unit allow the user to select from one of six channels (each of which automatically selects the best of three frequencies), and because the U3 works on the 2.4GHz band shared by Wi-Fi, no licence is needed to operate it anywhere in the world. All six channels can be used at the same time if you have multiple systems on the same stage, and the range is as much as 90 feet if there are no obstructions. Even with minor obstructions, there should be no problem reaching the stagebox or mixing desk in smaller venues, but you may run out of range if you have a gig at the O2! The manual advises to set up at least three metres from Wi-Fi routers to reduce the risk of interference.

All digital radio links exhibit some latency, but the 5ms delay typical of this system should not be an issue unless you are using in-ear monitoring, in which case those with sensitive hearing may be put off by the timbral changes that can occur due to the internal bone-conduction not being exactly aligned with what comes back via

Xvive U3 $199

PROS

- Extremely simple to use.
- Excellent sound quality.
- Rechargeable batteries reduce running costs.
- No licence needed.

CONS

- If the battery goes flat at an inconvenient time you can't change it.

SUMMARY

This is one of the simplest radio mic systems I've come across and is almost as quick to set up as plugging in a mic cable. Even if you forget to charge it, then as long as you put it on charge the moment you arrive at a venue to set up, you should be able to get through the evening.

the in-ears. This is a factor with all digital wireless systems, though, not just this one. Certainly the technical spec of the U3 system is impressive: its 48kHz, 24-bit converter resolution yields adequately low distortion, a 20Hz to 20kHz frequency response, and a dynamic range of 110dB, which is better than you'd expect from even the better UHF systems.

Joie De Xvive

The cases are made from cast metal at the connector end, with a plastic shell at the other. Both the transmitter and receiver have a small power slide-switch and a shallow recess for the channel selection button, while on the transmitter this switch is joined by another for choosing line or mic mode (all the switches are designed to prevent accidental operation). The units feel substantial and rugged without being excessively heavy, and the transmitter has a latching XLR to keep it in place. The active channel is illuminated in blue when the device is powered up, so this doubles as a power-on indicator, and when the two devices are in communication, a green LED on the receiver also illuminates. Two red LEDs on the transmitter flash briefly on power up and then go off. If they come back on, the battery is below 30 percent charged, and if they flicker it is time for a recharge as that means there's less than 10 percent power left. Clearly there's no phantom power, so capacitor mics can't be used unless they are battery-powered electret types.

The batteries are not user accessible, as far as I can tell, so if you do run out of charge during a gig you can't fit new ones. There's also no explanation as to how to replace the batteries at the end of their life, so it may be a case of returning the system to a service centre. A full charge takes around 2.5 hours, but a one-hour charge will get you through a two-hour gig. You can also run the receiver from a USB PSU in an emergency.

The sound quality was all I'd expect from a digital radio mic system: clean and quiet to the point that it stands comparison with a hard-wired mic. Setup is lightning-fast as there's no base station to set up and no antennae to screw into place — both units have internal antennae. No doubt a bigger system with twin-diversity receivers would have more range, but for the target market of playing pubs and clubs, range isn't likely to be an issue. If you do go out of of range, the sound breaks up but doesn't dissolve into noise in the way analogue systems can. For gigging performers who don't like complications and who want to add a cable-free mic or two to their show, this is a very practical and affordable system.

Alternatives

Line 6's affordable and compact wireless mic and guitar systems are worth checking out, as are **Rode**'s and **Audio-Technica**'s wireless options, alongside systems from the major EU mic companies. However, the **Nux B3** system is the only alternative I've found that has the same format as the XVive U3.

$ $199
W www.xviveaudio.com

Korg Konnect

Mini PA System

Korg's new mini PA is as versatile as it is compact and portable.

PAUL WHITE

The Korg Konnect is a portable stereo sound system, styled as a large 'ghetto blaster' and rated at 180 Watts peak or 90 Watts continuous power. It manages this by incorporating a central 6.5-inch woofer in a reflex enclosure, with one-inch tweeters placed on either side. A pole-mount socket is located in the base of the plastic casework, but the speaker is just as happy sitting on a desk or table.

Designed to meet a number of needs, from small 'coffee-shop' PA and dance groups to exhibitions and presentations, the Konnect weighs just 5.13kg and is only 452mm wide. Power comes from an external power adaptor that connects using a simple push-in barrel connector, but I feel the designers have missed a trick by not including a place to stow the PSU in the case itself (a carrying bag is available as an optional extra). I'd also expected the option of battery operation, but it seems that isn't catered for.

The Konnect features a built-in mixer with a choice of EQ presets for specific sound sources and a selection of effect types including reverb. All the processing is carried out at 32-bit resolution by a SHARC DSP section. Although it doesn't have the kind of physical EQ controls you'd find on a stand-alone mixer, there is a free control app available which allows for more in-depth editing, including detailed adjustment of the internal five-band EQ section. The app, which requires at least iOS 8.1 or Android 5.0, also permits wireless control from a smartphone or tablet.

There are five preset voicing types, called Music, Male Vocal, Female Vocal, Electric Guitar and Acoustic Guitar, but the app adds a further 12 options that include Bass Guitar and Keyboard. Using the app also enables the reverb type and settings to be adjusted, and gives you access to compressor, chorus and delay settings. For karaoke sessions, there's a 'centre cancel' function for attenuating vocals, the effectiveness of which depends on how the original track was mixed. An anti-feedback section is also included. User settings can be saved as Scenes for later recall.

Making Konnections

The mixer, which has four channels in all, can accept mic- or line-level sources (up to a maximum of two microphones), and also allows for music streaming to channels 3-4 over Bluetooth or connection via a stereo mini-jack. These two channels may also be used as mono line inputs, accessed via separate quarter-inch jacks. All the mixer connections and physical controls are on the rear panel, as are the DC input connector and power switch. To the right of these are the Master Volume control and the reverb Type/Level knob. A built-in limiter helps stop you overcooking things, and a red LED comes on when this is doing its thing. A Bluetooth LED flashes when the channel 3-4 select button is used to initiate proceedings, then lights steadily once pairing is achieved — which turned out to be a fast and painless process. Feedback suppression is just on or off, and affects only channels 1 and 2, but seems to work pretty well, relying on auto-adjusting narrow notch filters that pick up on feedback squeal.

When the app isn't being used, three channel select buttons determine which channel will be controlled by the Volume and Voicing knobs. Each channel also has a clip LED to show if the input levels are being overdone. The centre cancel feature affects inputs 3-4 only, and is simply on or off. There's no phantom power for capacitor microphones, and no dedicated

Korg Konnect $399

PROS
- Easy to use.
- Built-in effects and anti-feedback processing.
- Up to four inputs.

CONS
- Limited mic gain.
- External PSU.

SUMMARY
A practical catch-all mini sound system in a single box.

Alternatives

The most direct alternative is probably the **Mackie FreePlay**.

Though the Konnect does have some physical controls around the back, more options are available via the accompanying app.

instrument input option, so a passive guitar would need to feed into the line input via a preamp or suitable pedal.

Powering Up

After you've installed the free Konnect app, control from an iPad or phone is very straightforward, even in the so-called Advanced mode, and there's an in-built soft manual if you need it. In Advanced mode, there are EQ cut/boost controls for the five fixed frequency ranges. The rotary controls operate in a different way from most audio plug-ins that we're familiar with, though; after you select a control by touching it, a slider bar appears and that's what is used to make the adjustment. Reverb is always available, in one of four flavours covering small to large spaces, and the effects section of each mic/line channel can be set to add additional compression, delay or chorus processing.

I found there was plenty of mic gain for sung vocals, but if somebody is doing a spoken-word presentation with their mic a few inches away from the lips, you might find that maxing out the channel and master gains is not quite enough unless they have a fairly sensitive mic. Tonally, the system sounds pretty clean until you push it too hard, after which the plastic cabinet imparts a somewhat boxy tonality. Music playback is well balanced with a solid enough bass end, but again, if you push the level too much, it starts to sound a bit stressed.

Overall the Korg Konnect does as claimed, producing a clean sound for both voice and music at low-ish to moderate sound levels. For a gigging band it might come in useful as a mini monitor or as a stand-alone PA for front-room or small coffee-bar gigs, and it might also make a useful extra speaker for those pub gigs in L-shaped bars, where having a small speaker to push some sound around the corner can help. And when it's not doing any of that, you can use it as your home sound playback system or TV soundbar!

$ $399
T Korg USA +1 631 390 6500
W www.korg.com

MIX RESCUE

OUR EXPERTS TRANSFORM YOUR TRACKS

We load up the magic busses to take a psychedelic rock mix on a fresh trip.

SAM INGLIS

We're often told that technical quality is less important than musical quality. As long as you have a good song and a good performance, it's said, everything else is secondary. That may be true, but it's also true that a good performance of a good song can be spoiled by technical problems, and so it was with this month's Mix Rescue candidate. The Vietnam Flashbacks describe themselves as a "psychedelic jam band", and unlike some acts of that ilk, they have genuinely strong material to underpin their sonic experimentation. What they didn't have was mixes that did justice to this material.

Striking A Balance

The process that we now call mixing used to be known as 'balancing', and I still think this is a better description of the job that needs to be done. There are two main senses in which a good mix needs to be 'balanced'. First, the relative levels of each sound source need to make sense, with nothing being either too loud or too quiet; and, second, the overall tonality of the mix has to be effective. There must be enough action at the bottom end for the track to sound full, yet not so much that it comes across as boomy or muddy. Likewise, the mid-range needs to be present without being boxy or tinny, and similar considerations apply to the high-frequency region.

The two kinds of balance are interconnected. If you find yourself in a situation where nothing seems to sit at the right level, it's often because the overall tonal character of the mix has gone awry; and a bass-heavy or top-heavy frequency balance tends to make particular instruments poke out unpredictably on different playback systems. The Vietnam Flashbacks' own mix of 'Cherry Cola' was a case in point. The overall frequency balance was heavily 'scooped', with too much bass and a strong emphasis in the upper treble region. Consequently, the instrumental balance was also wonky, with the bass guitar dominating the mix, the hi-hat more prominent than the snare drum, and so on.

Two things are key to avoiding this sort of problem. The first is having a reliable monitoring system, and checking your mixes on headphones and other speakers where possible. The second is referencing. To keep your mix on the straight and narrow, identify a couple of commercial tracks in a similar style to yours, and return to them every so often during the mix process — but be sure to match the levels, because our perception of tonality is heavily dependent on loudness. In this case the band hadn't mentioned specific references; as the track seemed to me to recall the finer moments of the Rain Parade, I used one of their songs as my main comparison.

Hearing Things

Frequency balance issues quite often originate in tracking. If you're listening on an unreliable system, or you're working in a single room and don't have an isolated playback system, it's all too easy to position mics and adjust amps in such a way that your sources are captured with too much low end, or an overly aggressive mid-range,

Rescued This Month

The Vietnam Flashbacks are a psychedelic band from Leicester who record live in their rehearsal room-cum-studio. They are influenced by artists like Love, Hendrix and the 13th Floor Elevators as well as '70s music like Hawkwind, mixed in with indie and modernish bands like the Black Angels. The full line up is: Ben Moore (keyboards), Dave Kidd (bass and vocals), Lance Bennett (drums and vocals), Pete Illson (guitar and vocals) and Riki Maru (vocals). "Psychedelia is our trip," they say, "and if you like psychedelic rock then we are what you're looking for!"

excessive sibilance and so on. I find this is particularly common when people position mics while listening very loud on headphones, and if you're forced to work like this, it's definitely worth making test recordings and playing them back quietly before committing to a sound.

The Vietnam Flashbacks' multitracks were a mixed bag from this point of view. Drums are fundamental to rock music of every stripe, and the basic drum tracks on 'Cherry Cola' were both well played and decently recorded. I had the luxury of two kick-drum mics and no fewer than three snare mics to work with; often this just means more flavours of not-quite-right, but in this case, they provided useful alternative characters. The only fly in the ointment on the drum front was that the overheads sounded a little too distant, with the snare noticeably off-centre.

Moving on, the bass guitar sound was a little on the woolly side, but not disastrously so. This was lucky, because there was enough drum spill onto its mic to make wholesale resculpting a tad dicey. The multitrack also contained four keyboard parts, essentially playing variations on the same thing using different sounds; these required some work, as I'll explain shortly, but the most problematic raw tracks were the electric guitars and the vocals.

Riki Maru's lead vocal was the feature that really sold me on 'Cherry Cola' when I first heard it: a characterful and very English performance that was perfect for the song. What the multitracks revealed, however, was that this vocal hadn't been terribly well captured. It had evidently been recorded while monitoring on loudspeakers rather than on headphones, making spill an issue, but a bigger problem was an inappropriate mic choice that sounded both boomy and sibilant.

The four guitar tracks presented slightly different problems. One was a conventional, slightly dirty electric part; another, in proper psychedelic spirit, had been recorded backwards, and sounded quite muddy, perhaps an indication of the kind of monitoring-while-getting-sounds issues described earlier. The last two guitar parts posed more of a headache. Both of these ran through almost the entire song, and clearly were important to the arrangement, but both had been played through heavy wah and fuzz. The fuzz obscured note definition and added lots of fizz, making them into a fairly indiscriminate and harsh-sounding wall of sound, while the wah ensured that they never presented

Sound Radix's Drum Leveler plug-in was used to level out the kick and snare drum, and to reduce the amount of spill on those tracks. Soundtoys' Decapitator helped to beef up the kick, the key parameter being Thump, which adds an emphasis at the turnover point of the low-cut filter.

a consistent spectral signature that could be targeted with EQ.

Kitted Out

I started my own mix with the drums, partly out of habit and partly because it seemed particularly important in this case to establish a solid foundation that would put the other elements in context.

Home-studio drum recordings can often be improved by paying attention to polarity, and sometimes by using microscopic time adjustments to make the elements of the kit work as well as possible with the overheads. When I'm recording drum kits myself, I usually pack a tape measure to ensure that both overheads are equidistant from the snare drum. That evidently hadn't been done here, as close inspection of the waveform showed it arriving slightly earlier in one mic than in the other. Compensating for this difference using Eventide's Precision Time Align plug-in brought about a noticeable improvement, especially when the kit was auditioned in mono. I also adjusted for the off-centre snare by panning the two overhead mics at different settings.

Other drum processing was fairly conventional. I find Sound Radix's ingenious Drum Leveler plug-in much more natural and controllable than a conventional compressor when it comes to evening out the dynamics in a drum kit, and it offers controllable leakage suppression into the bargain, so I brought this into play on both the kick drum and snare drum. A small drum room »

The snare in the song's middle eight was treated to a pitch-diving delay effect courtesy of Soundtoys' Crystallizer, which in turn fed a plate reverb plug-in.

impulse response from EastWest's Spaces II added weight and depth to the snare, and although three tom close mics had been recorded, there was only a single tom hit in the entire song, so I cut that out and muted the rest. Finally, the raw kick sound was a bit lacking in power and substance, so I resorted to a couple of my favourite tricks to beef it up. First I applied some saturation using SoundToys' Decapitator: by enabling its Thump parameter and carefully adjusting the setting of the Low Cut dial, it's possible to add exactly the right amount of heft to the bottom end, in this case at 50Hz. The resulting signal was fed through FabFilter's Pro-MB multi-band processor, with a mid-range band set to compress and a treble band set as an upward expander to enhance the attack.

Top Down

Getting the tonality of a mix right can be especially challenging if you proceed by attacking all the individual elements in turn, because it's not obvious at the start of a mix what any given element should contribute to the overall frequency balance. The tonality of the mix as a whole is the product of all its elements and, as such, is most easily shaped using EQ on the master bus. So, having got the drum sound close to where I wanted it, I put up a very rough balance of all the other tracks and started experimenting with master bus EQ.

This isn't the time for surgical notches or dipping out specific frequencies: rather, it's about using shelving or tilt filters to put the tonality in the same ballpark as your references. Usually, this means adjusting the overall frequency balance to favour the treble and upper mid-range, and that was exactly what I did here, adding a very broad shelf to lift everything above 1.5kHz by several dB. These settings evolved as I worked on the mix — my final effort had two EQ plug-ins and a multi-band compressor on the master bus — but the important thing is to be working within the target zone from the earliest possible moment.

Low Points

Next up was the bass guitar. Sonically, this wasn't a million miles from where it needed to be, but it took a surprising amount of mucking about to bridge what should have been quite a small gap between the recorded sound and the sound I wanted in the mix. There was noticeable saturation in the raw bass sound, which should in theory have added welcome brightness to it, but in practice it had a muffled and slightly vague tone that needed firming up. My eventual plug-in chain involved EQ, full-band and multi-band compression, but the most important element was the SansAmp PSA-1 distortion plug-in that comes with Pro Tools. This might be old and free, but it's still incredibly useful on bass instruments, thanks to the way its effect can be focused in different frequency ranges. In this case, I used a fairly high setting of the Punch control to add some mid-range emphasis, and turned the Buzz parameter right down to eliminate lower mid-range bloat.

Meanwhile, applying an EQ boost on the master bus had revealed noticeable upper-mid harshness in the vocal track, on top of the other issues. The boominess at the low end was easily tackled using EQ, a boost at 2kHz gave it some much-needed bite, and a fairly aggressive de-esser setting got the worst of the sibilance under control — but the heavy lifting was done by Oeksound's Spiff plug-in. This can be rather a mysterious processor, and it takes some practice to get useful results from it, but one of the things at which it excels is smoothing out

An oldie but a goodie: the SansAmp PSA-1 plug-in helped to add definition to a slightly woolly bass sound.

Audio Examples

You can find the accompanying audio examples on the *SOS* website at https://sosm.ag/mix-rescue-0419-media.

» a vocal that contains spitty or aggressive consonants. There wasn't much I could do about the spill, but since it proved necessary to automate the vocal level in any case, I dropped it right down in the gaps between phrases.

The four keyboard tracks presented a different sort of problem. They had presumably been recorded by direct injection, and there was nothing wrong with them from a technical point of view; the challenge lay in fitting them into the mix. Most of the classic 'rock band' instruments are either bandwidth-limited by nature (electric and bass guitars), or percussive, leaving gaps for other instruments to fit into (drums), so they interlock fairly naturally to form a complete picture. Bringing four keyboard parts into this picture complicates matters — especially when each of them seems to fill the entire frequency spectrum on its own!

I used a combination of tactics to try to shoehorn them in. One was simply to drop some of them out for different sections of the song, generally thinning out the verses and creating a denser texture to lift the choruses. Another was to limit their respective frequency responses using a combination of EQ, cabinet simulation and other fidelity-destroying plug-ins.

My Cup Of Fizz Runneth Over

That left the four guitar tracks, all of which also played pretty much throughout the entire song. One of them was a pretty straightforward rhythm guitar part, transitioning into a short solo right at the end. I used automation to drop it down in level in the verses, and to bring it from the left into the centre for the solo, but it didn't need much additional processing. That left the backwards noodling and those challenging twin fuzz-wah parts.

LMDSP's Superchord plug-in is a fertile source of weird and wonderful effects, as applied here to the guitars.

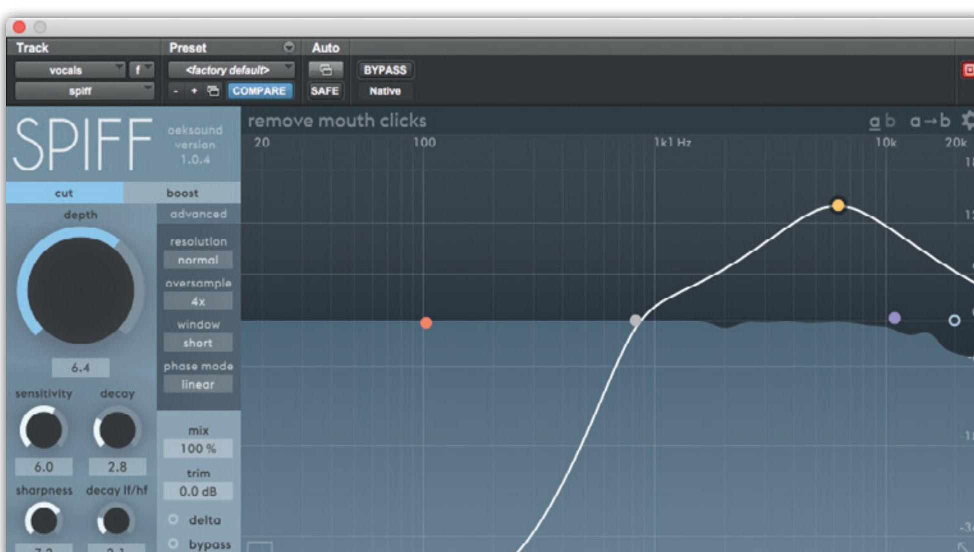
Although the band's own mix had its share of technical issues, as described at the start of this article, it also had good qualities. Above all, it was undeniably psychedelic, with a trippy, head-spinning vibe that was spot-on for their chosen musical style. Thus far in my own attempt, I'd concentrated mainly on the meat-and-potatoes stuff that would deliver a technically sound mix. With that now in place, it was time to add the audio hallucinogens back in, and since I couldn't find conventional ways to make the additional guitars sit in the mix, I decided to treat them less as parts in their own right, and more as sources of spacey effects.

This is generally something I prefer to leave to near the end of a mix, even if those trippy effects are going to be quite prominent, because otherwise it's too easy to get caught up in generating ever more creative noises and neglect the fundamentals. Often it's also the most fun part of mixing, so saving it until last is a way of rewarding yourself for all the hard work that went into the basic building blocks of the mix. And given that this track was explicitly billed as being psychedelic, I felt I had *carte blanche* to lay on the crazy stuff pretty thickly.

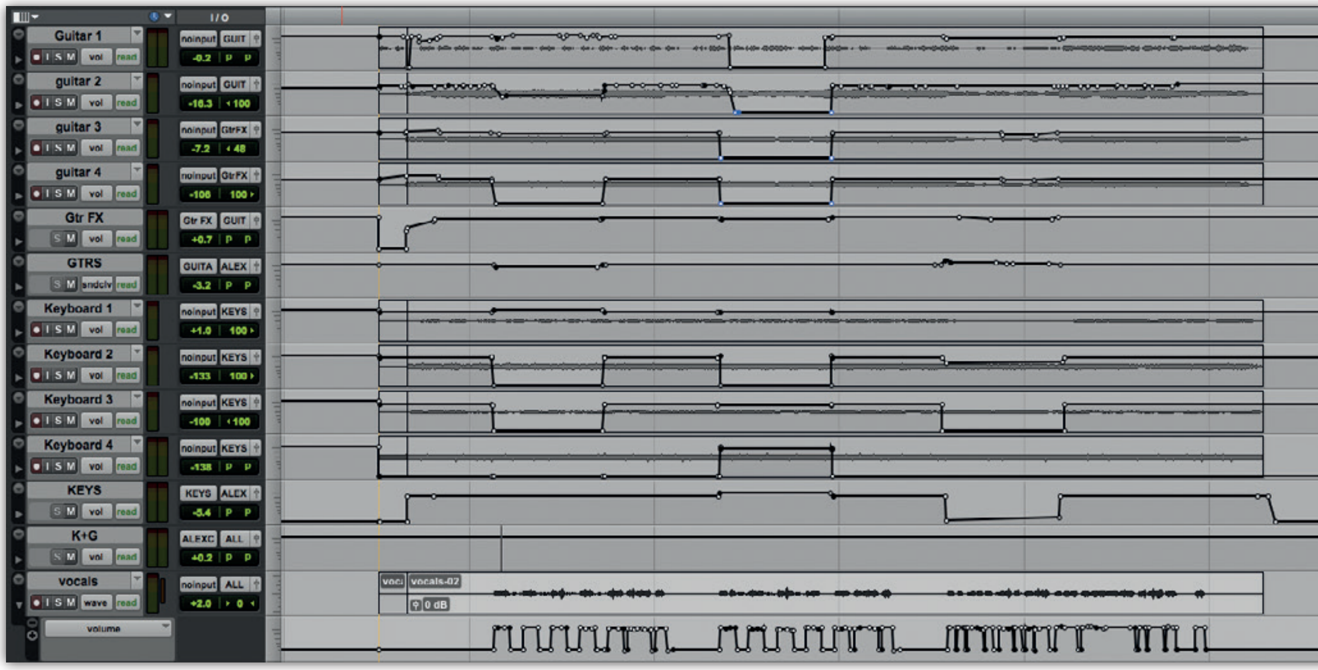
Mind Bending

After I'd removed a lot of low-mid mud with EQ, the tone of the backwards guitar part was basically OK, so I didn't feel the need to try to turn it into something it wasn't. However, it was obviously crying out for some psychedelic delay effects, and on top of that, I also used Soundtoys' PanMan auto-panner to make it lurch unpredictably across the stereo field, in classic Eddie Kramer style.

The fuzz-wah parts were harder to deal with. Adding effects such as delay didn't really make any difference, since the core



Getting the vocal to be bright enough without sounding harsh and spitty was tricky. Oeksound's Spiff plug-in was a great help in overcoming this challenge!



With four guitar and four keyboard tracks, the mix threatened to become overcrowded. Automation was used extensively to bring these in and out, thus adding variety while saving space for other elements.

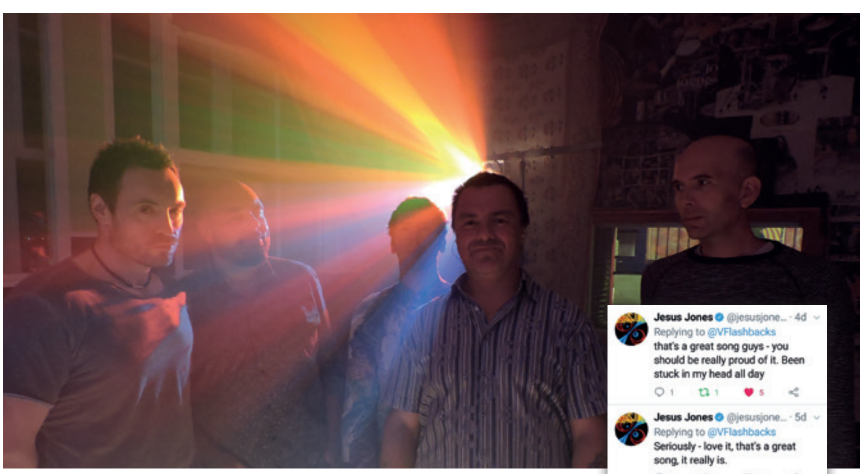
sound was little more than a continuous wash of noise. So, as well as trying to tame the fizz using EQ and cabinet simulation, I took a more off-the-wall approach to making things spacey, using LMDSP's fascinating Superchord plug-in. This gives you a set of tuned resonators modelled on piano strings, which respond sympathetically to the input signal and which can be manipulated in all sorts of odd ways. By loading up promising presets and tinkering with the controls, I was able to generate something that had more movement and life than the raw tracks, and a warmer, less harsh tone.

Emboldened by this success, I decided to take my psychedelic experiments further. A second instance of Superchord helped the keyboard parts become less static and more otherworldly, and I created another off-the-wall modulated delay effect using Eventide's Fission plug-in, fed both from the keyboards and the guitars. This got a bit out of hand if I left it running all the time, so I used automation to bring it in at strategic points.

Finally, I also wanted to create something a bit unusual to differentiate the song's middle section. This was much more sparse than the rest of the song, and left the snare drum sounding a little stark and proud of the mix. I discovered that I could use Soundtoys' Crystallizer plug-in to create a delay with a rapid pitch dive, so I fed this from the snare bus and added extra reverb.

Flash Forward

Sometimes artists get very attached to a rough mix, and it can be a struggle to persuade them of the merits of a remix, even when you feel you've effected a technical improvement. Thankfully, that wasn't the case here, and the Vietnam Flashbacks were very receptive to the changes I'd made — which was lucky, because the end result of all this work was a mix that sounded quite different from the band's original. As I mentioned at the start of this article, frequency balance and the balance of instruments go hand in hand, and both were altered fairly radically in my interpretation. At the end of the day, though, it's still the song and the performance that count, and the goal of mixing is to put them across most effectively. Reshaping the balance of the mix allowed it to translate better to different speaker systems and brought it more in line with commercial tracks, but above all, it helped the most important elements of the song to shine.



Remix Reaction

"The original mixes, whilst sounding good to us, just do not compare to the depth and clarity of these mixes. The stand-outs are how amazing the bass and drums sound and how polished it is. Everything just sounds better. It took us all by surprise, and the rougher original live recordings now sound like songs ready to be released. So far we have had them played on the radio and had the above comments."

STUDIO S.O.S

YOUR STUDIO PROBLEMS SOLVED

Photos: Dennis J Wilkins.

Follow one *SOS* reader on his quest to transform a box room into a studio worthy of the name!

DENNIS J WILKINS

Studio SOS articles are typically hosted by Paul White and Hugh Robjohns, who travel to the far corners of England in search of chocolate Hob-Nobs and studio spaces needing help. Alas, however, I couldn't talk them into driving to Colorado when I was embarking on a home studio upgrade. Instead, I had to read a number of their columns, search for some online advice, and carry out my own experiments with room modes, acoustic treatment and speaker settings. And the results were well worth the effort!

The Room

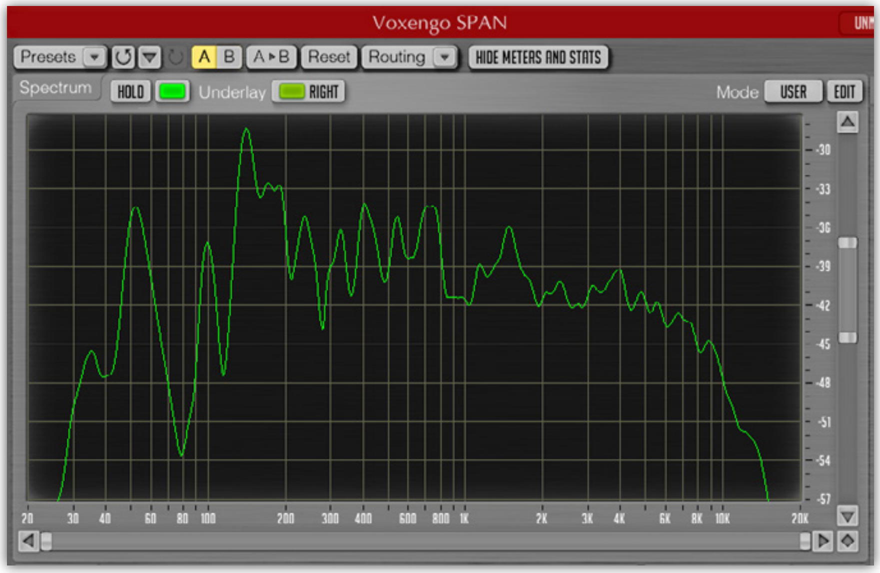
As many of you may find yourself in a similar situation, I'll describe the room I had to work with, show initial measurements I made before any treatment, and then the final (at least, the most recent) results. I started with a basic 'box' room, about the worst possible shape, with equal width and length and a height only slightly shorter. This is the kind of space that many feel cannot yield good acoustics, but there are a number of techniques to alleviate the potential issues that such a room promotes. My room is approximately 3.6 metres (12 feet) on each side and about 2.6 metres (8.5 feet) tall. The walls and ceiling are plasterboard, and the hardwood floor is carpeted.

Originally the room was used for video editing, and acoustics were not a concern since I was using headphones when working on video. These days I use the studio primarily for mixing and mastering, with some audio recording, such as vocals and guitar. For mixing and mastering I want absolute clarity, a smooth monitoring frequency response, and minimal contribution of reverberation from the room.

The original upgrade project was started about six years ago, and for this article I wanted to illustrate how the room sounded before I added any acoustic treatment, but it took considerable time searching through my audio PC to find my original test files! I found a 20Hz-20kHz test result that shows rather wide swings with peaks and dips that span over 20dB within a single octave. In Screen 1 you can see a plot I made from this data using Voxengo SPAN (a wonderful free spectrum analyser that you hopefully already have). While a 1dB change is difficult to detect by ear in most cases, and 3dB is detectable but mild, a 20dB change (a 100:1 change in power) is extreme!

Basics

There is a lot of excellent information on acoustics and treatment available on the Internet (and in *Sound On Sound*, such as 'Choosing & Using Porous Absorbers', July 2015, By Trevor Cox; 'When Are Diffusers A Good Idea?' by Paul White



Screen 1: The response at the listening position, prior to treatment.

and Hugh Robjohns, May 2015; 'A Beginners' Guide To Acoustic Treatment', December 2009, by Chris Mayes-Wright; as well as the many Studio SOS articles over the years). As Paul and Hugh have pointed out (many times!), there are a few basic principles to consider. For one thing, symmetry is good (left-to-right that is, not necessarily front-to-back!). Monitors should be placed equally distant from the left and right walls, and preferably spaced some distance from the wall behind them. The listening position should also be equidistant from the left and right walls, but not equidistant from the front and rear walls, the centre of a room often being a worst-case location! Rooms that are strongly asymmetrical side-to-side will be difficult to tame, though not impossible to improve. Asymmetry front-to-back can actually be helpful by scattering sound more randomly, but most of us with home studios need to deal with a basic rectangular box shape.

Even if you have a nice symmetrical room, there is still need for acoustic treatment. And it's good to start with a few critical positions: the so-called mirror points. These are the spots on the walls and ceiling (and possibly the floor) where sound is reflected once on its way from a speaker to your ears. This sound will arrive slightly later than the direct sound from the speakers, and will combine with it to give undesirable comb filtering. The delay depends on the size of the room, and can be estimated by considering the speed of sound (at 20 degrees Celsius it's 1125 feet per second, or 343 metres per second for you metric folks). In the case of my room, the side mirror points are about 1.2 metres (4 feet) from the closest speaker on each side, and 1.8 metres (6 feet) from my head, with a total travel distance of about 3 metres (10 feet), while the direct distance is about a metre (3.5 feet). So the delay is equal to the difference in path lengths (two metres or 6.5 feet) divided by the speed of sound. This works out at about 5.8ms, which is not enough for the reflected sound to be heard as a discrete echo; instead, it recombines with the direct sound to introduce filtering and coloration. The ceiling mirror point is about 1.2 metres (4 feet) above my head when seated, with a total travel distance of about 2.4 metres (8 feet), for a path length difference of about 1.4 metres and a 4.1ms delay, causing more audio confusion. And the cross-speaker mirror points (left speaker to right wall, and right speaker to left wall) yield another confusing reflection delayed by about 9ms. If these reflections are not treated, they will significantly degrade clarity and stereo imaging.

However, because low frequencies have longer wavelengths than high frequencies, LF reflections from mirror points won't be a significant issue in a typical project studio: the wavelengths are so long and the room dimensions so small that the time-of-arrival delays don't cause comb filtering. Instead, the direct and reflected waves are only slightly phase-shifted with respect to one another, so tend to add constructively, rather than create destructive notches (although the room modes still do that, » of course, because the path-length differences involved are much greater). This is why you've seen Paul and Hugh make significant audible improvements by placing simple acoustic foam on the walls, even though such foam has a decreasing effect below 500Hz and almost no effect below 200Hz.

I decided to use absorbers with a broader range of absorption than just foam, and built a number of these using dense Rockwool covered with an open-weave cloth, and faced with commercial acoustic foam. This saved considerable money over buying commercial units, and worked very well. Note that the wooden frame spaces the back of the Rockwool a few centimetres from the wall, an arrangement that actually improves low-frequency absorption a little. My basic units are 61cm (two feet) square.

I placed the side absorbers directly on the walls, but the ceiling unit required some thought. I ended up using sturdy screw-in hooks (screwed into the joists, not just the plasterboard!) and nylon cord. This enabled me to angle the unit (which is 2x4 feet, or 61x122 cm), which looks nice and also provides additional spacing from the ceiling which, again, helps trap low frequencies.

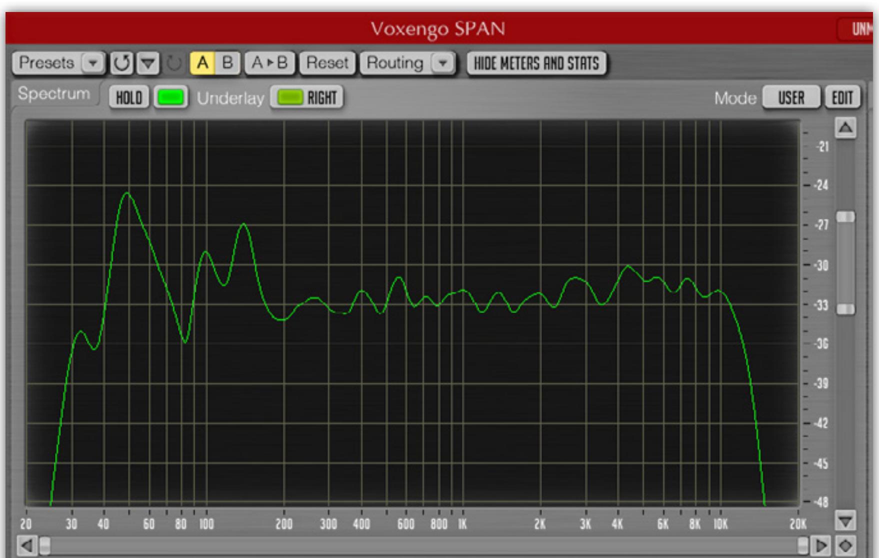
Vocal Booth

A temporary vocal booth can be created by relocating two of the hanging wall absorbers.

My studio is small enough that a typical vocal booth isn't practical, and since I'm never recording a full live band I don't need the isolation. With the treatment in place I found the room quiet enough to record vocals and small acoustic instruments (guitar, bass or harmonica). Positioning a musician and/or singer in the rear corner of the room allows a cardioid or figure-8 mic to be oriented to minimise pick-up of computer noise, but in the default room configuration, this arrangement introduces some boxiness into the recorded sound. I actually have a reflection shield that fits on a mic stand, but have not been happy with the tonal changes it imparts, so I came up with my 'non-booth' using the same absorbers that normally hang in front of the rear windows, simply moving them to an additional set of ceiling hooks. As you can see in the photo, this corner placement blocks my equipment closet somewhat, so I don't want a permanent pair of absorbers hanging at this location, but it's easy to move the hanging absorbers as needed. The recorded sound using this arrangement is not totally dry and has a short reverberation time with good tonal quality, and is very usable.

Deeper Issues

As Screen 1 shows, the room effects below 1kHz were severe, so I bought some commercial corner 'bass traps' made of dense acoustic foam to use in the rear corners (the front corners include a door that precluded large blocks of foam). Such absorbers are not very effective below 200Hz, but still help in the low mids. You can make your own corner traps simply by placing tall, open-backed panels with Rockwool across the corners at 45 degrees to each wall — again, having an open space behind will increase the effectiveness at low frequencies.



Screen 2: The room response, post-treatment but before applying IK Multimedia's ARC room correction.

I wanted even more bass trapping at the rear of the room than the corner traps provided, but with a large double window and another desk where I have my video/image editing system, how could I accomplish this? Years before, I had blocked the windows with fibreglass panels to shut out the light (for video and image editing). This had the added advantage of reducing external noise, but the aluminum-faced panels reflect sound within the room very effectively! Since my imaging computer and printer desks are still at the rear of the room, I couldn't support panels with floor stands, so I decided to hang two panels, each identical to the cloud panel, from the ceiling using the same type of hooks I had used for the ceiling absorber. Again, I spaced them from the window wells by about 10cm (4 inches) to increase their effectiveness as mentioned earlier. This design approach led to another bonus (see the 'Vocal Booth' box).

Over several months I added another

Absorbers were hung from the ceiling, a short distance away from the walls to increase their efficacy at low frequencies.

1.5 square metres (16 square feet) of panels to corners, ceiling-wall junctures, and areas on the side and front walls. These panels were dense Rockwool covered with fabric, mostly without wood frames. The results showed good improvement above 200Hz; the lower frequencies still had some large swings due to standing waves, but these variations were only about half the magnitude they had been before treatment (see Screen 2).

Another Trick

It's no secret that I also use electronic correction to help tune my room (see my January 2018 review of IK Multimedia's ARC 2.5: https://sosm.ag/arc25), and this kind of system works wonders for the lowest frequencies. As I indicated in my review article, this room correction system not only adjusts frequency equalisation, but also does something (proprietary) to

The author, Dennis J Wilkins, in his studio.

clean up phase/timing issues. Of course, it cannot eliminate really bad ringing or very long reverberation tails, so it works best in a well damped room.

Over the past few years I've changed my primary monitors twice and added a subwoofer, each change requiring new measurements and new ARC correction curves. I've also changed the measurement process from using a manual swept sine wave analysed with a spectrum plug-in to a procedure using Room EQ Wizard (www.roomeqwizard.com), an excellent donationware program that can measure not only frequency response, but phase response, harmonic distortion, sound energy decay characteristics and other valuable measurements for understanding room acoustics. I used it to plot the frequency response (this time up to 24kHz) of six locations around the monitoring sweet spot, and as you can see in Screen 3, there are minor variations — ±3dB or so, up to 20kHz — but no large bumps or dips (note the vertical scale is 5dB per line in this case).

The results are impressive, especially for a small box room. Of course the low-frequency response far to one side, towards the rear or in a corner of the room is not this flat due to residual standing waves, which brings us back to bass traps. Install as many as you can fit in the room before you try mixing! An electronic correction system can help at the listening position but the overall room will benefit more from any electronic method if you can tame the low end first. As Phil Ward noted in his review of Sonarworks Reference 4 Studio Edition, another electronic room correction system (*Sound On Sound*, October 2018), "It's really important to consider when configuring Sonarworks just how much it asks of the monitors at low frequencies... this issue will be particularly important with small monitors, where significant equalising gain, especially around and below any port tuning frequency, might well result in nothing but distortion, noise, wasted amplifier power..." So pile on all the bass trapping you can!

Good News, Bad News

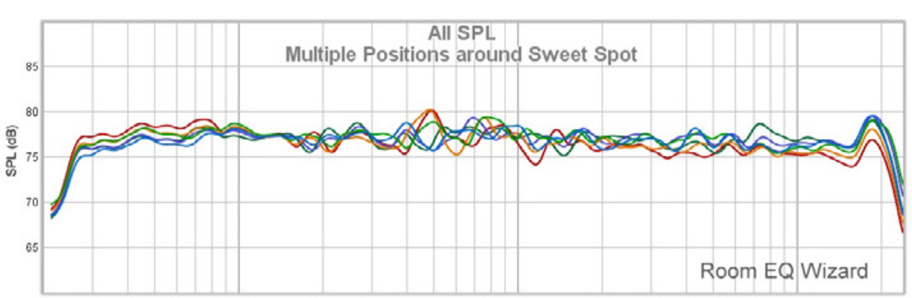
The good news is that with the changes I made, stereo imaging is spot-on, with a clear and stable phantom centre, and a smooth response over the entire audio frequency range. The bad news is that commercial recordings — CDs that I've used as references for years — now clearly exhibit audio defects that I never detected playing them in my car or home stereo system: sounds like piano stools squeaking, lip smacking, odd buzzes and other noises you wouldn't expect in a professional production. To be honest, I can hear these defects in my AKG studio headphones, but until I cleaned up my room, some of these anomalies were masked by reflections and room modes. These are the kinds of subtle defects that a mix or mastering engineer needs to hear.

Another thing one might consider a 'minus' is that the best results are only at my 'engineering' position, and not all over the room, as I noted earlier. But as Carl Tatz indicated in his article, 'The Elephant In The Control Room' (*Sound on Sound*, November 2015): "All you should be concerned about is the listening position; to hell with the rest of the room. This is as true in a bedroom as it is in a well-designed commercial studio control room... there is only one position that can be accurate and that's your position in front of the monitors." As my most recent results show, I've attained about as fine a response as possible in my small room.

I want to thank Paul and Hugh for publishing many good pointers in *Sound On Sound* over the years, and the many *SOS* authors who have provided valuable information on room treatment and recording technology. And my thanks to all the *Sound On Sound* staff for producing such a fine publication.

Room EQ Wizard

The Room EQ Wizard program is free (although the author will accept donations, and if you use the program, you'll likely feel he deserves any spare cash you have!) and is available for PC, Mac and Linux systems. With this software and a reasonably flat measurement mic, you can measure how your space is performing, and gain insight into the effects of changing room treatment, altering speaker placement, and other adjustments. Room EQ Wizard can measure not only room response, but also help you identify the sound character of speakers, microphones, preamps, amplifiers, and even software audio processors.



Screen 3: The room's frequency response, as measured at a number of different locations.

Session Notes

THE PRACTICAL CRAFT OF RECORDING

Our engineer records a rock trio while trying to also offer sufficient options to create a drum sample library!

MIKE SENIOR

Even though affordable project-studio equipment has allowed many producers to record and mix music without having to watch the clock on financial grounds, time often remains the biggest enemy. This is because the lack of formal session constraints encourages people to hedge their bets sonically, deferring important sound decisions until mixdown rather than recording with a clearly envisioned final result in mind. Much of the advice I've offered in past instalments of this column has been about how to avoid this trap by committing to recording things from the outset the way you want them to sound in the end — a much more time-efficient working method.

However, I'm occasionally asked to record in such a way as to keep lots of sonic options open, perhaps because the needs of the arrangement are still uncertain, or because I want to allow a different engineer enough freedom to develop their own vision for the mix. One such session was when I was recently asked to track a rock trio at a local commercial studio, and in this case, the drummer had an additional reason for wanting maximum mixdown flexibility: he also wanted to use my mic setup to sample his desirable '90s-vintage Tama Star Classic kit so that he could develop his own sample-library product.

Multi-purpose Band Recording

With this in mind, my goal with the miking was to be able to achieve a satisfying a band timbre without recourse to heavy mix processing, no matter how dry or roomy the

In this month's column, we mic up a three-piece rock band for maximum mixdown flexibility.

Photos: Daniel Plappert

Featured This Month

This month's column features Marcus Boeltz (www.marcusboeltz.net) on drums, Christian Bolz (www.christianbolz.info) on electric guitar, and Tobias Knecht (www.tobiasknecht.de) on bass guitar. We were recording together at Munich's Mastermix Studios (www.mastermixstudio.de). If you'd like to check out the drum sample library that came out of this session, it's called 'Real World Session Files 1', and can be purchased from www.wavepowerblog.com.

target sound at mixdown. In addition, I was keen that the mic signals should also allow the mix engineer some freedom to adjust the relative balance and tone of all the instruments if they wished. More specifically, the basic plan was to capture a full set of close mics for the drums, cymbals, and bass/guitar amplifiers that would be able to stand on their own if a dry mix was required, but then to supplement those with several stereo mic pairs capturing the whole band sound from different distances. Blending any or all of the mic pairs with the close mics would allow me to generate a wide range of more ambient perspectives, while retaining a useful degree of balance control over individual instruments.

As far as the drums were concerned, it might seem on the face of it logical to begin work with the close mics in this situation, but I actually chose to start with the closest mic pair. This is partly just a reflex on my part, but there are good reasons I've made it a habit. Firstly, the process of finding a sensible miking position for overall kit pickup involves comparing the live-room and control-room sounds by ear, so you immediately start listening critically to the natural acoustic sound of the drums. And once you manage to get that first mic pair sounding representative in the control room, that can provide a powerful reality check when you're judging the success of your close mic positions. In other words, because that first mic pair's more distant placement typically picks up a more natural sound, it really highlights whether your close-miking choices are misrepresenting the timbre of any individual drum or cymbal.

The mics I chose for this first pair were the studio's vintage AKG C414 EB large-diaphragm condensers, which I like for their smooth and unhyped tone, and I set them up in a favourite position of mine a little above the drummer's head, roughly two feet apart, and angled inwards to direct their brighter on-axis tone towards the snare drum. The mics are multi-pattern, and I opted for the cardioid option so that the mics would capture the kit's full width with a sensible balance, while also rejecting a certain amount of room ambience. The sound we got from the mics reminded me why I like this miking approach, as it doesn't over-emphasise the cymbals in the balance the way a more traditional 'overhead' miking position can. It was also good news that the drummer naturally balanced his kit with the snare dominating over the cymbals in the room, because that meant that if I based my drums mix on the overall-pickup mics, I wouldn't have to use as much of the (inevitably less natural sounding) close mics to get a sensible rock balance.

Drum Kit Close Mics

Speaking of which, my next stop was getting the close mics up and running. In addition to the overall-pickup mic pair, I put up four further microphones over the kit, targeting the two crash cymbals, ride cymbal, and hi-hat from around a foot above each. My first choices for the close cymbal microphones were Neumann U87s, as these have quite a smooth high-frequency »

For the kick drum, Mike combined an Electro-Voice RE20 dynamic mic inside the drum with a Neumann U47 FET condenser mic in front of the resonant head.

character and I was concerned that the close mic placement might over-brighten the tone. In the event this proved an unnecessary precaution, so I swapped them out for my small-diaphragm Shure KSM141 condensers, using their cardioid mode and high-pass filter switches to reduce the levels of kick and snare ambience. I put up the studio's Neumann KM84 and KM85 mics for the ride and hi-hat respectively, on the basis of their fairly neutral and understated tone. (In case you're unfamiliar with the latter model, it's pretty much identical to the KM84, but with built-in 6dB/octave 200Hz high-pass filtering — a benefit once again for reducing spill from the drums.)

I used small-diaphragm condensers for the toms too, though in this case they were the cardioid Avantone CK1s that I've previously had a lot of luck with in this application.

Inside the kick I had the Electro-Voice RE20, a cardioid dynamic mic, poking into the drum cavity through a hole in the resonant head to capture a tight, percussive sound with plenty of mid-range beater definition. Positioning mics inside a kick drum can be a bit fiddly, because of the strong resonance modes inside the shell cavity, so I was glad to have the assistance of fellow engineer Simon Gordeev (http://simgo.de) on this particular session, as he was able to lend a hand moving microphones around incrementally for me while I listened in the control room and directed him via headphone foldback. This working method can feel like something of a luxury in project-studio environments, either because the live room and control room are one in the same, or because there's a shortage of capable personel, but it doesn't half speed up the workflow when you can manage it!

Complementing the RE20 was the rich warmth of a Neumann U47 FET large-diaphragm condenser mic placed outside the drum, roughly eight inches from the resonant head. For the top of the snare, the drummer expressed a preference for

Here you can see three of the four snare-drum microphones that were used: a Shure Beta 56 dynamic over the drum; a Shure KSM132 small-diaphragm condenser under the drum; and an ADK A7 large-diaphragm condenser between the kick and snare to capture additional shell tone and rattle.

Session Impressions

Marcus Boeltz: "The brief for Mike was that I wanted my drum sample library to have a modern and punchy sound with various mixing options for the end user, and that his sampling setup should also work well for the live band recordings. I've recorded drums myself a few times, but always went for one of the few proven and pretty standard strategies I knew, so I was quite excited to see what Mike would come up with. I loved the fact that Mike wasn't just pulling off any of his proven setups but did quite a lot of experimenting as our schedule allowed for that. Later on, Mike had no fewer than 17 mics on my four-piece drum kit, which was both impressive and intimidating at first, but I was more than happy with what I heard at the first soundcheck. In my mind Mike nailed the session requirements perfectly: the kit sounded fantastic, and there were ample mixing choices available. The band were also more than happy with the sound when they came in the following day, and the whole tracking session (sampling as well as the live band) turned out to be a smooth and fast-paced ride. We even left the studio ahead of schedule!"

Shure's Beta 56 supercardioid dynamic mic, so I was happy to go with that, placing it about three inches above the rim, angled a little inwards. The sound from this mic had plenty of attack, as well as a touch of batter-head resonance, but with very little cymbal or hi-hat spill — another advantage of the drummer's naturally snare-heavy balance. The under-snare mic's position mirrored that of the over-snare for the most part, and the shielding effect of the drum itself helped keep cymbal spill levels negligible, even though I'd deliberately chosen a small-diaphragm cardioid condenser (Shure's KSM132) to capture the drum's tonal brightness. As it turned out, pointing this mic directly at the snare wires made them sound a little too aggressive, so I adjusted the mic's angle until it gave a slightly more restrained outcome. An ADK A7 large-diaphragm condenser placed between the snare and kick drums helped enhance the shell tone of both instruments, as well as picking up some nice sympathetic rattle.

By this time the close mics seemed to be complementing the overall kit mics nicely, but removing that stereo pair to get a tighter sound left the snare drum sounding slightly anaemic. So I decided to bring in one more snare mic just over the drummer's left shoulder to catch a more holistic sound from the drum, the idea being

In order to retain some tonal richness for the snare drum in drier mixdown presentations, engineer Simon Gordeev helped Mike to set up a slightly more distant snare mic above the drummer's left shoulder. It took some time to find the right mic for this, with both an Avantone CK1 and a Sennheiser MD441 being swapped out before the final choice of a Shure SM7B was reached.

that this could supplement the snare tone in any mix where the engineer decided to jettison the overall-pickup mic pairs. Initially I tried a hypercardioid Avantone CK1 condenser in this role to minimise cymbal spill, but its thin timbre from this position was far from appealing. I then decided to give Sennheiser's supercardioid dynamic MD441 a go, but this was still too bright, even with its switchable treble boost deactivated, so I finally reached for that old stalwart the Shure SM7B, which produced a meatier tone once I'd turned off its presence boost switch. (The mic's built-in low-cut function was also useful for reducing kick-drum spill.)

Bass & Guitar Mics

Wherever possible, I prefer to record bands together in the same room so that there's no barrier to communication between the performers and they don't have to mess around with headphone foldback to hear what's going on. Fortunately, I was preaching to the converted here, as the band were keen to work that way as well, although they expressed some concerns that spill between the instruments might cause mixing problems and asked about whether we should put up acoustic baffles for better separation. What I've discovered in practice, though, is that if the instruments are well-balanced in the room and the cabs are angled away from the drums, then the spill levels are surprisingly low. Furthermore, if you do your best to keep the instruments fairly close together, the spill between them » acts less like unwanted reverb, and more like additional multi-mic contributions. So I suggested that we proceed with no extra baffling to start with and defer that decision until we'd heard how effective the separation was in reality.

As with the drums, I used multi-miking with both the electric guitar and bass guitar to give more scope for tonal reshaping at mixdown. For the bass, I used the same Electro-Voice RE20 and Neumann FET 47 combination I'd already used on the kick drum. Both of those mics have comparatively linear low-frequency responses, which I felt would help preserve an appropriate balance between the different registers of the performer's melodic bass lines. The electric guitarist was alternating between two different combo amps for different numbers, and I used the same general-purpose miking setup for each: a Shure SM57 cardioid dynamic mic; a Neumann U87 large-diaphragm cardioid condenser mic; and a Bash Audio RM BIV1 figure-of-eight ribbon mic that Simon had brought with him. Each of these mics has its own unique sound character by design, and I also chose positions for them to maintain that individuality — after all, you're not going to get much tonal range from three mics if they all sound the same! Once again, Simon's help was invaluable here, as he was able to sweep the mics across the front of each cab until I heard something coming through the control-room speakers that I liked.

With small amps, comb-filtering between the direct sound of the speaker cone and reflected sound from the floor can colour the mic signals, so I made a point of using angled amp stands for both the guitarist's amps. To keep spill levels low, I kept all the mics quite close to the speaker grill in each case, which necessarily overloaded the raw mic signals with proximity-effect bass boost. Fortunately, though, it was easy to remedy this with the variable high-pass filters built into the Audient ASP880 mic preamps I was using. Just for safety's sake, though, I recorded clean DI signals for both guitars (via Radial Pro 48 active DI boxes), just in case the band wanted to expand their sonic options still further through the joys of re-amping.

Multiple Room Pairs

With all these mics on the go I was now able to transition from a very dry band presentation to something more natural and open-sounding, but for more obviously roomy sounds I was going to need some more distant mic pairs too. In the end I set up three different mic pairs which covered a lot of different bases. The closest pair were DPA small-diaphragm omni condensers spaced about two feet apart and 10 feet or so behind the drummer. These delivered a nice rich ambience for the drums, without overprominent cymbal splash, and also provided a balance biased towards the drummer, on account of the guitar and bass amps being on the other side of the kit facing away.

Both of the other mic pairs were placed on the opposite side of the kit to give a more guitar-heavy balance, both arrays at a distance of about 12 feet from the kick drum. I set up the studio's lovely Schoeps CMTS501 stereo multi-pattern valve mic in a Blumlein configuration (ie. with the coincident capsules in figure-eight polar pattern with a mutual angle of 90 degrees), aiming it at the drums. Because the Blumlein array naturally recesses the levels of sound sources arriving from the centre of the soundstage (relative to those at the edges), this distanced the drums a little more. Tonally, though, these mics picked up a much brighter cymbal sound than the DPAs.

Either side of the Schoeps mic I set up a Superlux R102 ribbon mic to create another spaced stereo pair. With these mics, however, I wanted to get a much more distant sound, so rotated them 90 degrees such that they pointed their figure-eight rejection nulls at the drums — a handy little trick for creating the illusion of a larger recording room. (Another option, if you have no figure-of-eight mics, is to point a pair of cardioids in the 'wrong' direction so that their cardioid nulls point towards the sound source.) As you'd normally expect of ribbons, these mic signals had a warmer and more rounded tone than the other pairs.

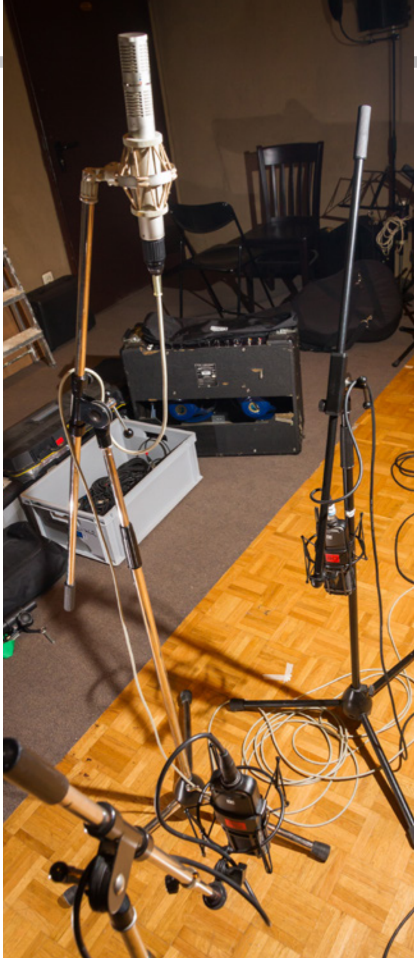
The advantage of having all these room mics was that they offered the mix engineer some leeway to decide how much ambience was added to the guitars as opposed to the drumkit, depending on how the DPA and Schoeps mic pairs were balanced. These mics also provided different amounts of ambience for the drums and cymbals. The Superlux pair then added the option of expanding the apparent size of the space.

A Plan For All Seasons

The only way to check whether you've got what you need while tracking, in my opinion, is to build a rough mix of all the tracks as you go. On this session, though,

The bass guitar amp was recorded with another Electro-Voice RE20 and Neumann U47 FET combination, but Mike also captured a clean DI signal via a Radial Pro 48 active DI box to allow the option of re-amping at mixdown if necessary.

Simon helped out again with miking up the electric guitar cabs, moving microphones in the live room while Mike listened for promising timbres in the control room and communicated with him via headphone talkback.



In front of the band were two sets of room mics: one Schoeps stereo valve mic, configured as a Blumlein pair of coincident figure-of-eight capsules; and the other a pair of Superlux R102 ribbon mics with their nulls aimed at the drums to reduce the ratio of direct-to-reflected sound picked up.

the time required to do this was significantly greater, because I had to evaluate not just one mix sound, but a representative range of different sounds in order to confirm that we had the mics we'd need in each scenario. There's no sense in killing your performers by having them play for you while you're doing this, so I did a number of test recordings as we progressed to give the musicians a few minutes to chill out while I experimented with different rough-mix blends.

It was a lot of work, but it paid off, because we ended up with tracks that needed only minimal processing to generate a wide range of different mix sounds. Furthermore, we never did end up adding any acoustic baffling between the instruments, because the separation seemed perfectly adequate as it was. But

With 31 channels of audio coming in from the live room, and a wide range of potential mixdown applications to plan for, evaluating the recordings required a lot of careful experimentation with balances in the control room before takes could commence.

don't take it from me; head over to the web version of this article on the *SOS* website (http://sosm.ag/session-notes-0419), where I've posted three different example mixes demonstrating the kind of sonic variety on offer. Plus, I've also uploaded audio files for all the raw mic signals, so you can use your own DAW software to experiment with mixing them for yourself!

Module Operandi

PAUL WHITE

Play and record external MIDI instruments in Logic.

The way that Logic handles external MIDI instruments such as hardware synths and drum machines has changed over time, and some people now find the method less than totally obvious. It is actually easy enough if all you have is a single mono- or multitimbral synth: create a new track, select Software Instrument as the track type and then choose External MIDI in the upper-left MIDI input box. In the box directly below the track type, select the audio interface input(s), mono or stereo, to which your synth is physically connected — 'Presonus 7-8', in the screen shown. In the Output section select the MIDI port connected to your synth, select the MIDI channel and, in the box below, set the audio output to the main stereo output (1-2), or to wherever else you wish to route the audio from the synth.

If you need more than one synth track, as would be the case for a multitimbral instrument, just enter the number in the field at the bottom of the New Track screen and you'll get multiple tracks, all set to consecutive MIDI channels. The track that you get looks just like a software instrument track but you'll see an External plug-in inserted instead of an instrument. Open the External plug-in and you'll see that the settings you entered when creating the track are all in place. Hit Record in Logic and you can record MIDI just as you would with a software instrument, but now your external synth is providing the sounds. At this point, patch selection can be handled by inserting Bank Select and Program Change messages in the Event List, or you can choose to set these directly on the external synth using its own front-panel controls.

An important fact to note here, however, is that although each External MIDI track displays the usual areas for inserting audio effects plug-ins, only the ones in the first track will actually do anything. Also, whatever you insert there will be applied to all the parts coming from a multitimbral instrument unless it has multiple outputs routed individually into Logic and sending separate audio streams.

Of course, if you've got enough separate outputs on your instruments and inputs on your interface you can leave your external MIDI as 'live' sources right through to your final mix. Otherwise, at some point you'll probably want to record the synth parts as audio so that you can apply individual effects and EQ to the different parts of a multitimbral source.

Despite numerous suggestions to our Apple contacts, Logic's bounce-in-place doesn't work for external MIDI, but it is easy enough to create a new audio track with the input set to the interface inputs where your synth is connected, then record that — though you must first mute the MIDI instrument track so you only monitor the sound from the new audio track. The External MIDI instrument will still play and its audio will still be routed to the audio interface: the track mute button affects only the audio you would have heard from the External MIDI Instrument track. To stop other MIDI channels from playing while you record individual parts as audio, use the Mute tool to mute all the MIDI regions in the tracks you are not recording in that pass. Individual effects and processing plug-ins can then be added to your new audio tracks after recording in the usual way.

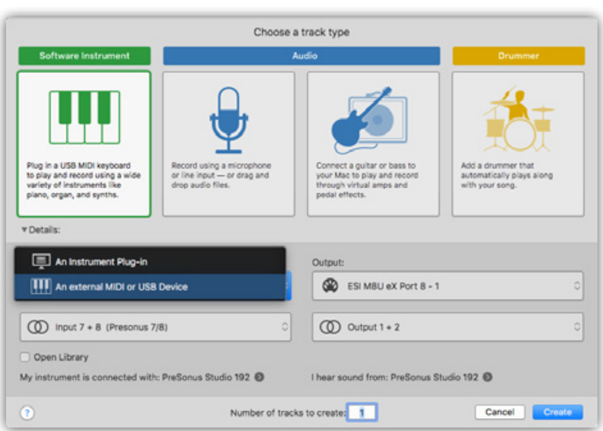
Environmental Science

That's all easy enough, but some years ago, back when Logic users had to make frequent visits to the Environment pages just to make things work, I set up all my external MIDI instruments as multitimbral Environment objects. This allowed me to enter patch names and bank-change protocols so that I could call up patches and banks directly from within Logic's Main page Inspector window. I could also use MIDI mixer controls to adjust the level and pan of each part of my multitimbral instruments, with the audio from the synth conveniently coming back into Logic via a Live Input Track.

This way of working was so intuitive and efficient that I decided to see how close to that setup I could get starting from scratch In Logic Pro X. Of course, if, like me, you created a setup like this in an earlier version of Logic, it will still open and be converted to Logic X format. Having done all the hard work, make sure you save your setup as part of a new Template so that you don't have to keep reinventing the wheel, so to speak. In fact that is exactly what I've done for my own sessions, but in the interests of science, here's one way to set up a similar system from scratch. I'm not claiming it is the only way to do it, or even the officially approved way, but it works for me.

First, rather than use the external MIDI Instrument plug-in, I started by creating my own version of a Live Input Track by creating an audio track, then inserting the I/O plug-in found in the Utilities section of the plug-in menu. Here you just need to set the output to 1-2 and the input to the interface inputs to which your synths are connected (7-8, in this case). Set the Track Input to 7-8 also. This is the track where your synth sounds will come back and also where you can record them as audio. (Once recorded, I just drag them across to a fresh audio track.)

Now, from the MIDI Enviroment menu, go to the MIDI Instruments layer and select New Multi-Instrument so a 16-part multitimbral Environment object is created for you. By default, all 16 parts are switched off, denoted by a diagonal line through each numbered block. Just click on a block to activate that part. Note that if you need to switch blocks off again, just select them and then untick



The External MIDI option can be found in the upper left MIDI input box on Software Instrument tracks.

You can use Logic's I/O plug-in to effectively turn any audio track into a live input track to monitor and record audio from external synths.

the 'Assignable' box in the Environment Inspector window. Your 16-part instrument defaults to being named after the port to which it is connected, but you can give it a more useful name. Click on the name and the Inspector opens up. Here you can change the MIDI port, if necessary, and also tick the Program and Volume boxes to allow control over these parameters. Tick all the inspector boxes for all the active parts if you need full control over them.

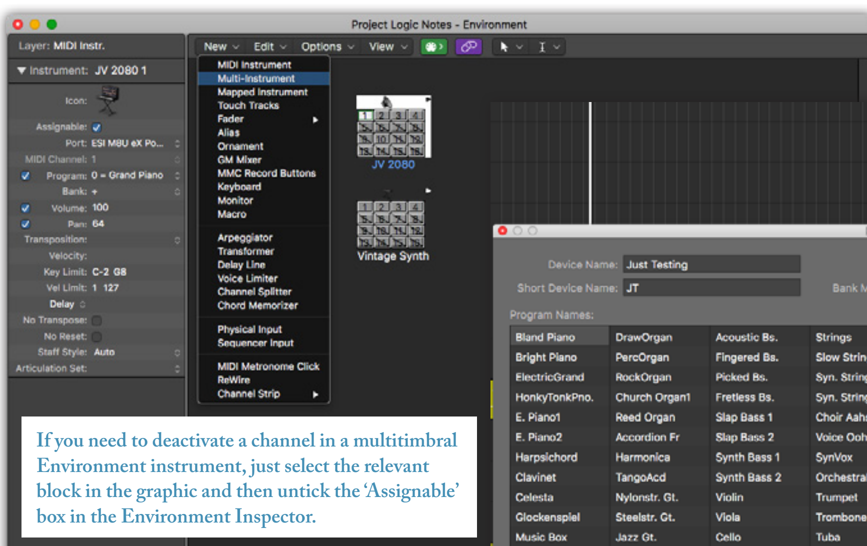
Now double-click on the loudspeaker icon at the top of the Multi-Instrument icon and you'll see a new window full of General MIDI patch names by default. Four boxes at the top of the window allow you to select a Bank Change message from a menu of common options (you can add your own bank names). You also enter a short name for the synth and a full device name, which defaults to the port name unless you decide to change it. An Options window allows you to cut, copy or paste the preset names shown and to choose either patch numbers or GM patch names. There's also a tick box in the Inspector for disabling transposition for channel 10, if you're using it for drums.

You can click on any patch name and replace it with your own, but as I remember all too well from when I owned an Oberheim Matrix 1000 (with 1000 presets), it can be fairly tedious! Fortunately, you can import lists of preset names if you can find them online, and users have even been known to resort to text-reading programs to extract patch names from a scan of the product manual. There are also completely filled-in Environment objects for some synths available online, thanks to some very altruistic users out there, and these can be imported directly into your MIDI Environment page or copied and pasted from another Logic song.

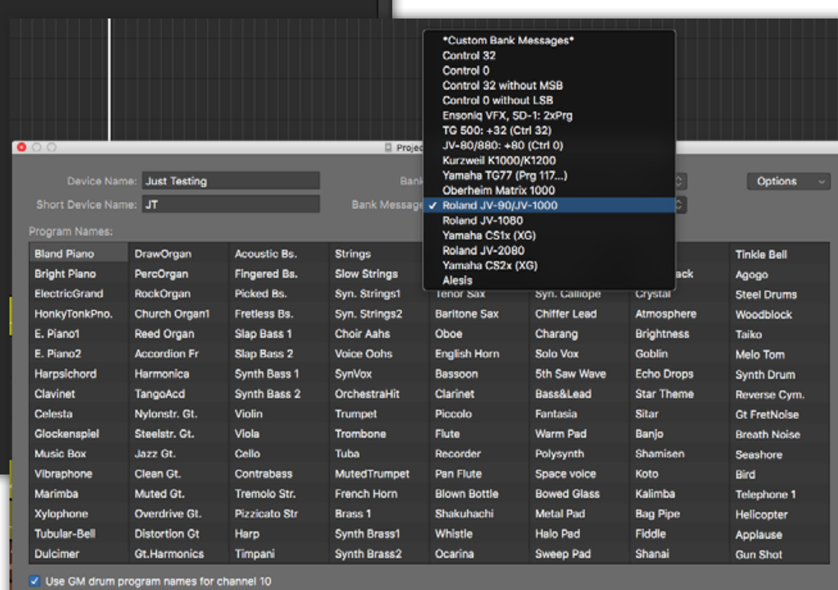
So, how to create the necessary MIDI control tracks? That's actually pretty easy and no longer involves 'knitting a scarf' in the Environment. Just select the Multi-Instrument Environment object and drag it into the Tracks section of the Main page. A dialogue box will open up, asking if you'd like tracks to be created. Tracks are only created for those instrument parts that have been activated, but if you turn off a part retrospectively, the corresponding MIDI tracks don't disappear, so you'll have to delete them manually.

I fell foul of one issue, which may be related to the fact that my synths are mixed externally, so they all go to the same two ports on my audio interface. With two or more Multi-Instruments set up, things could get confused unless I moved the audio track I created as the synth return to come below, rather than above, the MIDI control tracks in the Main window's track area. Until I moved the audio track to the end, any MIDI track I selected also selected the audio track and any port changes I made applied to all instruments, not just the one I'd tried to select. Other audio tracks placed above the MIDI control tracks worked quite normally.

The audio input track for the synths doesn't need to be set to input monitor to hear the synths, as the I/O plug-in takes care of that, and if you need to record one of the synth parts as audio, just set that track to Record and mute any MIDI regions you don't want to be included. All in all, it's not as onerous as I'd expected, though trying to find out how to deal with external MIDI synth patch names by searching Logic's Help was so frustrating that I gave up and resorted to trial and error! I hope this column will help anyone grappling with the same issues.



If you need to deactivate a channel in a multitimbral Environment instrument, just select the relevant block in the graphic and then untick the 'Assignable' box in the Environment Inspector.



Once you've completed the epic task of typing in your patch names, save as a Template and back it up in as many places as you can find —you won't want to do it twice. Now you can select your MIDI synth patch in the Track Inspector, where you can also change the bank.

Make It Live

We look at how to capture and reuse third-party samples in Live.

LEN SASSO

If you're using third-party sound libraries, it's often a good idea to capture the sounds you've found and edited for posterity in one of Live's sampling instruments. The small effort required makes it much easier to locate and reuse the sounds at a later date. The strategy is different for capturing one-shots, loops and layered sounds, and it also depends on whether you plan to apply individual effects processing to some of the sounds. I've used the Ensemble presets from Native Instruments' Antidrum Machine for my examples, as they map all available articulations across the MIDI keyboard, and that's what you're most likely to encounter in other third-party percussion and sound-effects collections.

Freeze!

The first step in capturing sounds from an instrument is to sequence the MIDI notes that trigger them. (You can skip this step when you're starting with audio files rather than an instrument, but you'll need to edit the audio files to capture just the sounds you want.) Keep the MIDI notes equally spaced and far enough apart to keep one sound from spilling over into the next. You will also need to adjust the note lengths for sustained sounds. If your source instrument is velocity sensitive, you may want to fine-tune the sequence by adjusting MIDI note velocities. The final step is to render the result to an audio file. The easiest way to do that in Live is to Freeze the instrument track and then either Flatten it, which overwrites the original instrument, or copy the Freeze file to a new audio track. If needs be, you can make the Freeze file conform to the sample rate and bit depth you've set in Live's preferences by Consolidating it (Command+J/Control+J).

The fastest way to convert the captured audio file to a Live instrument is to drag the file to the Drop area of a Simpler or to the 'Drop an Instrument or Sample Here' area of a new MIDI track, which automatically creates a Simpler. Next change Simpler to 'Slice mode' to sequence the individual sounds across the MIDI keyboard starting at C1 (MIDI note 36). In Simpler's Sample window, you'll see slice markers at each of the transients detected in the original audio file. The 'Slice By' drop-down menu about a third of the way from Simpler's left edge offers three other slicing options, and 'Region' is the easiest when using equally spaced slices — just set the number of regions to the number of slices. Other important settings are marked in red in Screen 1. Poly mode lets you play multiple hits at once, Retrigger ensures that each note retriggers the sample rather than layering it, Vol<Vel provides Velocity control over volume, and leaving Warp turned off preserves the length of each slice regardless of tempo. Although quick and easy, this approach lacks a number of options: any effects you add will apply to all slices, you cannot add, delete or swap slices without redoing the whole process, and each slice has the same settings. For example, you might want to warp slices that capture sequences so they conform to the song's tempo while leaving single hits unwarped.

By The Slice

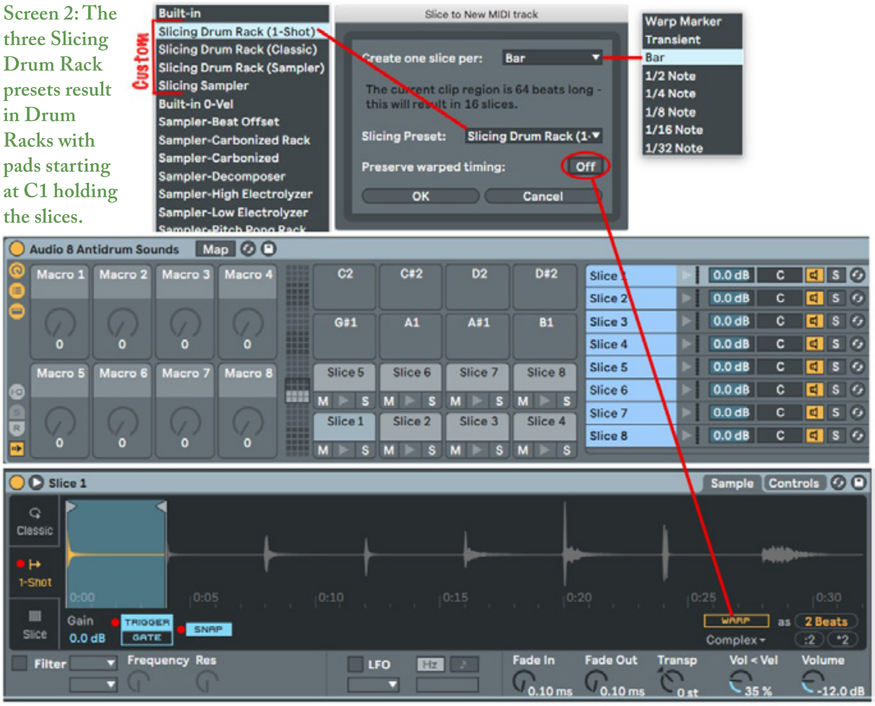
'Slice to New MIDI track', available for warped audio clips from the clip's context menu or Live's Create menu, creates a Drum Rack filled with Simplers or Samplers with a separate pad for each slice. It can also slice to a single Sampler and map the slices across the keyboard. You can slice by Warp Marker, transient, bar or note size. In each case, the slices are triggered by notes starting at C1, but you can then edit the trigger-to-note mapping. A number of slicing presets are offered, and you can add your own. To create your own Simpler or Sampler Drum Rack slicing preset, insert a Simpler or Sampler with your choice of settings in any pad of a new Drum Rack and make whatever Rack Macro knob assignments you wish. Place the Drum Rack in your User Library's Defaults/Slicing folder. For Sampler presets, place a preconfigured Sampler in that folder.

In Screen 2 I've sliced the audio clip from the previous example using the custom 'Slicing Drum Rack (1-shot)' preset. The critical settings for that preset are 1-Shot mode, Trigger and Snap. Whether each Simpler's Warp mode is enabled is determined by the 'Preserve warped timing' button in the slicing dialogue. For percussive sounds, which you don't want to be affected by the project's tempo, leave that off. After slicing, you can turn warping on for individual Simplers as needed. I prefer not to make Macro knob assignments in slicing presets because they apply to all pads. Notice that each Simpler holds the full, unsliced sample with boundaries enclosing a single slice. This lets you adjust the boundaries as needed.

The custom 'Slicing Drum Rack (Classic)' preset is intended for material that you want

Screen 1: Eight Antidrum sounds are captured in an audio clip, which is then sliced in Simpler to create a kit with triggers starting at C1.

Screen 2: The three Slicing Drum Rack presets result in Drum Racks with pads starting at C1 holding the slices.



to warp and possibly loop. Its Simpler is in Classic mode, and I use it with 'Preserve warped timing' turned on in the slicing dialogue. This preset is handy for audio files capturing sequences of different lengths that you want to trigger individually. For example, Antidrum Machine's built-in arpeggiator and accompanying velocity sequencer generates patterns that vary depending on both the number of held notes and their pitches. After recording a few of these in a single audio clip, capture them with the 'Slicing Drum Rack (Classic)' preset with warping turned on, and each sequence's tempo and looping will conform to your song's tempo.

Sample This

Slicing to a single Sampler rather than a Drum Rack assigns each sliced region to its own one-note key zone. You can then move and expand the key zones as well as combine them with velocity and Sample Select zones. The example in Screen 3 starts with three Antidrum Machine sounds: hitting a fire extinguisher with the palm of the hand, striking it with a mallet and a mallet roll. Each of these is captured in its own key zone. The hand- and mallet-strike key zones are expanded to cover the two-octave range C0 through C2. The mallet roll is restricted to C1 because Sampler doesn't offer warping, and the roll speed will change with pitch. The Root setting of each zone has been adjusted to align the three sounds' pitches. The result is a two-octave pitched instrument with the mallet roll available for C1 only. The next step is to create Sampler fade-in velocity zones for each slice so that lower velocities play only the palm strike, mid-range velocities add the mallet strike and high velocities add the mallet roll when playing C1. (You can make the mallet roll cover the whole range without pitch-based speed by Grouping the above Sampler in an Instrument Rack, deleting the mallet roll, and then adding a Simpler chain to play and warp the mallet roll. However, the mallet-roll speed will then depend on song tempo.)

Once you've captured a collection of sounds in a Simpler, Sampler or Drum Rack, consider applying some effects processing and then recapturing the results. That's especially useful with effects that change with each pass. Here are three examples:

Echo: follow your Drum Rack, Simpler or Sampler with the 'Liquourice Whip' preset for Live's Echo effect. You may need to lower the Gate Threshold setting on Echo's Character tab to hear the effect. Echo will process the same incoming sound in a different way each time. Collect some of those in a new Simpler, Sampler or Drum Rack.

Resonators: follow your Drum Rack, Simpler or Sampler with an Audio Effects Rack with several chains holding Resonators effects. Dial in a different chord on each Resonators effect. Map one of the Rack's Macro knobs to the Resonators' Note knobs and another to the Racks Chain Selector. Automate or use MIDI note ranges to change roots and chord types while playing or sequencing the source instrument.

Vocoder: enclose your Drum Rack, Simpler or Sampler in a new Instrument Rack and mute its chain (speaker icon). Create a second chain and drag in a vocal clip, which will create a Simpler. Set the vocal Simpler to Slice mode, edit the slices as needed and precede it with a Pitch effect to separate the notes that play it from the notes that play the first chain. Add a Vocoder to the second chain, set its Carrier to 'External' and set Audio From to the first chain '- Post FX'.

Screen 3: Three sounds captured in Sampler chains with individual pitch and Velocity ranges.

SIMON SHERBOURNE

Breakout

Reason may be a self-contained studio, but it can still play nicely with others.

Reason is often perceived as a self-contained software studio, but it's perfectly capable of working with hardware MIDI instruments. As with VST instruments and audio tracks, Reason treats external instruments slightly differently to most DAWs, making sure that your MIDI gear is integrated into the device Rack as well the Sequencer. This month we'll look at how this works, and learn how to monitor and record your synths without latency issues.

The External Instrument Device

Most DAWs base their functionality around tracks, relying on multiple types of tracks to manage different situations. They have audio tracks for recording and playing audio, MIDI tracks for sequencing MIDI data, and instrument tracks that combine MIDI sequencing with an audio input route.

In Reason, each MIDI track in the Sequencer belongs to a module in the Rack. Internal instruments have their own MIDI track plus an audio Rack module for connection to the mixer and insert effects. This may seem complex but it means the instruments in your project can interact freely in the modular environment instead of being contained in separate tracks.

To work with external MIDI instruments, Reason uses a dedicated device called, sensibly enough, External MIDI Instrument (EMI). This is the last device in the list of built-in Reason devices in the Browser. An EMI device has its own MIDI track, and provides a conduit to route MIDI out of any MIDI ports on your computer. The device also has gate, note, and CV connections so that you can sequence or control external devices from Rack modules rather than (or in addition to) the Sequencer. Note that the EMI device is only used for MIDI output from Reason. MIDI inputs from external sources are managed like MIDI controllers, using the Control Surfaces tab in Preferences, or routed to Rack devices via the Advanced MIDI hardware interface. More on that another time.

Screen 2: If you prefer, instruments can route directly into the Rack or Mixer instead of an audio track.

Plug Out

Let's look at the most common and basic scenario: playing an external MIDI instrument via Reason, then later capturing it as audio. To start, drop an EMI device into the Rack, then use its central pop-up menu to select a MIDI output. In Screen 1 I've chosen an RK005 MIDI interface which is connected to the synth I wish to play. Set the MIDI channel using the selector on the right of the EMI panel. With the EMI device targeted in the Rack or Sequencer, you should now be able to play your hardware device from your main MIDI controller and sequence it much like any internal device.

There are two other controls on the EMI's front panel: Program and CC Assign. Program lets you recall presets on your external device. This control can be automated, allowing you to trigger patch changes in the Sequencer timeline. The CC Assign setting affects the solitary knob on the EMI interface, which is used as a source for controller data to your instrument.

CC data from your controller keyboard will also be passed through and recorded into the Sequencer if necessary. The CC knob gives you an additional independent way to send and record CC data, and can also be controlled from CV, allowing you to connect a Rack modulation source to your instrument. If you capture some CC automation into the Sequencer using the EMI knob, you can then change its value and layer other parameters.

Audio

Unfortunately unlike, say, Ableton Live's External Instrument device, Reason's EMI module does not provide an audio input port for monitoring or recording your hardware instrument. Instead, you'll need to make audio connections yourself, either directly into the Rack or Mixer from the master hardware I/O module, or via an audio track.

The way you choose to connect audio depends on how you like to work. An audio track is the simplest and neatest option, especially if you want to record the audio from your external device. Use a direct connection into a Mix Channel if you're likely to leave the external gear running live throughout the project, or if you want to process the audio before recording.

The quickest way to route an input directly into the mixer is to flip the Rack, scroll up to the Audio I/O module at the top, then right-click on the input where your instrument is connected. Choose 'Route to New Mix Channel' (Screen 2) and a new mixer input module and strip will be created, patched from the hardware

Screen 1: The External MIDI Instrument device lets you output MIDI to any port on your machine.

input. If it's a stereo input, add the right leg manually.

If you want to use an audio track, add one from the Create menu, right-click menu or 'Add Track' button in the Sequencer. Then set the input using the 'IN' pop-up menu in the track's header panel in the Sequencer. As well as physical inputs, you can choose other mix channels that have their 'Use as Record Source' button engaged.

Monitoring

With a live Mix Channel, your instrument will be monitored through Reason at all times unless you mute the channel. With an audio track, monitoring will depend on the monitoring settings in the Audio tab of the Preferences (Screen 3). There are three options: Automatic, Manual and External. In Automatic mode, Reason will assume that you only want to monitor the input when recording, although you can override this by clicking the green input monitor button on the track.

Automatic mode is not so great for working with external sound modules as you'll most likely want to monitor the source while recording and editing MIDI. Manual mode may be more appropriate, allowing you to leave the audio track monitoring the live input until you've recorded it, then switch the track to playback monitoring thereafter.

In my setup I often use External mode, as I'm using a hybrid mixer/audio interface and tend to monitor my hardware sources with zero latency through this mixer instead of Reason. External mode disables the monitoring buttons on Audio Tracks, and leaves them permanently in playback mode. You then manage your monitoring manually on the external mixer: once you've recorded your part in you can mute the instrument channel in the external mixer. If you need to perform any punch-ins it's generally good to switch to Automatic monitoring so that you'll hear both the pre-recorded material and your new take as you drop in.

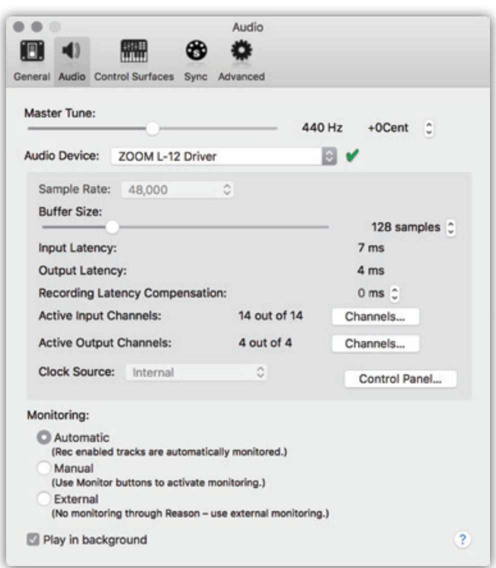
Screen 4: A comparison of audio captured from a MIDI instrument with different monitoring modes and buffer sizes. External mode should record in time, without need for manual alignment.

Latency & External Instruments

We've tackled the topic of audio latency and input monitoring before when we looked at recording guitars in Reason. But it's worth recapping, and in any case there are some important differences when capturing MIDI sequences as audio compared to playing an instrument live.

The basic picture is that audio signals coming in and out of Reason incur time delays caused by the input and output buffers (and potentially delay compensation in the Mixer). Audio playing back from Reason is delayed by the output buffer, and external audio sources monitored through Reason are delayed by both the input and output buffers. External audio sources that you monitor externally to Reason (as I do with my Zoom mixer) are not delayed at all.

These timing discrepancies present a tricky challenge when monitoring and recording external sources against existing tracks or the click in Reason. If you're using External monitoring mode (or Manual mode with input monitoring Off) and listening to your hardware instrument directly, things work out quite well. MIDI from Reason is output in time with audio playback and your hardware will sound in time. What's more, if you record your synth into Reason, the audio will be nudged to compensate for the buffers and should line up to the original MIDI events.



Screen 3: Reason's master monitoring mode determines how audio latency is managed during recording.

Automatic monitoring mode is a different story. In this mode the audio you hear from your hardware is being delayed by the buffers so will sound late. How noticeable this is obviously depends on the size of your buffers. What's more, Reason does not compensate when recording, because it makes the assumption that you're playing live and compensating by ear. Of course when it's a MIDI track triggering back your synth this isn't the case, so you inevitably end up with audio recorded late in the timeline. Screen 4 shows some recordings made with different monitoring modes and buffer sizes.

Reason's manual offers some suggestions for time-aligning MIDI controlled sources, such as adjusting slice markers or using the Regroove mixer's Slide setting. We'll look at these next time alongside considerations of sync with MIDI Clock, Ableton Link, etc. Honestly, though, the simplest approach if you want the convenience of monitoring through Reason is to get your buffers as low as possible in the first place, then manually nudge recorded audio after it's been captured. To do this, zoom in on the start of your recording (like in Screen 4), switch off Snap, then trim the front of the audio clip right up to the first event. Then move the audio in time to visually align it with the original MIDI notes.

Export Economy

Need to export different mixes from the same project? Cubase Pro 10 makes it easy...

JOHN WALDEN

Engineers often need to create alternative mixes of a project. It's common, for example, for a professional engineer to be required to provide a 'vocal up' version of a mix, but there are various reasons any of us might want to create alternative versions of a project — different arrangements that extend or shorten a song, different effects options, or perhaps broadcast-friendly edits to disguise expletives! Whatever the reason, Cubase has a number of tools that can help, and in this article I'll explain how the Arranger Track and Cubase Pro 10's MixConsole Snapshots can help you create alternative mixes and arrangements more easily. Most of what Cubase offers is good, but I'll also discuss workarounds for a few 'cons'.

Power Arrangers

I looked at the Arranger Track (in Pro, Artist and Elements) in *SOS* July 2010 (http://sosm.ag/cubase-0710), when considering how to create advert-friendly, 30-second cues from a longer project. It basically provides a way to move the playhead to different points in your arrangement on playback, so it can be used for any kind of timeline-based re-sequencing of a song's structure. Having created an Arranger Track (in the same way you create any other kind of track), creating a new arrangement requires just a few simple steps.

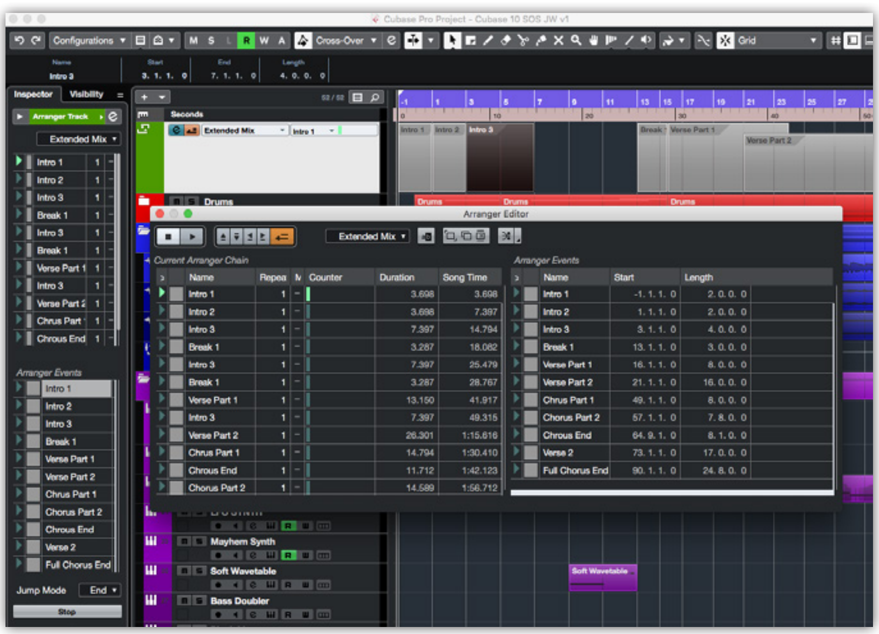
First, create some events on the Arranger Track. Each one defines a time/bar range, and they can be played back by the Arranger Track in any sequence you define. Events can be given unique names to replace the defaults (A, B, C etc.), and it's fine for events to overlap — for instance, you could create separate four- and eight-bar events starting at bar 12, beat one. Next, in the Arranger Inspector panel or in the Arranger Editor window, create a chain of these Arranger events; add and move them into the sequence you want to hear on playback. For each instance of each event, you can choose the number of times it should loop before moving to the next event in your chain. Finally, toggle on the Activate Arranger Mode button (next to the 'e' edit button for the Arranger Track in the Project window's channel list). Now, on playback, instead of playing in linear fashion along the timeline, Cubase will follow the nonlinear sequence of events specified in your Arranger chain.

This is all super-easy to do, but it's a really powerful feature because it allows you to experiment with all sorts of timeline-based variations of your musical arrangement without dragging parts around your arrange page. There are a few more detailed considerations though...

Since I first wrote about this in the July 2010 article, Steinberg added the ability to create multiple Chains via the Arranger Track's Inspector panel. Individual Chains can also be duplicated and renamed. So, assuming your Project contains all the necessary materials (audio tracks, virtual instruments, etc.) for all the different mix versions you wish to create, you can now keep all your arrangements in a single master project, simply switching between the different Chains to choose which you'll hear on playback.

At some stage, you'll want to render your various arrangements using Export/Audio Mixdown, and there are three steps to this process. The first requires you to use the Flatten Chain option (in the Arranger Editor window or the Arranger Track's Inspector). This lays out the current Arranger Chain as a conventional linear project on the timeline. You can choose in the Arranger Editor to create this flattened version in a new Cubase project or, if you prefer to keep everything in a single master project, you can select the Current Project option as the Destination. Once flattened, you can use the Export/Audio Mixdown command, as usual. A third step is to visit your project's History panel and move back to the step where you flattened the project — this will return you to the un-flattened version in the Project window while, critically, retaining the audio mixdown you just created via the export process.

Even if your Arranger Track sequence plays back smoothly, listen carefully to the flattened version prior to the final export. Problems can occur, for example, where audio or MIDI clips have elements that are cut at the start or end of any Arranger »



The Arranger Track now allows you to construct multiple Arranger Chains — alternative arrangements of your track — in a single Cubase project.

SAMSON

SATELLITE

USB / iOS Broadcast Microphone

SOMEWHERE,

SOMETHING

EXTRAORDINARY

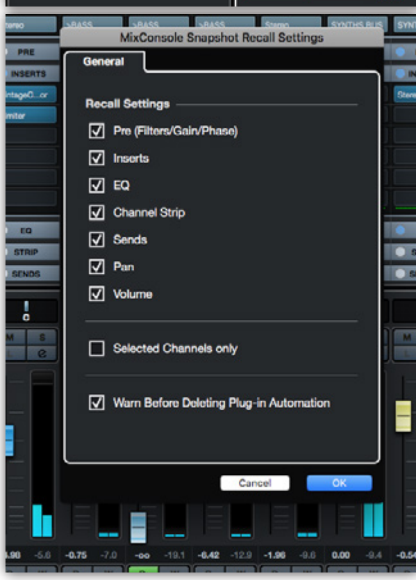
IS

WAITING

TO BE

DISCOVERED.

Cubase Pro 10's new MixConsole Snapshots system makes it easy to experiment with different level, pan and processing options, amongst other mixer settings.



» Events, and you might find that some minor editing tweaks are needed to address this. If the flattened version requires any major surgery, this is perhaps one occasion when flattening to a new project is a more sensible choice, since it would otherwise require you to undo a lot of steps to move back to another arrangement.

Snap To It

But what if you're happy with the arrangement and simply want to create and recall different mix versions? This is where Cubase Pro 10's new MixConsole Snapshots feature comes in to play. You can create up to 10 MixConsole Snapshots in a single project, and these are saved within the project file. They're set up via a dedicated tab in the main MixConsole's left zone (alongside the MixConsole History tab, which can also be useful when experimenting with different mix options). The MixConsole toolbar has a button (the small camera icon) for creating Snapshots, and there's a dropdown menu for other Snapshot tasks, such as selecting, updating or renaming. In the main MixConsole tab, you can add notes to each Snapshot too, and that's a great habit to get into, as it's easy to lose track of lots of specific mix changes in different Snapshots.

The key thing you need to note if you're to get the best from this new feature is what information is and isn't stored as part of the Snapshot. Obviously, basic volume and pan settings are stored in a Snapshot, so typical mix-variation tasks (eg. vocals up, guitars panned wider, drums down) can all be managed using Snapshots. More impressively, though, all the main insert, EQ, Channel Strip and send settings are also included. So you can easily audition different send levels to a reverb/delay, different EQ settings or different combinations of insert effects on any tracks. The last of these is a great way to try out different compressor plug-ins on, for example, your drums or lead vocals, or for exploring different stereo-bus processing chains.

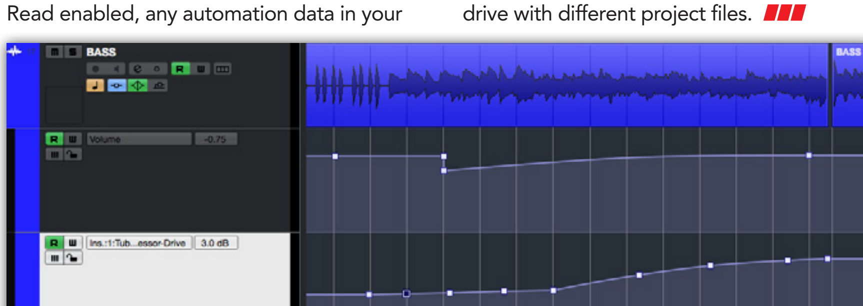
What's Not To Like?

It's important to understand that Snapshots recall settings of the MixConsole only, not of the Cubase project as a whole. In particular, note that you can't try different automation moves on, say, a track's volume or send levels and store each version in a Snapshot. And for every track that remains Read enabled, any automation data in your project will override the static settings of your Snapshots. The exception is if you've created automation for an insert effect and then load a Snapshot that doesn't include that instance of the insert effect — in this case, the insert's automation data will be deleted. Cubase can warn that this is about to happen, but it's well worth being aware of! The workaround is to use and save the bypass status for insert effects, sends and the main Rack sections, rather than actually removing the plug-ins. If you want to retain the sort of detailed level automation you might use for a lead vocal, but still create a 'vocal up' mix, then another strategy is to use a VCA Fader to boost/attenuate the vocal track — store the static VCA Fader setting in the Snapshot.

The status of the channel Mute, Solo, Read and Write button are not (yet?) stored/recalled by Snapshots, something which would have been really useful. Taking the Mute buttons as an example, storing the mute status would be handy if you wished to experiment with alternative tracks. Say you had two completely different lead vocals, or perhaps a guitar solo and a sax solo to choose between — you'd be able to simply mute the required combinations of tracks in different Snapshots. Of course, you can achieve the same end result using fader settings and insert/EQ/Channel Strip/send bypass buttons, but that's a little more fiddly.

Make Your Mind Up

While creativity and experiments are good fun, we all have to exercise a little discipline eventually — you do actually need to declare a mix 'finished' at some point! But having the ability to keep some fairly major mix and arrangement options open until the very end can be incredibly useful in some projects. Despite a few limitations, the Arranger Track and the MixConsole Snapshots facility are incredibly powerful tools that make this possible with a minimum of fuss, and without littering your drive with different project files.



The MixConsole Snapshot system doesn't include automation data. This is only an issue for insert effects, and then only if you add/remove insert effects via Snapshots when those inserts already have automation data created for them.


"The Prophet X is a beast of a synth. You get to places creatively that are really quite inspiring."
SONICSTATE
"The content and the electronics are superb. Tremendous timbral versatility. The factory programs are lovely, unique, evocative, and versatile."
ELECTRONIC MUSICIAN
"It's a hugely powerful and expressive synth. Its sonic palette is immense."
SOUND ON SOUND

Voice Choices

Understanding the voice count of your Pro Tools system can avoid nasty surprises on large projects.

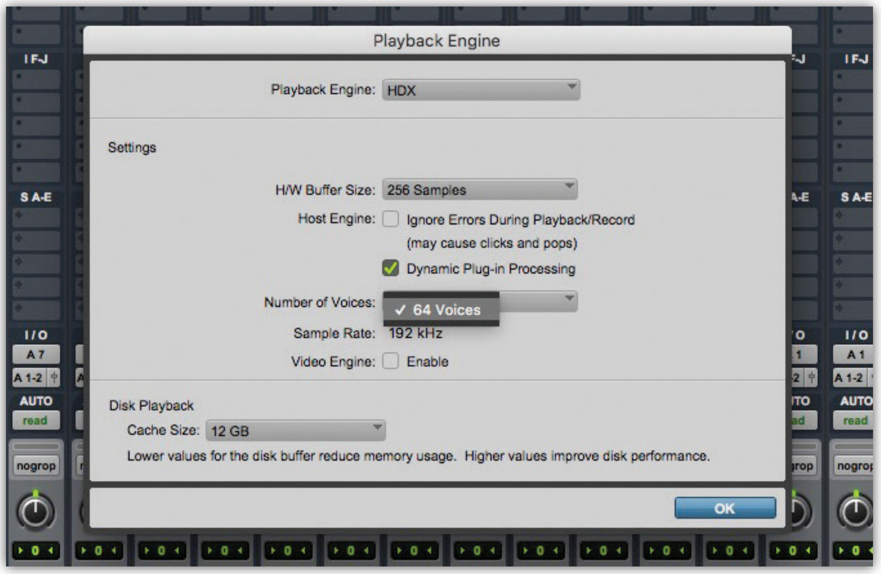
JULIAN RODGERS

At NAMM 2019 Avid announced changes to the voice counts available in some versions of Pro Tools. At the time of writing these changes are still earmarked for a future release, but the plans are in the public domain, and the easing of restrictions for users of both hardware and host-based versions of Pro Tools will be welcome news for those who push their systems to the limit. We'll look at these changes, but before we do, it is worth explaining how the voice count system works, as many users find this is something that only comes to their attention when they have exceeded the capabilities of their particular system.

What Is A Voice?

A 'voice' is an audio stream in Pro Tools. Voices can be routed to and from tracks, or inputs and outputs, and there is an upper limit on the number of voices available in any Pro Tools system. This number is independent of the number of tracks in a session. A mono stream uses one voice, stereo uses two, 5.1 uses six, and so on. People who are used to thinking in terms of tape tracks and mixer channels will probable get the distinction between the audio streams and the routing of those audio streams. Having 512 voices in a Pro Tools system doesn't necessarily mean you'll be able to work with 512 tracks with impunity, and there are several things to be aware of that will eat into your quota of available voices.

On earlier generations of Pro Tools systems, the management of voices was largely a manual operation, with the user assigning specific voices to specific tracks to get the most out of limited resources; a track which played only at the beginning of a song could share a voice with a track which played only at the end, as the total voice count wouldn't be exceeded. In modern host-based and HDX rigs this system has been superseded by two choices: Dynamic Voice Allocation and Voice Off.



The number of available voices is halved with each doubling of sample rate.

Dynamism

As most users no longer have the option of allocating voices manually, the 'dynamic' part of the name Dynamic Voice Allocation refers to a historical distinction which no longer affects most people. Some users have wrongly thought that it works in the same way as dynamic voice allocation does on a polyphonic synthesizer, reallocating voices on the fly. This is not the case. If you are running a session requiring more voices than your system has available, then the lowest-priority tracks, as dictated by their position in the session, will either be inactive or have their voice set to 'off'; either way, they won't sound without intervention.

Although the future release announced at NAMM 2019 will introduce changes, at the moment Pro Tools systems impose the following limits on the total number of voiceable tracks. The upper limit for a Pro Tools system is 768 voiced audio tracks, but the number of voices available in the system as a whole is hardware-dependent. »

If track count exceeds the number of available voices, you'll see this warning.

Tracks that Pro Tools has made inactive due to the non-availability of voices show the voice indicator in blue.

HDX systems with a single card, HD Native systems and software-only Pro Tools Ultimate systems are restricted to 256 voices at 44.1 and 48 kHz. The total number of voices available halves with each doubling of sample rate, so 128 voices are available at 96kHz and 64 at 192kHz. The voice count for an HDX2 system with two cards doubles compared to HDX1, so at 48kHz a total of 512 voices are available, and for a three-card system, 768 total voices are available at 48kHz.

Once the voice limit is reached, tracks are either made inactive on an HDX system, or their voice parameter is set to 'off', as illustrated by the voice indicator going blue. Without the option to manually assign voices, what can the user do to mitigate this?

The first solution is to reduce the number of tracks in the session. This can be done by bouncing groups of tracks together, for example by printing to an audio track, though other methods are available.

There is also an easy way to choose which tracks get affected. The Pro Tools mixer allocates voices from left to right in the Mix window or from top to bottom in the Edit window. To give a particular track priority over others simply drag it up or to the left — in a session large enough to hit voice allocation limits, it is probably easier to do this by dragging tracks in the tracks list in the sidebar.

Don't Lose Your Voices

There are reasons other than raw track count why voices get consumed on a typical Pro Tools system, and why you might hit that voice limit sooner than you might think. The first one will make sense to anyone who has ever used a tape machine with a mixer, and plenty of us who haven't, which is that punch recording in Pro Tools requires additional voices, in much the same way as it might require both an input channel and a monitor channel on your console. Playback from Pro Tools uses one voice and the input path to Pro Tools uses another during punch recording. This is why seamless punches are possible, and why it is possible to trim back from the punch point to reveal material recorded during pre-roll.

On an HDX system, crossing over from the DSP domain to the native domain and back again also incurs a hit on voice count. Because of this, it is wise to think about plug-in choice and particularly plug-in order, as is it possible to consume many voices by mixing AAX DSP and Native plug-ins on the same track. One area where this is unavoidable but not at all obvious is when using HEAT in the Pro Tools mixer, as this is a native process just like instantiating a native plug-in. (One of the updates to Pro Tools 11.1.3 included a firmware update for HDX cards which added an additional 64 voices which don't show up in the system usage meters, specifically to help with the voice burden of moving audio in and out of the DSP cards.)

> "As of 2019, Avid have confirmed that the new maximum voice count for Pro Tools Ultimate will be going up from 768 to 1152."

Voice Boost

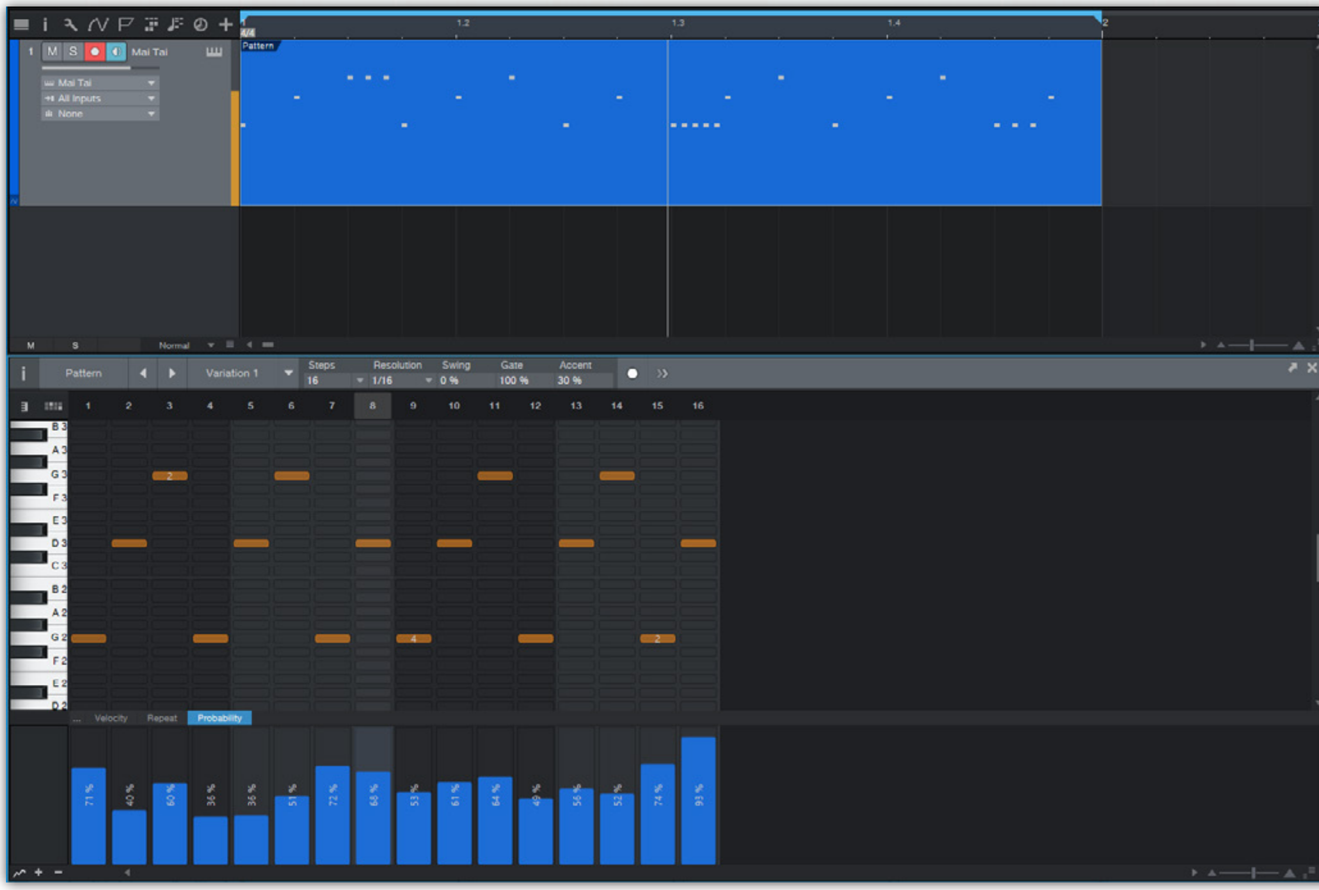
As of 2019, Avid have confirmed that the new maximum voice count for Pro Tools Ultimate will be going up from 768 to 1152. This is the maximum number of voiced audio tracks — but, as discussed earlier, there will still be other ways in which voices get consumed in a session, for example by using native and DSP plug-ins together.

All Pro Tools Ultimate users will get access to at least 384 voices, up by 128 from 256, and Pro Tools Ultimate users on host-based systems will also get the option to rent additional voices in packs of 128. The logic is that as only the biggest projects need access to more than 384 voices, a flexible approach for software-only Ultimate users will be beneficial, particularly as this could be an extra which is billable to the client in many cases. For users who regularly work on sessions exceeding this limit Avid would, I'm sure, argue that an HDX system is more appropriate. The situation is simpler for HDX users. An extra 128 voices per card will bring the total for an HDX1 system to 384, while HDX2 will support 768 and and HDX3 system will reach the maximum of 1152. All of these voice totals are for a system running 48kHz or below, and should be halved at 88.2 or 96 kHz.

There are other changes announced for 2019, too, and hopefully by the time you are reading this the much-anticipated Mojave compatibility will mean that Pro Tools is officially supported on new Macs. As an added bonus, Avid have announced they are doubling the limit for MIDI tracks from 512 to 1024. I can't imagine what I'd do with that many MIDI tracks, but I'm sure there is someone ambitious or foolhardy enough to tackle a MIDI project on that scale!

Pretty Patterns

The new pattern editor in Studio One 4 isn't just for drums!



You can make a basic arpeggiated melody more interesting by messing with the probability and repeats parameters. The latter are also reflected in the timeline (top).

ROBIN VINCENT

Studio One 4 introduced pattern-based step sequencing as an alternative mode to the familiar piano-roll MIDI editor. It's one of those features that makes you wonder why every DAW doesn't already have it. It's simple and intuitive in a way that's reminiscent of the creative tools we've come to enjoy in hardware. Pattern-based sequencing is most often used for drums, but as we'll see in this month's workshop, the Pattern Sequencer in Studio One can be just as easily directed to synthesizer and instrument sounds, and can very quickly generate something unexpected.

Insert Pattern

There are a few notions you need to get your head around in order to not find yourself befuddled by Studio One's pattern system. The first hurdle to overcome is how to put a pattern onto your timeline. The process feels quite convoluted at first, because there's little connection or flow between patterns and instrument parts such as MIDI clips. Ignore your desire to right-click or search around in menus: the Insert Pattern keyboard shortcut is the way to go. Click in the timeline where you want to put a pattern, select the track that it's for and hit Ctrl+Shift+P for instant pattern insertion.

The inserted pattern will default to being one bar long, with 16th-note resolution, so containing 16 steps. Changing the number of steps and the resolution will alter how your pattern is displayed in the timeline and also the speed at which the steps are played. This is best understood using a drum pattern, so, with a pattern inserted on an Impact XT track and a kit loaded, paint in 16 hi-hat steps plus a single kick drum on the first step so that you can identify the beginning of the pattern in the timeline. (You can make use of the little Quick Fill buttons that appear on the top right of the editor as rows of four boxes — these are super-handy, try them out!)

With 16 steps at 16th-note resolution, you get all 16 steps playing within the one bar, obviously. Increase the resolution to 32nd notes and your 16 steps are playing at double the speed, so the pattern plays twice in the same bar. If you set the number of steps to 32, the pattern will expand to show 32 steps; if you used the Quick Fill buttons, the new steps will already be populated, whereas if you drew them in by hand, they will be empty. If you go the other way and change the resolution to eighth notes then the bar is divided into just eight steps, and they'll play at half the speed of the default pattern.

Fiddling with the resolution and step counts thus affects the duration of notes in a pattern relative to the project tempo. Subsequently, it affects how the pattern fits in the timeline. If you drag the right edge of the pattern in the timeline to four bars you should be able to see the pattern repeating (remember that kick drum we put in to mark the start of the pattern?). When the resolution and step count are set to the same value — 16, 32 or whatever — the pattern will always fit into one bar. But as you change the ratio between them, your patterns occupy more or less space in the timeline. For instance, if you want a two-bar pattern, the resolution has to be half the number of steps, so 16 steps will fill two bars if the resolution is set to eighth notes.

If that relationship feels like too much maths, wait until we start setting different resolution values for each note!

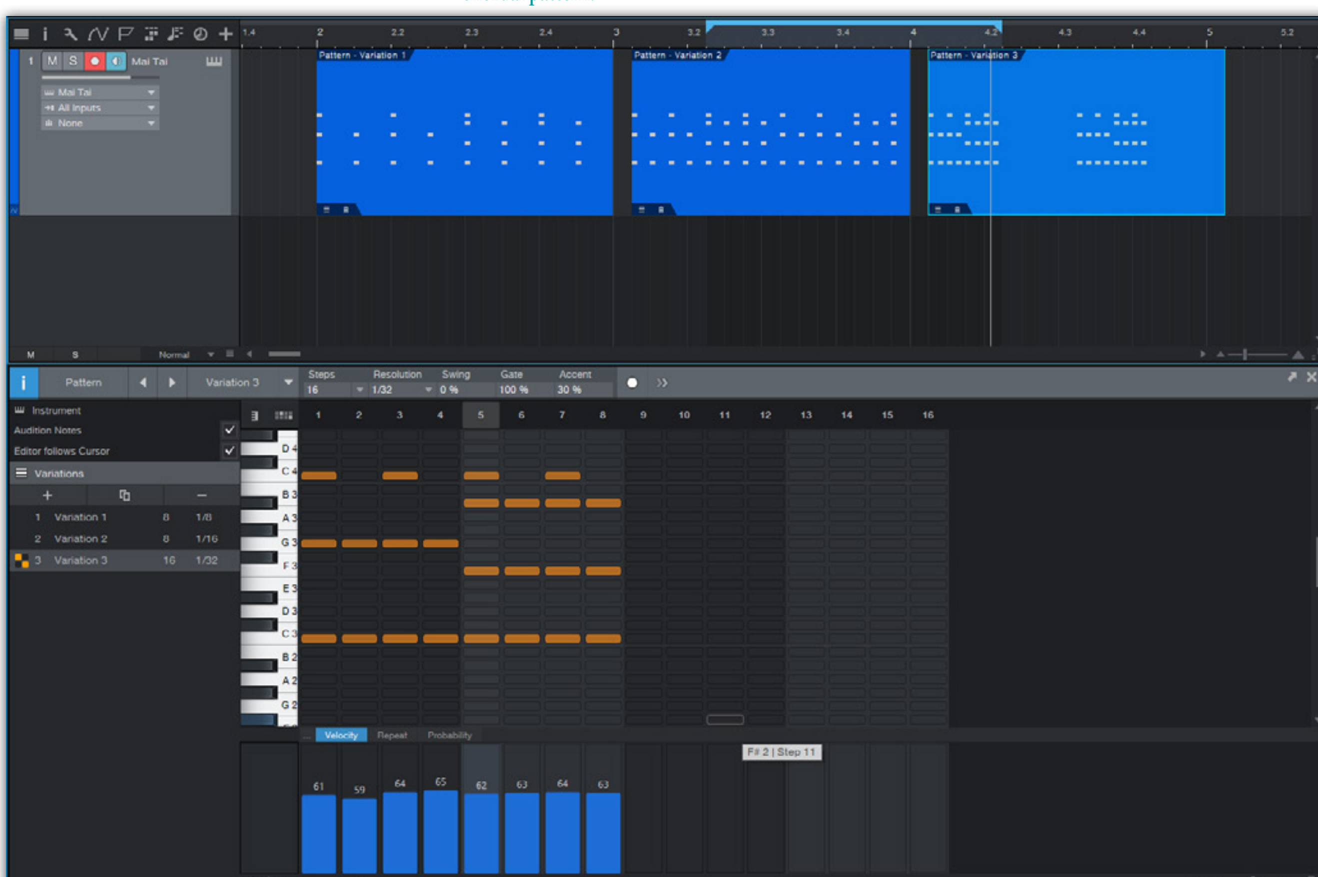
Rhythmic Melodic Patterns

On with the purpose behind this month's tutorial, which is to look at how we take a simple melodic pattern to interesting places. And we're talking about the rhythm of the notes rather than pitch. First, remove all other distractions and create a track with Mai Tai (the default sound is perfect for what we're about to do). Hit Ctrl+Shift+P to add a pattern. Let's keep it simple with 16 over 16, but you might want to bring the tempo down a bit, into the 80s. A pattern on an instrument track defaults to the piano-roll mode: this is what you need when writing melody, but comes at the expense of a few features such as the Quick Fill buttons and individual, per-note step and resolution counts. We'll come back to these in a minute, but in the mean time, pattern-based melody can feel a lot like a custom arpeggiator and that's a great place to start. If you have a MIDI keyboard attached, find yourself a three- or four-note chord, click the Step Record button and play each note in the chord in turn, over and over until you have filled the 16 steps. Alternatively, enter notes with a click. Loop the pattern in the timeline, click Play and you've got a great but ordinary arpeggiated pattern.

With about three clicks and a drag you can now completely transform this into something very different.

At the bottom of the pattern editor you'll find automation lanes. These are not the free-flowing lanes of the piano roll but per-step functions. (If they are not visible, click the tiny mountain-like button at the bottom of the piano keyboard to reveal them.) By default there will be three tabs labelled Velocity, Repeat and Probability. Let's start with Probability, which dictates how likely a note on a step is to play on any given cycle of the pattern. At 100 percent, a step will always play. At 50 percent it will play half the time, and so on. Drag your mouse through there, and your nice and clean arpeggiation quickly falls apart in very interesting ways.

Next, click on Repeat and add a couple of repeats to a couple of notes. These notes will then play the number of times you specify within the space of that single step. The rhythm and structure of your pattern has now taken on a life of its own. And while we're thinking about automation, you could also add a lane for the filter cutoff and have a different filter setting for every single step. You can of course do this for every parameter, »



Changing the step size and resolution gives instant variations on the same simple chordal pattern.

and I hope you can see how endless the possibilities become!

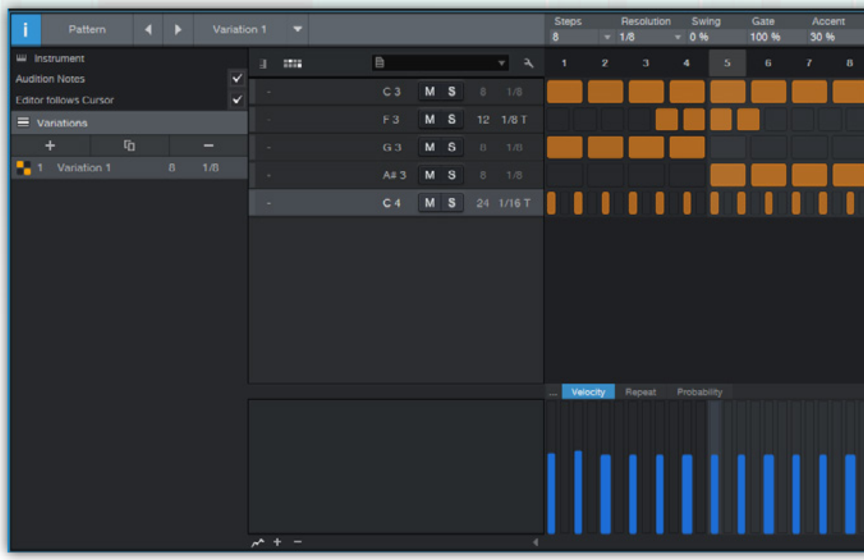
Patterns & Variations

By default, copies of a pattern in the timeline are just 'ghosts' of the original. To vary the content of a pattern you need to either insert a new pattern or generate 'variations' of an existing one. A pattern can have multiple variations, and the advantage of this is that you can select and swap between variations in the timeline within that originally inserted pattern's part. What Studio One doesn't do very well at the moment is manage these patterns. Each newly created pattern is called 'Pattern' and each variation is just 'Variation' with a number. It would seem sensible to assign a numerical value to the patterns automatically, but you have to rename them by hand. You can drag and drop variations from the pattern editor to the timeline, but only from the selected pattern.

If you open the pattern editor's inspector, you will see the variations listed; the original pattern is always called Variation 1. You can add a completely new and blank pattern variation, or make a copy of the original in order to tweak it. These variations become available from a pop-up menu when you click on the bottom-left corner of the pattern in the timeline. Variations don't have to bear any resemblance to the original in time, steps, notes or duration.

Chordal Resolution

Meanwhile, back in the timeline you can use the same step/resolution relationship to vary the meter of the pattern, from long drawn-out notes to bursts of sound. Select a polysynth sound in Mai Tai and press Ctrl+Shift+P to generate a new pattern. Let's set this to eight steps at eighth-note resolution. Click Step Record and enter a chord on each step. Copy and paste to duplicate the pattern in the timeline, and then duplicate Variation 1 in the pattern editor. In the second variation, change the resolution to 16th notes and select Variation 2 as the content of the pasted pattern. You'll now have a bar with your eight chord stabs, followed by another bar with two lots of eight chord stabs played twice as fast. You could go much further. This is a good place to start playing with the Gate percentage, or maybe start tweaking the probability in the fast patterns, and it gets rhythmically interesting really quickly.



Editing your pattern in Drum mode allows you to adjust resolution and step size on a per-note basis, which is ideal for creating polyrhythms.

Pretty Poly

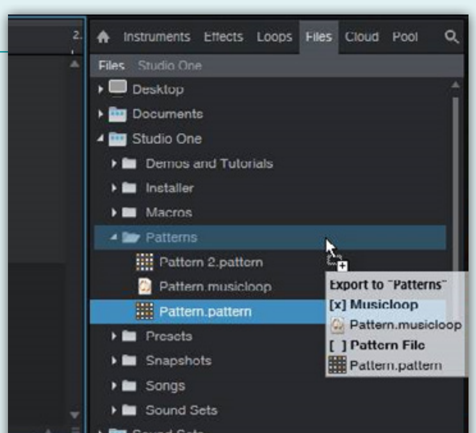
The Pattern Sequencer is a great place to experiment with polyrhythms — two or more phrases playing together that do not share the same rhythmic structure. Duplets over triplets would be the most common example, where one rhythm is playing two notes in the same space that another is playing three. As I say the theory is not that important and Studio One doesn't really offer any help in that regard. Instead let me show you where to fiddle and you can experiment for yourself.

To make this work, you need to mess about with the rhythm of individual notes. Create a simple pattern of three- or four-note chords — say four stabs of one and four stabs of another, in an eight-step, one-eighth-resolution pattern. In melodic mode, we can't access the note lanes individually, but switch to Drum mode and we can see just the lanes of the notes that were played. Now we can change the step count and resolution of each note independently on their lanes. Choose a note and set it to 1/8T and 12 steps. Choose another note and set it to 1/16T and 24 steps. Both lanes still match the bar length of the pattern, but they now use different rhythms. You can also use the Quick Fill buttons to get a better idea of how the rhythms are working together. You can set these parameters to whatever you want and have steps spilling out into other bars, taking a couple of chords to some very strange places.

The beauty of the Pattern Sequencer in Studio One is that it can generate ideas. It can revitalise melody lines, introduce probability and right royally mess about with your chordal structures. It challenges you to do things differently and to consider things that perhaps you never would have thought of.

Pattern Browsing

In other pattern-friendly DAWs such as FL Studio, all your patterns are always available in the browser. In Studio One, you have to manually export your patterns into the browser by dragging and dropping them from the timeline into a folder under Files; make sure you press Shift to change the export option to Pattern rather than Musicloop.



Studio One defaults to saving patterns as Musicloop files: press Shift to choose the Pattern option.

CRAIG ANDERTON

Plugging Away

Tidy up your plug-in menu in Cakewalk.

Plug-ins are addictive — and as they accumulate, the need to manage them increases. Fortunately, Cakewalk by BandLab has several plug-in-related tools that can help smooth your workflow.

VST Settings

Start in the VST Settings section under Edit / Preferences. Make sure all folders containing VST plug-ins are included in the Scan Paths section. Most VST2 plug-ins ask where you want them installed (I created a folder at C:\Program Files\VSTplugins, and install my VST2 plug-ins there for convenience), but some companies place them in arbitrary folders, often C:\Program Files\Steinberg\VSTplugins. These plug-ins may seem to be missing until you include their host folders. VST3 plug-ins have a nominal standard home, which is at C:\Program Files\Common Files\VST3.

For scan options, 'Automatic Background Scan' is unobtrusive, doesn't add any time to program startup, and ensures Cakewalk won't miss a new plug-in you've installed — even if you install it while Cakewalk is open. If you don't want to see the scanning notification that starts when you open the program, choose Manual Scan. However, this won't make the program load any faster, and you'll have to remember to scan after installing or removing a plug-in.

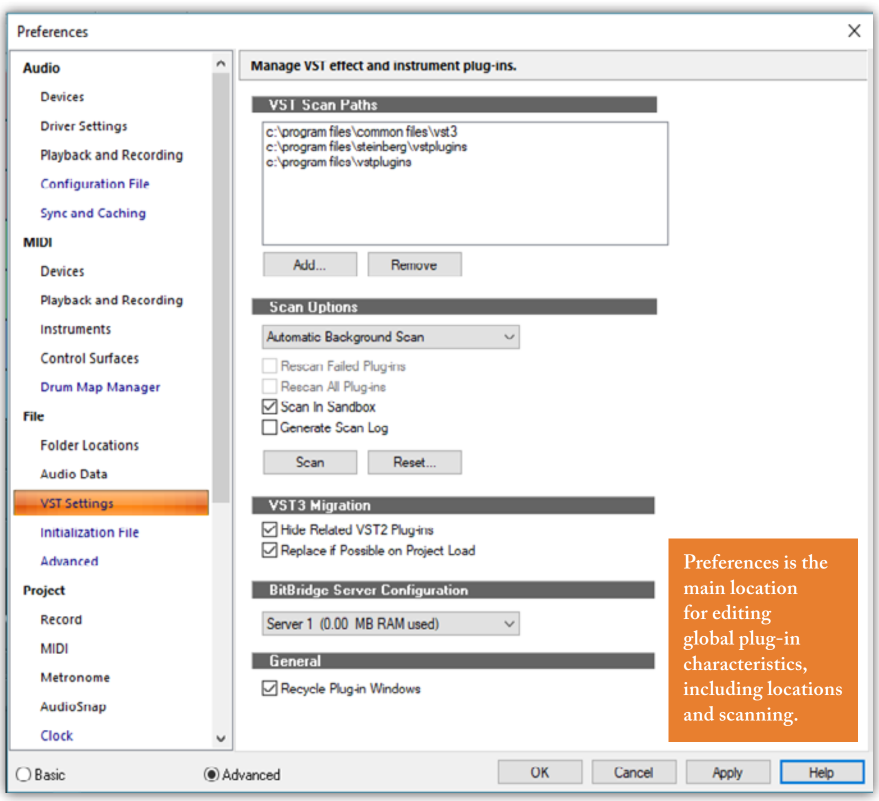
The 'Scan in Sandbox' option slows scanning somewhat, but isolates the scanning process. If there's a problem with a plug-in, Cakewalk carries on regardless of the scan's status. This means you can correct any issues later, rather than hit a speed bump in the creative process.

There's also an effective solution for when your plug-ins seem hopelessly confused, as can happen after updates or Windows issues. First, tick 'Scan in Sandbox', then click 'Reset', acknowledge that you understand any plug-in configurations will return to their defaults, and sit back while everything re-scans. This solves most of the "my plug-ins are acting weird!" issues.

As to VST3 Migration, in an ideal world you would just tick 'Hide Related VST2 Plug-Ins' and 'Replace If Possible on Project Load', and be done with it. However, there are differences between VST2 and VST3. For example, some VST2 amp simulators can accept Program Change commands to switch presets, but their VST3 versions cannot. Although VST3 was introduced in 2006, companies were slow to adopt it because VST 2.4 worked fine — and while the VST3 world has become quite stable, there are still instances where a VST2 version is preferable. I ticked the VST3 Migration options, but if a VST2 version works better overall, I rename the VST3's .dll suffix to .dxx. That way, Cakewalk loads the VST2 version instead.

The BitBridge server configuration options are explained well in the Help section, but hopefully you've done everything possible to eliminate 32-bit plug-ins, as this promotes stability in x64 systems. (True story: at a conference, the developer of a DAW that dropped 32-bit plug-in support was asked by an attendee about the best way to run a 32-bit plug-in within the program. After a pause, the developer said "Ask the manufacturer to create a 64-bit version." That may sound snarky, but it makes the point.)

Finally, the 'Recycle Plug-ins Window' option caused confusion initially, because engaging it meant that opening a new plug-in closed any already open plug-ins. To defeat this behaviour, untick the box. I prefer to leave it ticked — you can still open a new plug-in window while leaving the others untouched by holding Ctrl when opening a plug-in.



Preferences is the main location for editing global plug-in characteristics, including locations and scanning.

Plug-in Manager

The stand-alone Plug-in Manager (PIM) application is being phased out as more user-friendly functionality is added to the built-in plug-in Browser. According to BandLab's Noel Borthwick, the only reason the Plug-in Manager (which isn't being maintained or updated) is still being included is because you can use it to create plug-in layouts and exclude plug-ins. It's also the only way you can create nested folders, or set plug-in options simultaneously for all plug-ins in a selected folder.

Excluding plug-ins is convenient to prevent Cakewalk from scanning a plug-in that's keyed to a different program, but here's a cute trick: Cakewalk ships with four plug-ins (VX-64, PX-64, TL-64, and Boost 11) excluded. Choose Utilities / Plug-In Manager, click on VST Audio Effects (VST), then tick Show Excluded. Highlight them, »

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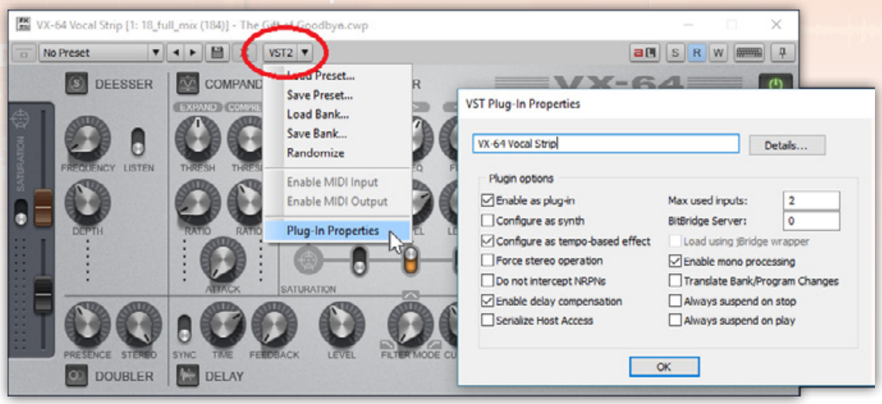


The VST drop-down menu's Plug-in Properties page reveals many editable parameters. In the background, VX-64 has been removed from its default position in the Excluded Plug-ins list, so it's now available for use.

» click on Enable Plug-In, and if they weren't already installed because you had Sonar on your computer, you'll have four new plug-ins. VX-64 and PX-64 are particularly wonderful.

Two of the PIM's formerly most useful functions are available elsewhere. Scan VST Plug-ins can be done from Preferences (although it's not done in the background, so you have to wait for it to complete), as can setting the Scan Options. To access a VST plug-in's properties (the DX plug-ins don't have an equivalent function), click on the VST plug-in's drop-down menu.

And while we're covering the drop-down menu, don't overlook the ability to save and load FXP presets, or the cool Randomize parameter. It often produces junk, but every now and then, randomising plug-in parameters can spark creative thinking. Also note that under VST Plug-in Properties, you can change the plug-in name that's displayed in the FX Rack. This is convenient when a company uses their name as a prefix to all their effects, or to shorten names so they fit easily within the FX Rack label. Clicking the Details button shows the plug-in's path so you know the folder from which it came, as well as the plug-in's unique ID.

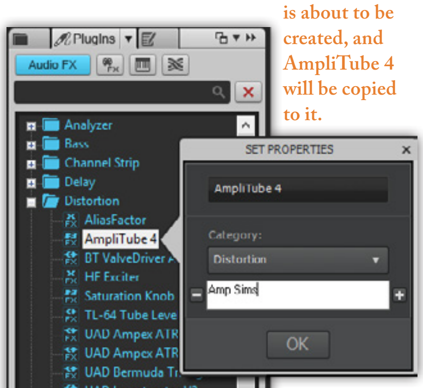
One final point about the PIM: layouts are saved as PGL files that you can back up, or transfer to other computers running Cakewalk. If you open it as a text file, you can change names (and if you've changed the name, see the original one), as well as check on the CLSID identifier. The Browser saves its layout and category information to a file called Library.db, located at C:\Users\[your name] \AppData\Roaming\Cakewalk\Library, which you can also transfer among computers and back up (which you need to do if you ever have to reinstall Cakewalk, and want to recreate your various customisations).

Custom Categories

You can organise plug-ins within the Browser by selecting 'Sort by Category' from the Plug-ins drop-down menu. (Not surprisingly, 'Sort by Type' and 'Sort by Manufacturer' don't allow reorganisation.)

You can move or copy a plug-in to a different category, so the same plug-in can show up in multiple categories, or in new categories. For example, IK's AmpliTube 4 defaults to the Distortion category. I consider that category more suited to saturation, analogue tape effects, and the like, so I'd rather have it live in an Amp Sims category.

To move an effect, right-click on it, choose a new category in the Set Properties box, and click OK. To copy the effect to a new category, right-click on the effect you want to copy and click on the plus (+) sign in the Set Properties box. Click in the field that just appeared, and type a name for the new category (like Amp Sims). Now when you click OK, the effect will be both in its original category and the new category you created. If you right-click on the clip in the Browser, you can see the categories in which it resides.

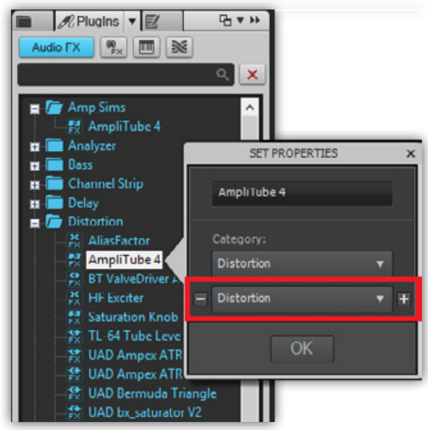


A new category is about to be created, and AmpliTube 4 will be copied to it.

If needed, you can Ctrl-click on multiple plug-ins and assign them to a new category all at once. However, any folders that contain plug-ins you want to assign to a new category must be open. If you close or open any folder, all your selections will be deselected.

If you don't want the plug-in residing in its original location, right-click on the plug-in, and you'll see the locations listed where the plug-in appears. In a field that's preceded by a minus (-), use the drop-down menu to select the category from which you want the plug-in removed. Click on OK, and the plug-in will disappear from that category.

If a plug-in doesn't exist in at least two places, then you can't delete it from its original location. This is a safety feature



Selecting the category in a field with a minus (-) sign, then clicking OK, removes the plug-in from that category.

so you can't remove a plug-in and not get it back. To make a plug-in go away, you still need to use the PIM to exclude the plug-in manually.

There are some useful keyboard shortcuts for the Browser. To open the Audio FX, MIDI FX, Virtual Instruments or ReWire tabs, click Ctrl+A, Ctrl+M, Ctrl+I or Ctrl+R, respectively. You can also use a variation on this to close all open folders. For example, if you have a lot of folders open in Audio Effects because you were moving multiple effects into a new category, select a tab that's not Audio Effects, then type Ctrl+A. When you return to the Audio Effects tabs, all the folders will be closed.

Lastly, any Browser categorisation changes apply only with 'Sort by Category' selected. However, you can convert this to a layout. Select 'Manage Layouts' from the Plug-ins drop-down menu, and after the PIM opens up, the layout will reflect your customisations. You can then save it as you would any other plug-in layout from the PIM.

Q Why does auto-release sound so bad with parallel compression?

When doing what I thought were some last listening checks for a mix, I heard some annoying rapid volume changes in the piano track. I tracked it down to the Waves V-Comp plug-in that I was using for upward compression on the piano bus. The release time was set to 'auto'. On long sustained chords, the gain-reduction needle was wiggling rapidly back and forth as the chord decayed. Setting the release instead to a fixed time of 1.5s solved the problem.

I usually don't use the auto setting, but for some reason had tried it and forgot about it. The Waves V-Comp manual just says: "Auto adjusts the release time according to the input signal to achieve a 'nominal' release." Could you please explain what this kind of auto-release setting actually does? What is it trying to accomplish?

***SOS* Forum post**

***SOS* Technical Editor Hugh Robjohns replies:** The exact implementation varies between different manufacturers' designs, but in its simplest form the auto-release (or auto-recovery) setting switches between two different release time constants — one is very fast, and the other very slow — depending on the amount of gain reduction being applied at the time. The aim is to give the best of both worlds: a fast recovery for brief transients (to maintain a sense of loudness), and a slow recovery for more gentle level-smoothing.

So when a large amount of gain reduction has been introduced — something that would typically be triggered by a brief transient peak — the recovery time will be very fast, and this rapid change of level up and down maintains the impression of a loud dynamic sound, while still being dynamically controlled. However, for smaller amounts of gain reduction, such as for much more modest changes of source level, the recovery time is much slower, so that there is no audible change of noise floor (an effect often called noise-pumping).

With auto-recovery selected, if you look at the gain-reduction meter when the compressor is hit by a loud transient, you'll see it swing over to the appropriate amount of gain reduction, then fall back rapidly to about 4-6 dB of gain-reduction, and at that point it will slow right down as the rest of the attenuation gently ebbs away.

In normal downward compression applications, this auto-recovery system usually works brilliantly well. In fact, I usually leave the compressor set to auto unless I require a particular time-release setting for a specific effect. However, when configured for parallel compression (which I'm almost certain is what you mean here by 'upward compression'), you're deliberately operating with large amounts of gain reduction all the time. Consequently, the auto mode will be operating with a very fast recovery time and, just as you noticed, that's not what you want in your situation! Hence the need to manually select a slow recovery for your specific application.

Many compressors , including the Waves V-Comp pictured here, offer a very useful auto-release or auto-recovery facility, which allows the compressor to respond differently to transients and to sustained sounds that exceed the threshold. But that's rarely a good option for compressors used in parallel with the dry source signal.

Q What is making my 'kicksnare' mic distort?

A big thanks for Mike Senior's very useful Session Notes article in the January 2019 issue (https://sosm.ag/session-notes-0119) about achieving a great drum sound in small spaces. I copied the complete mic setup from this article and the recordings I made in my isolation booth sounded wonderful! However, I couldn't use the Neumann TLM103 as my 'kicksnare' mic, as it couldn't handle the sound level produced by the drummer. Even with zero gain at the preamp this mic produced a distorted signal. What was I doing wrong?

***SOS* Forum post**

Hugh Robjohns replies: Drums certainly produce very big transients, so close-miking them is often a challenge for a lot of mics. However, the Neumann TLM103 is rated to handle 138dB SPL with just 0.5 percent distortion, which is pretty darn impressive — so while it is *possible* that the mic was being overloaded, I think it unlikely in this situation. A more likely culprit is that your mic preamp is overloading because, at 21mV/Pa, the TLM103 is a pretty sensitive mic. In a close-mic situation on a loud kit that means the mic is going to generate a signal level pretty close to 0dBu — in other words, line level — and many mic preamps don't expect that, and cannot cope with it.

In situations like this, an easy solution (although theoretically not the best, from a noise perspective) is to switch in the pad in the microphone. However, the TLM103 doesn't have an in-built pad. The next-best option is to engage the pad on the mic preamp. You haven't said what preamp you're using, but I'm guessing you'd already have tried that if your preamp had one. So the final option is to invest in an in-line XLR mic attenuator, and connect that between the mic and preamp. In-line attenuators come in a variety of values, but -20dB is normally enough. Shure, Audio-Technica and various other manufacturers make switchable attenuators, typically with three different settings, and these are very versatile, but fixed-value attenuators are available for far less. If using such an attenuator makes distortion go away, you'll know it was the preamp that was overloading. While I strongly suspect this was the case, it's not the only possibility.

If the distortion continues in spite of

When recording drums in small spaces, a sensitive mic such as the Neumann TLM103 can put out very high levels that not all mic preamps can cope with. If your mic and preamp lack a pad facility, an in-line XLR attenuator can be used to avoid overloading the preamp.

the level into the preamp now being much lower, then it must be the mic that was overloading. Now, while it might have been because the transient peak was over 140dB SPL, I suspect you'd know if the drums were that loud because your ears would be rather sore, which you haven't mentioned! And if the problem was that the mic was broken or poorly in some way, I suspect you'd find it distorted in other situations too, which you also haven't mentioned.

There is one other possibility to consider: your preamp has a feeble phantom power supply — one which is unable to make available sufficient current or voltage to the microphone. Under normal circumstances I'd say that wasn't very likely either, because the TLM103 has a very modest phantom current requirement (3mA). But if you're running a lot of current-thirsty mics off the same preamp, as one often is when recording drums, or if you have one or more faulty mic cables in the rig, then the phantom supply could be being dragged down far enough to cause problems.

To check if this is the reason, you could substitute the preamp with a different one, or use an external phantom supply. But perhaps a more practical route if you have a multimeter would be to measure the phantom voltages between pins 1-3 and 1-2 on the mic XLR. If you unplug the TLM103 (but leave all the others mics connected and powered up) the voltages at the preamp's XLR should be identical and between 44 and 52 Volts (the spec calls for 48 ±4 Volts). If you then reconnect the TLM103 and check again inside the mic cable XLR at the mic end, it should read about 27V (the reduction being due to the current drawn by the mic). If you get significantly different figures then the phantom supply isn't up to scratch. But, as I say above, I strongly suspect that your preamp was just being overloaded, and that investing in some in-line mic attenuators would be a good idea.

***SOS* contributor Mike Senior adds:** Thank you for your kind words about that article! It was an unusual session, but it turned out well, and I'm glad it has helped you improve your own recordings. I won't attempt to rehash any of the great information that Hugh's already provided. He's basically said everything I'd have said, only better, as usual... The only thing I'll add is to confirm that I did indeed encounter no problems at all with distortion on the kicksnare mic on this specific session, and we were recording through an Audient ASP880 mic preamp, I seem to remember, which as Hugh's review (in *SOS* August 2014: www.soundonsound.com/reviews/audient-asp880) pointed out, has masses of headroom — more than enough to cope with the output of a TLM103 in a very noisy place!

Q What do reverb preset names actually mean?

I'm looking through my reverb plug-in's preset list, and I'm wondering: what do all those cryptic preset names mean? And I heard you should always send vocals to something with the word 'Plate' in it. Is that right?

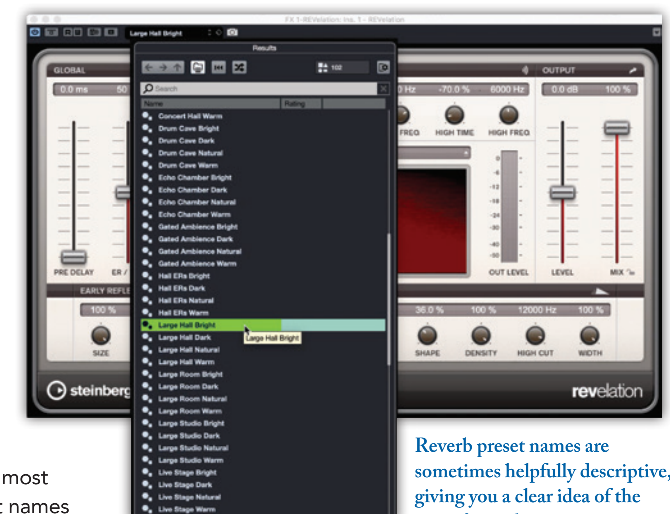
James Totterdale
via email

Mike Senior replies: Well, the names of presets are only useful if they give you an idea of what to expect sonically, and that's a bit hit-and-miss in my experience. I'm most sceptical about preset names with instrument suggestions in them, particularly if that's unqualified by any further information. Reverb use depends so much on the stylistic expectations and the nature of the recordings themselves (particularly what kind of spill, if any, is baked into the recordings), so a simple 'Snare' preset would rarely be of interest to me in practice. On the other hand, 'Epic Snare Boosh', 'Tight Snare Ambience', or 'Icy Rimshot Tail' might well entice my mouse click under appropriate circumstances.

It's also quite common for a preset that's ostensibly named for one use to work very well for something completely different, or to provide a great base for editing into another form. So, in short, take those kinds of preset names with a huge pinch of salt!

What I do find quite useful, though, are preset names which indicate something about the technical nature of the reverb that's being emulated. So if it says something like 'Church', 'Plate', 'Forest', 'Spring', 'Chamber', 'EMT250' or 'AMS Gate', I'll immediately have some idea of what to expect based on my past experiences with those acoustic environments or pieces of vintage technology. In that vein, you might find my 'Use Reverb Like A Pro' article from *SOS* August 2008 (https://sosm.ag/reverb-like-pro-pt2) useful, as it mentions a number of classic reverb devices and why they're popular.

As for always sending vocals to a plate reverb, you'll be unsurprised that I'm wary of such categorical assertions at mixdown. Having said that, though, plates are particularly good for adding warmth and sustain to vocals, and because they don't



Reverb preset names are sometimes helpfully descriptive, giving you a clear idea of the sort of sound you can expect, as shown here... though sadly that's not always the case!

have a natural acoustic signature they tend to distance a vocal less from the listener than a chamber or hall or room or whatever, so it's not a huge surprise that they get used a lot on vocals. I frequently use them on things like acoustic guitars and pianos too, where I want a lot of sustain, often with a high-pass filter to focus that sustain on the higher strings that naturally sustain less on the instrument.

Photo courtesy Twentieth Century Fox

INSIDE TRACK

Secrets Of The Mix Engineers: Justin Shirley-Smith, Joshua J Macrae & Kris Fredriksson

The highlight of Queen biopic *Bohemian Rhapsody* is a meticulous recreation of their most famous live show — using a fresh mix of the original audio recordings.

PAUL TINGEN

By the time you read this, *Bohemian Rhapsody* may well have grossed more than one billion dollars worldwide. It's already the highest-grossing musical biopic of all time, by a factor of five, and the highest-grossing drama film without action or fantasy scenes. Rami Malek has been widely praised for his astonishingly believable portrayal of Freddie Mercury, and at the time of writing, the movie had just won four of its five Oscar nominations. Queen manager Jim Beach produced the movie, and guitarist Brian May and drummer Roger Taylor were involved in almost all aspects of its making — which took no less than eight years.

The music of Queen, of course, is central to the film's success, and the band's original recordings were used almost throughout. The soundtrack has been honoured with a BAFTA Award for Best Sound to John Casali (production sound mixer), Tim Cavagin (re-recording mixer), Nina Hartstone (supervising dialogue editor), Paul Massey (re-recording mixer)

Left-right: Gwilym Lee (Brian May), Ben Hardy (Roger Taylor), Rami Malek (Freddie Mercury) and Joe Mazzello (John Deacon) recreate Queen's legendary Live Aid concert in *Bohemian Rhapsody*.

and John Warhurst (supervising sound and music editor). The movie also won Academy Awards for Best Sound Mixing and Best Sound Editing, and the quintet clearly performed miracles with the dialogue and in putting the final movie sound mix together. The original Queen music, however, was supplied by the band's own sound team, consisting of Justin Shirley-Smith, Joshua J Macrae and Kris Fredriksson. May and Taylor are credited as Executive Music Producers, while Macrae, Fredriksson and Shirley-Smith are credited as Queen Music Co-Producers.

Macrae is Roger Taylor's regular engineer and manager of the drummer's The Priory Studio, while Shirley-Smith and Fredriksson are Brian May's engineers and run the guitarist's private Allerton Hill Studio — both studios are located in Surrey. The three engineers enjoy mixing and co-production credits on almost all recordings that Queen, Taylor and/or May have been involved in over the last few decades, including 14 tracks on *Bohemian Rhapsody: The Original Soundtrack* — the other eight are original album tracks. The soundtrack album itself has reached number one in many countries, and got to number three in the UK and the US, and it still remains in higher regions of the charts four months after its release.

Needle-drops & More

The involvement of Macrae, Fredriksson and Shirley-Smith began with a visit from movie sound supervisor John Warhurst. Macrae: "When John came down to see us, the movie-makers had in mind to re-record all the Queen songs that were to be used in the movie. I think John had done a number of films where there wasn't a positive connection with the musical artist involved, and where they did not have access to the masters, so they were fully prepared to go down the route of re-recording every note and vocal. In fact, they had already hired an arranger who had a good understanding of Queen's music, and were discussing who to hire to replay the guitar parts and so on. When John came in with the script, he asked: 'What do you have for us?', clearly not expecting much."

Joshua J Macrae is full-time engineer at Roger Taylor's studio in Surrey.

Shirley-Smith: "Some of what they wanted were what they called 'needle-drops', which is simply the original studio recording being played in the movie. The other ones were live performances. For the needle-drops his first question was: 'Do you have the original multitrack masters?' The answer to that was yes, and better still, we also already had pre-mixed stems from some of these masters, because we had reverse-engineered these original mixes in the past for other projects. To his surprise and delight, we then told him that we also have the multitracks of many live recordings, which meant that they also didn't have to re-record these.

"The live recordings we have are of pretty good quality, which was important, because the movie-makers were very particular about the quality needed. The recordings we have, for example of Hammersmith Odeon and the Rainbow Theatre in the 1970s, are really beautiful. As we spoke, John became happier and happier, realising how much original Queen material could be used. He then asked us whether we could make stems of tracks where there weren't any, and again we said we could. The film-makers could not believe their luck!"

Moving Mountain

To understand how and why Shirley-Smith, Macrae and Fredriksson were so well prepared when Warhurst came knocking on their doors, we need to backtrack a little. One essential bit of background is the fact that the three have worked for Taylor and May for a long time. "I have been with Roger since 1987," explains Macrae, "I played drums in a band with him, the Cross, until 1991, and then started engineering for him. We moved to the premises where we are now in 2004. The studio is private, with occasional use by friends. In the old studio we had a 56-channel Amek Mozart desk, but we are fully in the box here."

"I used to play drums and still play guitar and a bit of piano," continues Fredriksson. "I studied music and engineering, and I began working for Brian in 2001, having worked at Mountain Studios in Montreux before that. There's so much going on within the Queen world that it's a full-time job for all three of us. We do archiving, there are projects like the *Bohemian Rhapsody* film, Brian put out a new song on New Year's Day [*'New Horizons'*], he does guest appearances with other artists, and there's the Queen catalogue to look after, which involves remastering, remixing, reissues and so on. It's several jobs into one, with no time for anything else. Mad but good!"

"I had piano, violin and 'cello lessons as a child," adds Shirley-Smith, "and then played guitar in a band and wrote songs. I applied for some jobs at recording studios in London, but got a job at Queen's Mountain Studios, at age 18, in 1984. I have been part of the Queen family for a long time now! Mountain Studios was a commercial studio, so I engineered many

Justin Shirley-Smith is one of two staff engineers at Brian May's Allerton Hill Studio.

other artists there as well. David Bowie, who lived close by, was always in and out, and Chris Rea worked there, as did AC/DC, and so on, and there often was stuff happening related to the Montreux Jazz Festival. Brian asked me to come to the UK in 1991, even before Freddie died, to finish his solo album [Back To The Light, *1992*]. We had a tiny desk and two Sony 24-track machines, and a massive billiard table which was pushed into a corner. After that Brian got a 60-channel Neve VR desk, which is still there." Today, Mountain Studios is a museum dedicated to Queen, with one poignant feature being the room where Mercury laid down his last vocals, in May 1991, recorded by Shirley-Smith and David Richards.

For Posterity

As Fredriksson explains, the trio's long service means that they have a deep knowledge of Queen's recording history. "All three of us, along with Queen's archivist Greg Brooks, are involved in the tape libraries, which is separate from our job of running the studios. Sometimes we do transfers in one of the studios, sometimes in other facilities, while the vaults themselves are in lots of different locations, depending on the formats, and for obvious reasons — we don't want to run the risk of losing everything! The aim is to preserve every single Queen recording, but because there's so much, what we archive at any given time is always project-driven. Years ago, we started with the important stuff, the stereo quarter-inch masters of the albums, and copied them to other analogue formats, and then later we copied them to digital when we had that, and in 2002, when we got 192kHz, we also started copying the multitrack masters."

Shirley-Smith: "The funny thing is that the older the technology, the more robust it is, so in archiving we also focused on what we call eradication projects, for example U-Matic and DASH recordings. These are the most vulnerable, and also, we were not sure for how much longer we would have machines to play these things on. Analogue tapes, and in particular two-inch tapes, are the most robust, although we do have to bake them. We have an incubator that fits 12 reels of two-inch. We bake analogue tapes as a matter of course, four days at 50 degrees C, and these tapes always play back without any problems on our Studer A827 24-track, which is one of the last ever made. I think it's called a Gold Edition."

Fredriksson: "The later Queen albums were recorded on two 24-tracks, so we're dealing with 48 tracks of audio. If you also copy all the out-take reels, you end up with a lot of material! Over the course of the '00s there was an initiative to create 5.1 mixes for every Queen song that has a promo video, and we also delivered Queen songs for several movies — for example, *Blades Of Glory* [*2007*] used the song 'Flash'. In all those cases we had to create digital stems of the songs, and this meant that we had to reverse-engineer the mixes. This involved first of all carefully going through everything to make sure that we used the correct takes and edits."

Shirley-Smith: "The guitar, and especially the vocals, might take up six or seven tracks, and sometimes the comps were done on the desk, so there wasn't a single track on the multitrack where you could say: 'This is the lead vocal comp.' In some of the cases where there was such a track, we would unpick it, creating the same comp again from earlier generations of tape, because on some songs they had been forced to bounce from one multitrack to another. So we'd go back one or two generations, and once we had the correct performances, re-comp them, and often also apply CEDAR audio restoration."

Matching Mixes

Reverse-engineering the mixes didn't only involve retracing the often very complex overdubbing and editing steps taken by Roy Thomas Baker — and other Queen producers such as Reinhold Mack and David Richards — before compiling

Recording The Fox Theme

Kris Fredriksson: "The band had been to see a test screening, and when you go to see a Fox movie, it starts with the Fox theme tune. Brian had the brilliant idea of redoing that, and this made for a great start to the film, because it immediately makes clear to the audience that Queen are involved. It sets the film up beautifully, and also gives the soundtrack album an identity. We ended up having 66 tracks of Brian's guitar on that track, plus Roger's percussion. We recorded far more guitars, but only used these 66. I still find it fascinating after all these years to watch Brian's process. He had actually worked the entire arrangement out in his head, and never played it on the guitar until he came into the studio, when he said, 'What key is it in?' He then played all these guitar parts in a short space of time. It just pours out of him."

Justin Shirley-Smith: "It is great fun doing those guitar harmony stacks, because he does it so quickly. They are done in short bursts, so obviously you have to have many tracks ready to record. Brian's signal chain is his guitar going into a treble booster pedal and then a Vox AC30 amp, and we record that with a Sennheiser 421 at the front, and these days very often a Shure SM57 at the back. Both mics go through the Neve VR desk, and are recorded directly into Pro Tools, on separate tracks. If you flip the phase on the 57 track and fade it into the 421 track, it gives a lot of low end, that warmth and body. It allows you to adjust the sound quite dramatically in the mix."

everything in a tidy Pro Tools session and restoring the audio. Macrae, Fredriksson and Shirley-Smith also had to remix these sessions, making their new in-the-box mixes sound exactly the same as the original analogue mixes.

Fredriksson: "Queen obviously put a lot of time into writing the songs, and then they arranged and rehearsed them before they went into the studio, after which they would record and overdub in incredible detail. Finally, there was the mixing process, which was a whole other performance in itself! They were all hands-on on the desk, and used every single trick in the book, vari-speeding things and just generally messing around with things in the mix. If you compare the naked tracks on the multitrack to the final mixes there often is a phenomenal difference. Plus a lot of these mixes were done in sections and then edited together."

Shirley-Smith: "In approximating their mixes, we had to look at where the song was mixed, what board was used, what outboard they had used, and so on, and this would give us a good starting point. We have the original stereo mix in the session and we then A/B that against the mix we are doing. It's a very painstaking process to achieve exactly the same mix, and then to print that in stem format. With later sessions there would be recall notes, but in the '70s and early '80s people were not doing that, although there would sometimes be an occasional EQ knob drawing on the back of some track sheets to give a bit of a clue."

Macrae: "Welcome to our world! Occasionally we would ask Brian or Roger if they remembered what went on, but of course, we've worked with them for such a long time that some of us were there at the original mixes. Plus, after all these years we know what Roger and Brian like, so we don't go down cul de sacs. In approximating these mixes many of the UAD plug-ins were very helpful. They have great channel emulation plug-ins and things like that, and once you find the right emulations of the gear used at the time, you can home in on the sound you're after much quicker than if you are using just a random plug-in."

Shirley-Smith: "I hate having to do things twice, so we were very systematic. We played every tape we transferred to WAV from beginning to end, including any blank bits and tones, not just the part that we needed, and logged what we had in the database, so we never have to go back to the original tapes again. For the same reason, after we reverse-engineered a mix, we would also print stems. This also meant that we had stems for many tracks. These pre-mixed stems can be used for any movie project, and of course the makers of Bohemian Rhapsody were very happy with that. With this head start Paul Massey was then able to add his magic in the dubbing theatre."

Kris Fredriksson, Brian May's other staff engineer, at Allerton Hill.

Get The Band Back Together

'Reverse-engineering' past studio and live recordings was only a part of Fredriksson, Shirley-Smith and Macrae's work on the *Bohemian Rhapsody* soundtrack. The trio also mixed Queen's famous Live Aid concert from scratch, and did some new recordings, notably the opener, '20th Century Fox Fanfare' (see box). Another three songs on the soundtrack also contain newly recorded elements: 'Doing All Right', 'Don't Stop Me Now' and 'We Will Rock You'.

The latter two are among Queen's greatest hits, but 'Doing All Right' was originally written and performed by the group Smile, whose members were May, Taylor and bassist/singer Tim Staffell. Fredriksson: "'Doing All Right' is pretty much a brand-new recording. There's a small section in the middle that comes from an unreleased recording from Smile from 1969 — they had a recording contract, with Mercury Records! They are seen performing that song in the movie, and they wanted to re-record the song, so we did a session with Tim in Abbey Road, where he sang and played the bass. All three members of the original band sing a verse, so they were back together after 50 years, which must be record-breaking."

Shirley-Smith: "On 'Don't Stop Me Now', when we did the remasters in 2011, we unearthed all the rough mixes and outtakes and found this rhythm guitar that had not been used for some reason. At the time we did a mix including that guitar, and then for the soundtrack album Brian overdubbed all the guitars from scratch the way he plays them today, which gives a whole different feeling to the song. The 'movie mix' of 'We Will Rock You' is steered by what happened in the movie. In the original script it starts off with Brian teaching them how to start the song, in the studio, and then it cuts to live performance. So that's how we did the song. But in the movie they then changed it: first there's the scene in the studio, then it cuts to live performance and then back to the studio. However, this would not make any sense if you listen only to the audio, so Roger and Brian decided to put the original edit on the soundtrack album."

History In The Making

On to the centrepiece of *Bohemian Rhapsody*: the band's legendary Live Aid concert. It is both the climax and the emotional resolution of the movie, and the most important segment of the soundtrack album, with four songs (out of a total of six performed). Queen's set left a huge impression on the audience at Wembley Stadium, despite the fact, remarks Fredriksson, that "Queen weren't on the original bill. They were added after the concert sold out, so it wasn't even a Queen crowd!"

The concert has never before appeared on an album, but the original stereo mixes »

Allerton Hill Studio is home to one of the last Studer A827 machines ever made.

» and footage were released as part of the Live Aid four-DVD set in 2004. Three years later, it appeared as a bonus feature on a special edition of the *Queen Rock Montreal* DVD, mixed by Shirley-Smith and Fredriksson from the original multitrack. For *Bohemian Rhapsody*, Shirley-Smith, Macrae, and Fredriksson decided to redo the entire mix. They could do so only courtesy of the courage and foresight of Jeff Griffin, BBC Radio 1 concert co-ordinator at Wembley Stadium in 1985, whose attitute was in stark contrast to the US crew, who deleted the recordings of the Philadelphia Live Aid concert.

Shirley-Smith: "Bob Geldof told Jeff that he was not allowed to multitrack record the concert, because some artists had said they would not appear if they were recorded. But Jeff told him, 'I'm sorry, this is such a historical event, I cannot not record this.' If it wasn't for Jeff, we would not have had a decent multitrack recording. However, you were never going to get a perfect recording in those circumstances, with just 24 tracks, and all these bands playing short sets, and having a very swift turnaround. They had designed a circular rotating stage divided in three, so each act could set up in their third of the circle, and then they literally just swivelled it. It was amazing, and an approach still used today. Although each band could use their own live setups, there were always going to be problems with the technical quality of the recordings.

"For example, there was a little distortion on Freddie's lead vocal. This was one of the reasons why the film team thought they might need a vocal double. They thought that they would need perfect sound to go with the high-definition pictures, and that the slight distortion on the vocals would not be good enough for a Hollywood movie. In fact, they did sessions in Abbey Road during which they recorded a singer mimicking every lead vocal track Freddie had done that they wanted to use in the movie. The reason for this was twofold: one was that they had to make the shifts believable between singing and spoken word, with Rami speaking, and for that it's much easier to work with the perfectly recorded, dry double vocals. Secondly, when you solo live vocals from any concert recording, they are not going to be pristine. The film crew commented on issues like, 'The snare drum is distorting in the vocals,' or 'There's a tiny crackle here.'

"In the end, the decision was made to work with the live recordings and improve them as much as possible. The only singing by the vocal double that was used are bits for which no recordings of Freddie exist, for example where he sings 'Happy Birthday' to himself at the piano. We are all very picky about technical quality, but you have to balance that against the emotional impact. You want the energy of the original performance. No-one will be sitting in the cinema thinking, 'Oh, my goodness, there's a tiny bit of distortion on the lead vocal.' You just go: 'This is exciting!' In fact, that little bit of distortion adds to the excitement. And also, that concert is so well-known, if you change anything in it, you're going to be nailed to the wall!"

The Live Aid concert scene in *Bohemian Rhapsody* is stunningly realistic, thanks to a combination of the original audio, state-of-the-art CGI, and the director, actors, choreographers, costume and set designers going all out to replicate even the tiniest detail from the 21-minute set. Fredriksson recalls:

The multitrack that shouldn't have existed: Queen's iconic Live Aid performance was captured on a single reel of 24-track two-inch tape.

Perfect Stems

Josh Macrae describes how he, Justin Shirley-Smith and Kris Fredriksson have developed a unique and very efficient way of creating stems, which allows them to print all stems at the same time. The issue they managed to resolve was how to make the compression act the same on the individual stem tracks as on the stereo bus, so that the sum of the stems sounds exactly the same as the stereo mix. Running off stems one by one through the mix bus will obviously not do that.

Macrae: "If you take elements out of the mix, and send them through the compressor on their own, the compressor is obviously going to behave differently than when the entire mix is going through it. What we do first of all is hide the mix prints of the stereo mixes with compression in the playlist, and create a new stereo mix print playlist, and a 'Mix Comp Key' track — and we print the stereo mix on there, but without the Impact compressor. That means that we have an uncompressed mix print on that track. Next up we copy the plug-ins on the submix track, in this case the Massenburg EQ and Impact compressor, across all stem bus tracks, so each of these tracks has exactly the same EQ and compression as was on the 'Sub=M=' track, and hence on the compressed stereo mix.

"During the stereo mix, all instrument tracks were routed to the 'Sub=M=' track, but we now re-route all instrument group tracks to the corresponding stem bus tracks, bypassing the 'Sub=M=' track entirely. Next up, we create a pre-fade send to a bus, let's say bus 5, on the 'Mix Com Key' track which has the uncompressed mix. The thing to bear in mind is that you're sending stereo to mono here, which will be hotter than the stereo. After some experiments, we found that we have to pull the send fader down by 2.7dB to make it come back at zero. (This value of 2.7dB only works from Pro Tools 11 and up due to a mix engine redesign). Next we connect the side-chain keys of all those Impact compressors on the stem buses to the same bus, in this example bus 5, so the stereo mix print without compression goes through them, and all stem bus compressors are behaving exactly the same way as they did with the entire stereo mix in! This allows us to print all the stems at the same time, once again with the compressors behaving the same way as when we did the first stereo mix. We then print all that on the stem print tracks, and if you have all the stem print faders set to zero, and play them back, it will sound completely identical to the original stereo mix with compression."

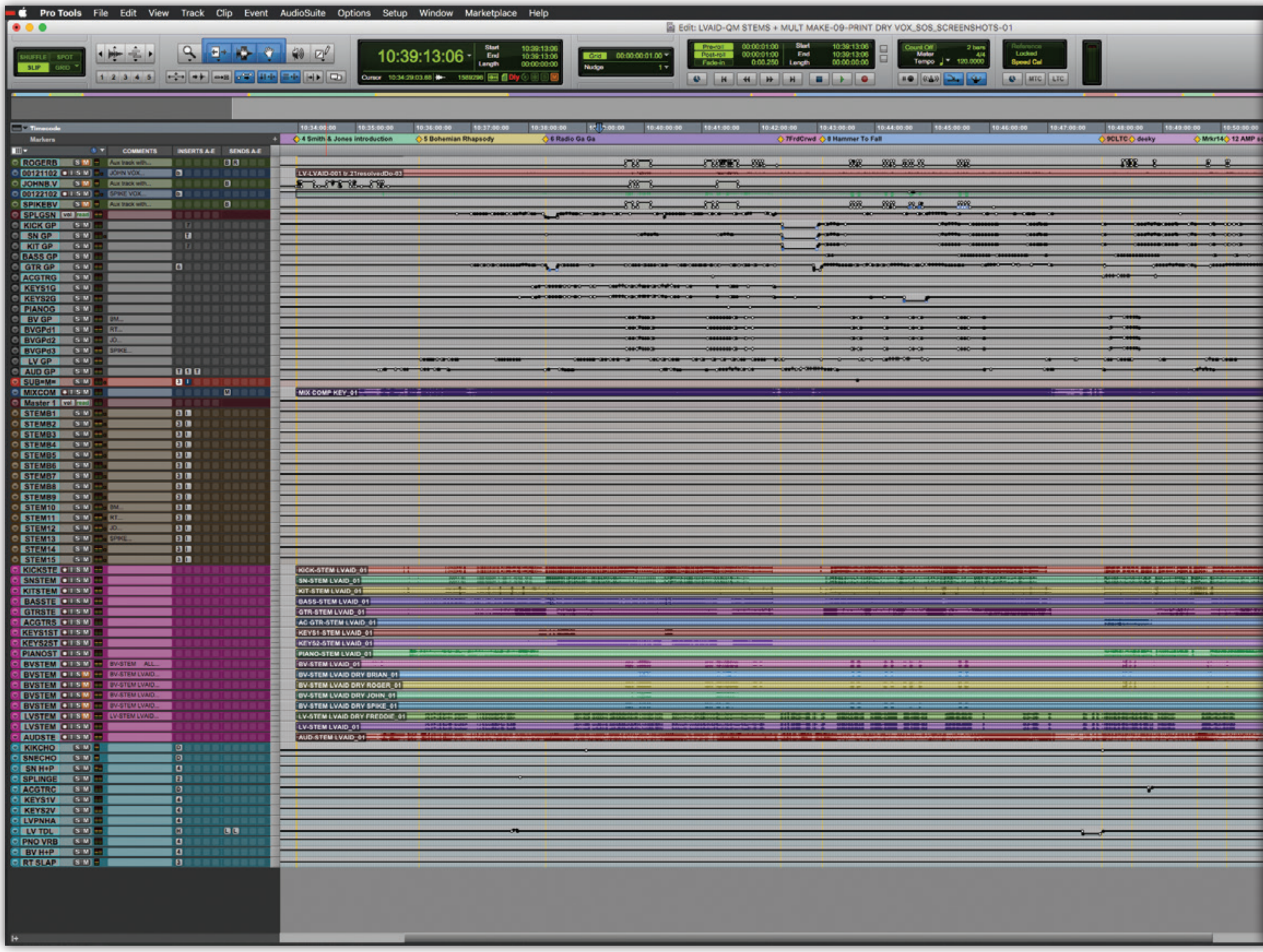
"I went along to see them rehearse early in the process, in some tiny old hall, where they were working with this amazing choreographer, learning every single move from the moment the band walked on stage, in real time. The attention to detail was phenomenal."

Tape To 'Tools

Shirley-Smith explains why the trio took the decision to mix from scratch and not to retrace their steps from 2007. "Kris and I had done stereo and 5.1 mixes at the time, on the Neve 88R desk in Sphere Studios in London. We felt we could do a better job today, because we're always developing new techniques. Because the 2007 mix was done on an analogue console, we could not really use that mix as a starting point. We had always had a copy in our archives, but starting again also gave us the opportunity to seek out the 24-track master in the BBC archives. We did our own transfer of the master, using Pro Tools HD IO. As it happened, there was very little difference between the two."

Macrae: "We actually loaded the entire master tape transfer into the Pro Tools session for the 2007 Live Aid mix, and deleted all the automation, with the exception of some EQs on the toms and overheads to take out some guitar feedback, some fader moves on the toms group, and on the aux tracks that each backing vocal has, just bringing up those parts where needed. This was an unusual situation, because I normally lay out each session manually from scratch. We don't use a template. Despite it being the same band, no performances are the same, and it doesn't take long to set things up. Even a vocal plug-in chain takes just a few clicks. We listen, and decide what the session needs. Sometimes you find so much vocal reverb in a hall that you don't need any."

Fredriksson: "We always load entire concerts into one Pro Tools session, »

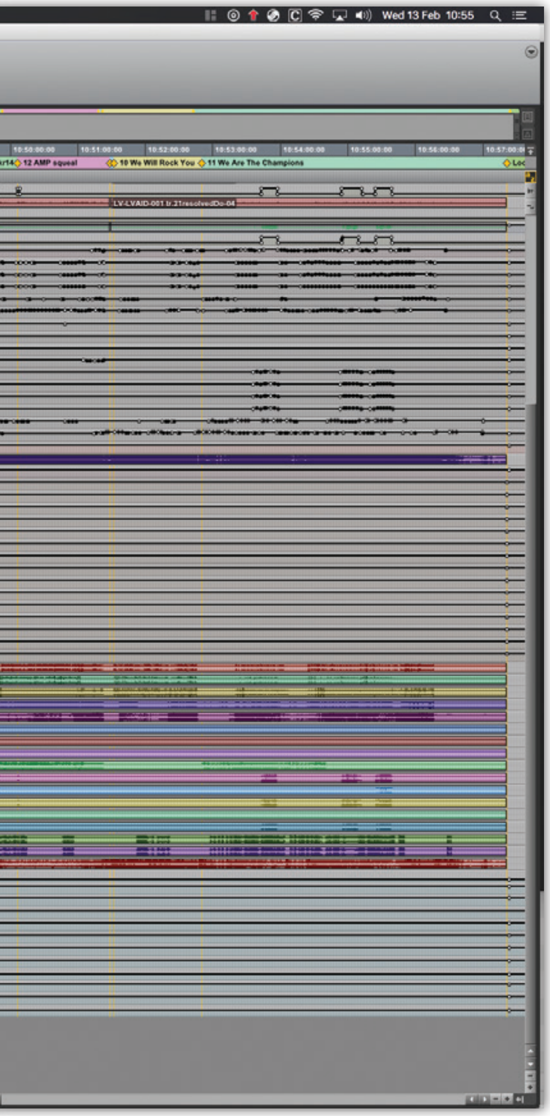


» and have done so since 2002, when we started putting out a lot of live concerts. A two-inch multitrack is 17 minutes at 30ips, but in this case they had run it at 15ips, so the one reel could contain the entire performance. In other cases they daisy-chained two machines. Because we always bring all the material into one session, and then consolidate that, all files end up as long as the entire concert. The session becomes like a very long analogue tape that could not exist in reality. There sometimes are significant overlaps on these live concert tapes when they used two machines; in those cases we listen through and work out which tape sounds better and use that."

No Fixes

While mixing the Live Aid concert, the trio were at pains to avoid getting nailed to the above-mentioned wall. "It was all about getting it to sound powerful," comments Macrae, "and keeping the ambience — it is a rock band playing in a stadium! The main thing was keeping it real and not over-processing things, and not changing any of the audio content, so there were no edits, no fixing anything the band played, and there was no tuning. The only things we fixed were technical issues, like feedback squeal. The classic moment is when Brian's guitar amp had a tube failure in 'Crazy Little Thing Called Love', resulting in a horrible whistle that doesn't have a fixed frequency, making it difficult to EQ it out. Instead, we used CEDAR Audio [*processors*] to get rid of that. In the 'Ay-Oh' section, with Freddie singing back and forth with the crowd, one of the roadies is tuning the bass guitar, and this sounded on the PA. Because you really want to hear just Freddie and the crowd, we did our best to remove the tuning noises as much as possible. But because this concert is so iconic, we could not fiddle with it. Any iffy notes were left! It has become known as the best live performance ever, so we also didn't want to fizz it up and give it a modern feel. It had to remain authentic and raw."

The only thing that could arguably be called 'fizzing it up' was a way of enhancing the ambience that ticked all Queen's boxes in its grandiosity. Shirley-Smith: "Paul Massey wanted more of the ambience of a band playing in a stadium. With many live recordings, they try to dry things up and lose the ambience. By contrast, we always try to keep the ambience, because it is the vibe, and we certainly did this when we mixed the Live Aid concert. But Paul wanted even more ambience. The film guys were even talking about hiring a venue just to put a PA in there and play the music back in it! But at



The entire Live Aid set was mixed from a single Pro Tools session. At the top are the original audio files transferred from the tape multitrack; the lower half of the session contains stems re-recorded into Pro Tools.

one point Queen + Adam Lambert were playing the O2 Arena in London, and were being filmed, so we had already arranged for extensive extra room mics to be put up in the hall. We went back to the Pro Tools sessions, printed new stems without our reverbs and delays, and played those through the PA at the O2 Arena, without the audience present, and recorded the room."

Fredriksson: "We used stems because you might want to adjust that mix in the room. You never know how it's going to sound. In fact, all the live music in the movie was played through that PA, and the ambience recorded. But we did not use any of that on the soundtrack album. The other thing that happened was that in 'We Will Rock You', the beat the audience in the movie is asked to do is stomp-stomp-clap. But for some reason, every audience in the world always goes clap-clap-nothing. Nobody had a clean recording of a huge crowd going stomp-stomp-clap. So Brian offered to get the O2 Arena audience to do this during the quiet moment in the set, when it's just him talking to the audience. That's how we got a recording of 20,000 people doing stomp-stomp-clap!"

The Ultimate Live Recording?

The Pro Tools mix session for Queen's Live Aid concert, complete with two-minute introduction by Mel Smith and Griff Rhys Jones, is exceptionally well organised. It consists of a total of 90 tracks, divided in six sections: original multitrack print tracks, 15 instrument group aux tracks, three master tracks, 15 aux stem buses, 17 stem print tracks, and 12 aux effect tracks. The multitrack print tracks in turn break down into a total of 22 audio tracks: two audience tracks, two toms with electronic drums tracks, two overheads tracks, hi-hat, kick, snare, bass, two synth tracks, piano, acoustic guitar, two electric guitar tracks, a track for Mercury's vocal mic when he was at the piano, a track for his handheld mic, and backing vocal tracks for May, Taylor, Deacon and Spike Edney, who also played keyboards. These audio tracks are sent to the associated aux group tracks below them: kick, snare, kit, bass, guitar, acoustic guitar, keys 1, keys 2, piano, four backing vocal groups, lead vocals and audience.

If the session had been purely for audio, this would have been it, but because Macrae, Fredriksson and Shirley-Smith also had to provide stems for the movie, they routed these 15 instrument group tracks to 15 corresponding stem bus tracks below the master track, and printed these on 17 stem audio tracks below that. Underneath these audio print tracks, at the bottom of the session, are 12 aux effect tracks, with the usual assortment of reverbs and delays. One of these is called 'Splinge', and is a particular delay effect beloved by the Queen members.

The three engineers used Brainworx's bx_console plug-in across all the original »

As well as reverb from a Lexicon 480, Freddie Mercury's live vocals were also treated with a ping-pong delay.

» audio tracks. "The bx_console is a Neve emulation channel strip," Macrae explains, "which we used for EQ and compression, just like you'd do with a normal desk channel. It helps glue everything together. The kick also has the Softube Summit Audio TLA-100A [*compressor*], to keep it in place and beef it up a bit, and a send to the 'KikCho' aux effect track, which adds some [*Avid*] D-Verb reverb, with a medium non-linear reverb.

"The snare, after the bx_console plug-in, also has the Softube Summit Audio Grand Channel, adding EQ, compression and saturation, and a send to the 'Snare Echo' audio track, which has another D-Verb, with a small non-linear reverb, plus it has a send to another snare reverb aux, with the Lexicon 480. The bass and the piano both have the UAD 1176 A [*compressor*], and the piano a send to 'Piano Verb', which again has a Lexicon 480. Both electric guitar tracks have sends to the 'Splinge' aux, which has the Waves SuperTap Delay, set to a kind of random arrangement, to give the feeling of the sound knocking about in an auditorium. Brian generally doesn't like reverb on his guitar, and prefers delays.

"After the bx_console plug-in, once again, both vocal mics have the UAD 1176, set to a high ratio, and the fastest attack and fastest release, catching the really loud, hard transients. Third in the chain is a UAD LA-2A, which is a softer and slower compressor, and more for general level and sound. After that there's the UAD Neve 1081 EQ for some crispiness, and the last plug-in on the inserts is the Tokyo Dawn Labs Nova, which is a frequency-conscious compressor that takes out the occasional nastiness, frequency specific. It acts here like a de-esser. Both vocal tracks also have sends to the 'LV P N Hall' aux, which has the Lexicon LX480, set to 'Medium Hall + Stage', and the 'LV TDL' aux, which has the Waves H-Delay."

The instrument group tracks have no plug-ins on them, other than the Softube Transient Shaper on the 'Snare Group' and the Waves REQ6 on the 'Guitar Group', and some treatments on the 'Audience Group' from the Avid EQ3 1-band and the TDR Nova. For the stereo mix, the instrument group tracks were sent to the 'Sum=M=' aux, which has a UAD Massenburg Designworks EQ and the Avid Impact compressor.

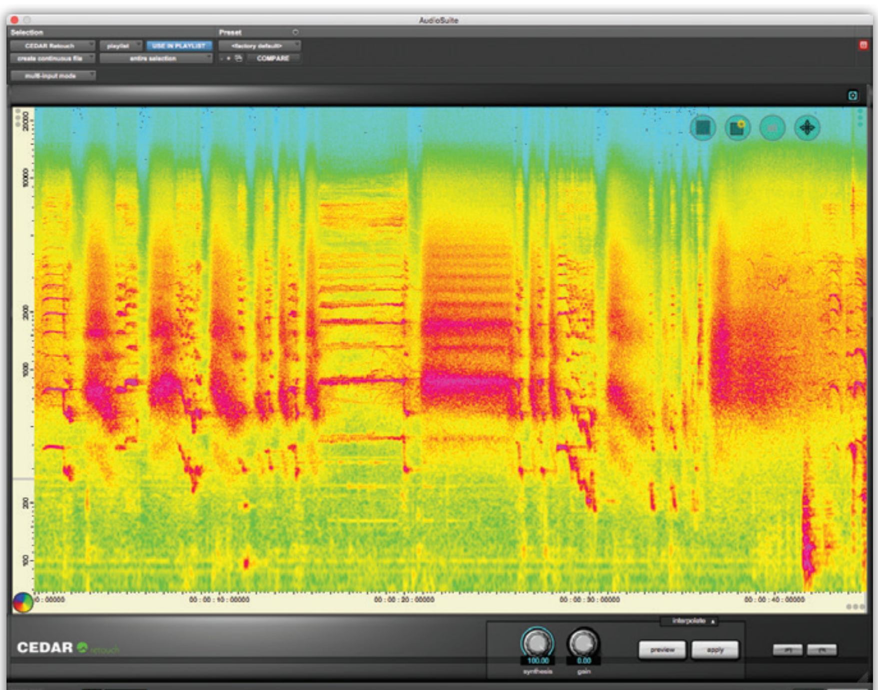
"All the volume automation for the audio is happening on the instrument group tracks." Shirley-Smith clarifies.

"Only corrective volume changes, for example dipping an 's' or a pop, or changes in volume to correct the fact that someone adjusted the mic amp halfway through, are done on the audio track itself," Fredriksson adds. "If you don't do that, these things will drive the compressors the wrong way. The Massenburg pulls out the PAL line frequency, which is a whine inherent in the PAL television format, at 15.625kHz. You get this ringing and it drives me absolutely mad, so we have to notch that out. The Impact is the bus compressor."

While mixing they listen to Genelec loudspeakers: 8260A SAM with 7271A SAM sub at May's studio and 8250A SAMs with 7270A sub at The Priory. Shirley-Smith: "We also have a pair of NS10s, and listen to computer speakers, headphones, car stereos, mono radio and so on. You have to also hear things in real-world situations. I will listen to things focused in the studio, and then also unfocused, while doing something else with the track playing in the background. If it plays in another room it works even better."

The Masters

The final stage, at least as far as the soundtrack album went, was mastering. Fredriksson: "We have been working with Bob Ludwig at Gateway Mastering for about 15 years now. We spent a year



CEDAR Audio's Retouch was used to remove problematic noises from several of the Live Aid tape tracks.

The work of Macrae, Shirley-Smih and Fredriksson was only part of a huge team effort on the audio front. Most of that team are shown in this group photo taken at Twickenham Studios: from left, William Miller, Tom Melling, Kris Fredriksson, Joshua J Macrae, Nina Hartstone, Paul Massey, Miranda Jones, Brian May, John Warhurst, Tim Cavagin, Louise Burton, Alistair Hawkins and Justin Shirley-Smith.

remastering the entire catalogue in 2011, just dealing with the stereo masters when the band moved to Universal Records. That work was done very meticulously, with us digging through archives all around the world finding tapes to make sure we had the best possible sources. All that was mastered by Bob, and these are the album tracks that appear on the soundtrack album. The newly mixed material was mastered by Adam Ayan, also at Gateway, and this blended in well.”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FOALS

Yannis Philippakis & Brett Shaw: Producing *Everything Not Saved Will Be Lost*

Foals' first foray into self-production yielded not one but two albums — but making them wasn't easy!

TOM DOYLE

Four albums into a 12-year career that had taken them from sweaty house-party gigs to sold-out arenas, Foals decided to have a rethink when it came to making their fifth record. Exhausted after the world tour to support 2015's *What Went Down*, and temporarily sick of making music altogether, the Oxford-formed band's frontman Yannis Philippakis couldn't even face a guitar for nine months.

Then, in late 2017, he rented a production room in 123 Studios in South London, close to his Peckham home. There he slowly began creating sketches for what would in fact become Foals' fifth and sixth albums, the two companion releases *Everything Not Saved Will Be Lost* (Part 1 out now, Part 2 to follow in the Autumn).

And So There Were Four

By the time production proper began, the long-time quintet had been reduced to a four-piece following the amicable departure of bassist Walter Gervers. Instead of seeking a replacement, Philippakis and his bandmates — guitarist Jimmy Smith, drummer Jack Bevan and keyboard-player Edwin Congreave — decided to carry on in this new configuration. In the studio, synth bass lines now became more of an option on certain tracks, while the bass guitar roles were filled by either Philippakis or Congreave.

"One aspect of Wally leaving was that it meant that these roles in the band became a bit more fluid," says Philippakis.

Photos: Kit Monteith

Foals singer Yannis Philippakis stepped into the production role on the band's new album *Everything Not Saved Will Be Lost*, Part 1.

"Having Edwin taking over bass duty in the live room and opening out the bass lines in certain tracks was great. It created a different energy in the room that I think filters into the album."

On previous Foals albums, the band had worked with producers Dave Sitek, Flood, Alan Moulder and James Ford, but they decided to self-produce *Everything Not Saved...* Yannis Philippakis took on the primary production role in the band, but admits that it took him a while to become comfortable with this. "I was quite reluctant, to be honest," he says. "It was more something that was driven by Jack in the band, cause he just thought that it was time for us to try and go it alone in a way. There'd been a feeling in the band that we'd made a number of records and we'd never really got to scratch that itch of what it would be like if we saw everything through to the finish line on our own.

"The desire with this one was to explore new textures, without somebody coming in and moulding or polishing something. We wanted it to be a freer expression of the band that we are. Giving ourselves time was the big one and allowing ourselves the space to make mistakes. 'Cause, y'know, sometimes by taking the wrong turn when you're recording, you can end up in the right place."

Starkness Rising

Alone in his production room at 123 Studios at the beginning of the making of *Everything Not Saved...*, Yannis Philippakis encountered some very basic problems, which led him to recruit the studio's Brett Shaw as engineer and co-producer. "In general, I'm a Luddite," laughs Philippakis. "Like, I've only just learned how to use Logic. Every time I'd turn stuff off in that writing room, I had to get Brett to come and help me anytime I'd restart.

"One of the reasons why I got this small writing room in Brett's studio was to learn how to use Logic, 'cause up until then, for the last decade I'd been recording stuff just straight onto a loop pedal that I couldn't extract the layers off, or actual cassettes, or into my phone. The problem with it is not only is it unusable in many ways, but also, I fall in love with the sound of it. Then you're faced with the sort of digital starkness of things that are recorded 'properly'. That's been one of the things that we've battled with as a band; we fall in love with the grittiness of dirty recordings, whether that's four-track or on loop pedals that are oversaturated."

An unexpected benefit of this new approach was that Philippakis ended up recording parts in his Logic demos that would actually end up on the album, most notably the degraded, dubby bass line that is a central feature of the track 'Syrups'. "I think most of the first half of 'Syrups', with the exception of the drum beat and the vocals, is essentially the Logic version," he says. "And I didn't know what I was doing. I knew I had the preamps on too cranked and I was recording hot into the computer. But the result of it is some great sounds and textures in the early loops that are only found in naïvety, y'know. We wanted to capture all of the accidents and all of those new moments.

"The difficulty in making records in general," he adds, "is that often you never capture that initial burst of excitement,

the first time when the magic comes into the room and you're playing something as you're discovering it. And the sound of that initial musical discovery, whether it's one person on their own in Logic or four people in a live room, we'd found that we'd never really managed to capture that in the past."

Parallel Lines

In tandem with the singer's early Logic experiments, the newly four-piece Foals gathered in a South London rehearsal studio to jam through new ideas. As recording later progressed over the span of a year, they would repeatedly shuttle back and forth between 123 and the practice room, reworking and sometimes utterly transforming tracks.

"Where the record has ended up is through the tension between the two different approaches," says Philippakis. "The tracks that I guess really got built up in a sort of sedimentary way, because of all the going back to the live room and then having to re-translate stuff, would be 'In Degrees' and 'Sunday'. That was quite a beast to record. It has three different drum kits in it. It was a massive session, that one."

"Yeah, there were some pretty large sessions by the end of it," laughs Brett Shaw. "The biggest was probably this 10-minuter called 'Neptune', which will be on the second album. I remember looking at the folder at the end of the session and it was 50GB worth of audio in there. I've recorded albums that are smaller than that!"

Space Saver

Brett Shaw set up 123 Studios in a warehouse space in Peckham four years ago, after the rent was tripled on his previous studio in Shoreditch. Shaw has been involved in music since the age of 16, when his band South were signed to »

Following the departure of bassist Walter Gervers, bass duties were handled jointly by Yannis Philippakis and Edwin Congreave (left). 123 Studio owner Brett Shaw (right) engineered and co-produced the album.

UK trip-hop label Mo' Wax and went on to provide the score for the now-classic 2000 Brit crime film *Sexy Beast*. In more recent years his engineering and production credits have included Florence & the Machine, Lady Gaga, Clean Bandit and Daughter.

The building of 123 took some work, much of it by Shaw himself, and the facility's control room was designed by acoustician Nick Whitaker. "When I got here it was just a total shell of a warehouse," Shaw remembers. "It didn't have a roof on. I had to get the landlord to put one on and then I spent two to three months building it, getting very good at carpentry and stuff along the way. The wood walls have only been there for the last year or two. Going to listen to music in spaces that have wooden walls, even classical music, the way the strings bounce off, the reverberation is a lot nicer. I like the way it looks and it gives a softer reflection."

Two years ago, Shaw installed an SSL E-Series desk at 123, but for tracking he tends to favour his array of outboard preamps, which includes a Chandler REDD 47, Tree Audio Branch and Seventh Circle vintage Neve and API clones. "They give you a lot of colour on the way in, so I just wanted a desk that was essentially good to mix on, not adding a lot of colour," he says. "The SSL adds a little bit of a flavour, but it's just a sort of versatile desk that has a useful workflow."

Over a year was spent refining the material in a rehearsal studio and tracking at 123 Studios in London.

After Shaw lent a hand to Philippakis in his production room, the Foals frontman suggested that the whole band get together with the engineer/producer in the main studio at 123 for a trial period. "He was like, 'Do you fancy doing a week or two just to see how it goes and get some ideas a bit further developed?'" remembers Shaw. "Then in those sessions I guess we got on to the extent he could see that this might work. He sort of made the joking comment, 'What are you doing for the next six months?' I was a bit nervous. Then it just went on and off and on and off for the best part of a year, basically."

"We sort of took over his life a little bit," laughs Philippakis. "The studio's great. It has a great atmosphere to it. It's very laid back and it's very light which was a pleasure to work in. Somewhere that felt light and warm."

"Yannis is the main sort of driving force of Foals," Shaw stresses, "and he has a very clear direction of how he wants things. But he's not so technical, so he left a lot of that

A Circle Of Mics

Having no real constraints on their studio time, Brett Shaw and Yannis Philippakis chose to set up a circle of mics in the live room at 123 to try out different vocal sounds. "Yannis sort of insisted on finishing all of the music before we got into the vocals," says Shaw. "I set up usually six microphones in a circle and he could just sort of skip between any of them at any point. You can just record a bit and see what it sounds like."

The circle of microphones generally comprised a Shure SM7, Neumann U67, Flea 47, an Altec 633A 'salt shaker', an AEA R44CX ribbon and a Yamaha NS10 speaker cone used as a mic. "The 'salt shaker' is a real lo-fi, gravelly thing," says Shaw. "That worked on some of the songs where Yannis really pushes his voice hard and heavy and dirty. The SM7 is probably the mic he's used most on the other albums. He'll sort of fall back on that, 'cause he knows that it's worked for him throughout the years.

"On the slower songs and warmer vocals, the Flea 47 sounded great. He'd get up close to that and it'd give it a big, deep dimension. The ribbon AEA again sounded pretty nice on the softer stuff. And then we messed around a lot on the backing vocals singing through the NS10 cone. He'd often like singing through an Echoplex."

"The Echoplex was quite integral to the vocals," says Philippakis. "Often it's the main effect. I feel like between the NS10 cone and the Echoplex, those were the two vocal textures that kind of allowed us to use different mics, but we always knew that there would be a kind of repeated thread. There would be a sonic motif running through both albums. And also it was just to get away from clinical vocal sounds where it's too crisp and too clear. I feel happiest when there's dirt and there's some impurities in the sound."

In the vocal chain, the preamps tended to be the Tree Audio Branch and the Chandler REDD 47, while various compressors were used. "I'd usually go through a Teletronix LA-2A, sometimes a Distressor, sometimes an API 2500," Shaw says. "Just a little light compression on the way in. I got in

trouble once for overdistorting a vocal cause I liked the sound of it [*laughs*]. He told me not to commit to things after that. On the end of 'Syrups', I was driving the tube stage of the LA-2A by distorting the output and then attenuating it on the Distressor afterwards, taking the level back down. I think it sounds great. But it was just a bit too sort of grain-committed. And it got used on the album exactly how it was. But yeah, after that I sort of backed things down a little in terms of my recording!"

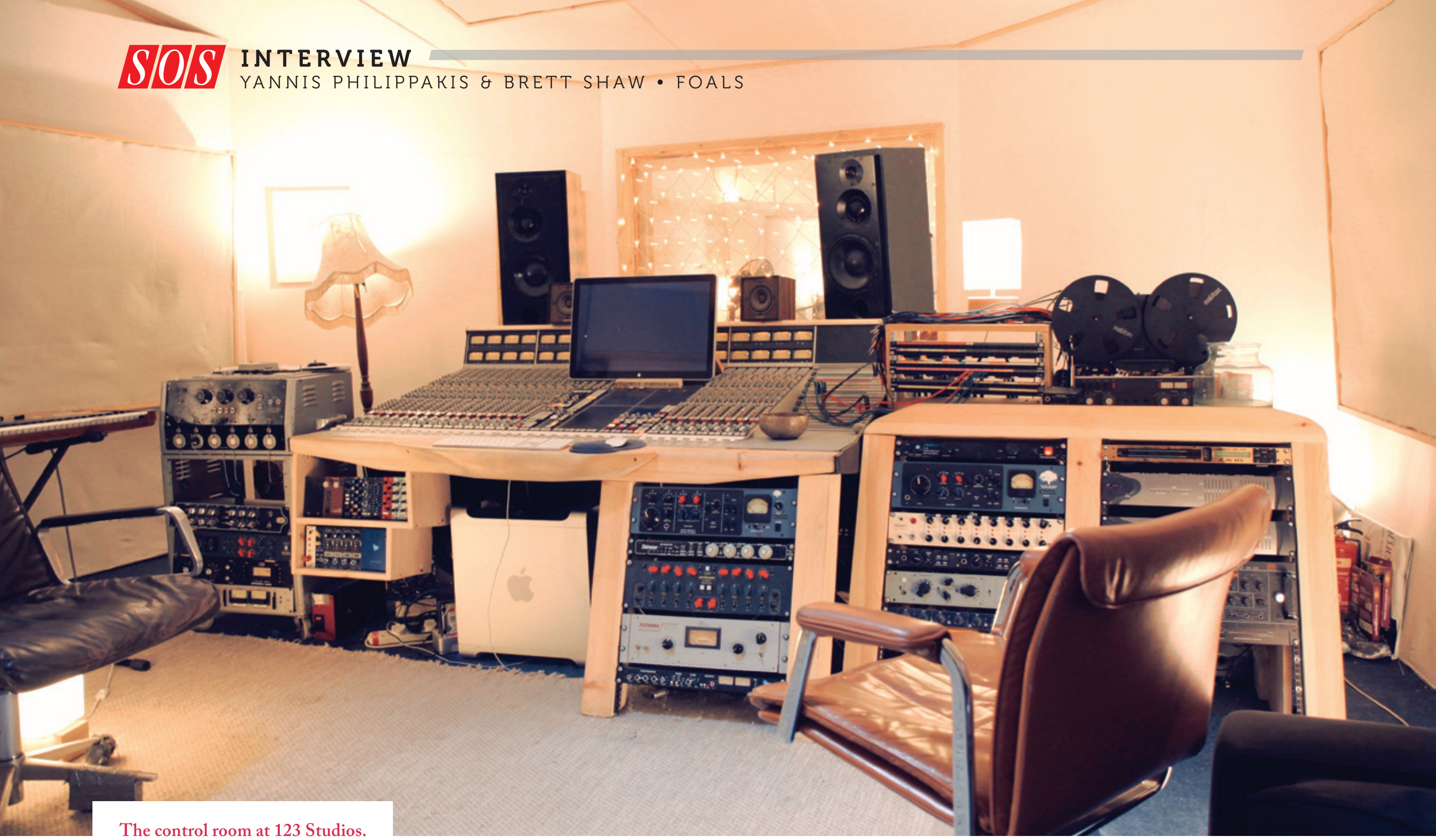
At one point in the vocal process, feeling that cabin fever had set in, Philippakis disappeared to Greece for a week. "I'd been in the studio solidly for over a year, basically," he says. "Y'know, one of the aspects of this way of working was I felt that there was more than the usual amount of pressure on me, because fundamentally the buck stops with me about the production. So there was a point definitely where I was cabin-fevering. I needed to go to Greece just to clear my head for a few days. And actually I think it was great. Actually, I should've spent most of my time in Greece...

to me. It was a bit like a creative forum. Everyone could put ideas forward and we'd really try them, whatever they were."

Tracking

Given the nature of Foals' music — whether it be their harder, grungier side or their more dance-friendly tunes — there was huge importance placed in the preliminary stages on the rhythm tracks. Working with drummer Jack Bevan's Tama kit and the studio's 1966 Ludwig Super Classic, Brett Shaw dug deep into his collection of mics.

"It was always different, depending on what song we were doing," he says. "One of the advantages of having so much time was that I could experiment for each song with different miking techniques. But generally, the overheads were a Neumann U67 and a Flea 49: one on the top, one at the back, in a sort of Glyn Johns style. Or on some of the slower songs, it was Coles 4038s as overheads. Snare was usually a mixture of a Neumann KM84 that I'd crush a bit going into a Tree Audio Branch preamp and distort a bit, next to an Audix i5, which is not a very sexy mic, but it sounds good on a snare. The distorted 84 would do the warm, fat thing and »



The control room at 123 Studios.

the Audix would do more of a clean thing next to it and you can balance between the two sounds.

"Kick drum was the usual AKG D12, using a FET 47 or Electro-Voice RE20 at the back. Sometimes I'd put close room mics in, figure-of-eights, pointing at the floor, either side of the kit, which were either Coles or sometimes an AEA R44 and an RCA 77. There was no strict setup for any song. One thing I did use quite a lot on the slower, softer songs was the 77 sort of quite low over the middle of the drum kit, looking just over the snare and the kick drum. But that was just too mushy on the fast, hard songs. So I swapped that out for the Bock iFET looking down just over the kick drum and the snare drum, basically quite close to the drummer's knee."

Meanwhile, a single bass rig was shared between Edwin Congreave and Yannis Philippakis, comprising an Ampeg SVT for the miked sound and Zod Audio's ID DI. "It's a guy out in America who makes them," Shaw says of the latter. "But they've got the biggest, fattest, warmest sound I've ever heard in a bass DI. Yannis would come up with lots of ideas for bass. He's a bit like a riff machine. You sort of wind him up and watch him go and he'll just come up with 10 different ideas. He attacks the bass a lot harder than Edwin, who tends to play with fingers."

Guitar effects pedals became very important to the sound of the album.

In initial tracking, with all four of the band playing, the team were looking to nail really only the drums and bass. "Sometimes we kept a guitar," says Philippakis. "If it had a kind of charm to it, then it was useable. Largely we wanted to try and capture some excitement, to get the core skeleton of something and then dress it up as we went on. There's a couple of tracks where it is actually all the initial takes. The 10-minute-long track 'Neptune', that jam space that happens in that song was something that was totally natural in the room and we just played it out."

"I tried to keep it as live as possible," says Shaw. "Because they are essentially a jam band — a lot of their writing comes out of jamming together, which is a really good thing. There aren't enough bands like them doing that kind of intricate jamming thing that songs come out of."

Pedal Power

Recording Jimmy Smith and Yannis Philippakis' guitars was a highly experimental and freewheeling affair,

Heading to Studio La Marquise in Paris allowed the band to explore Vincent Taurelle's synth collection.

» involving chains of guitar effects. "They've got more guitar pedals than I've ever seen a band have, ever," Shaw laughs. "They've got, like, three crates of guitar pedals that would get splayed across the middle of the control-room floor. At any one time you could reach for about 200 guitar pedals and get interesting sounds."

"Yeah, well," adds Philippakis, "Jimmy went on this shopping spree of pedals just before we went in there, so we had all these new toys. It would've been rude not to use them really!"

Chief workhorses in the effects pedal department included Audio Kitchen's The Big Trees valve preamp; OTO's BIM 12-bit delay unit, BAM Space Generator and BOUM Warming Unit; along with a Klon KTR Centaur overdrive, MWFX's Judder analogue sampler/repeater; and Hologram's Dream Sequence pitch-shifter/sequencer. In addition there was also much use of a Maestro Echoplex and Roland Space Echo for delay, and the Eventide H3000 Harmonizer.

According to Shaw, Smith and Philippakis can be quite competitive over which of them could create the most interesting or unusual guitar sound. "Cause they've got this thing where if one of them gets a sound first, then the other one's not really allowed to step on their shoes," he laughs. Overall, Jimmy Smith tends to use a Fender Jaguar through either a Fender Twin Reverb or Roland Jazz Chorus. Philippakis favours Hiwatt amps or his 1960s croc-skin Selmer, and almost exclusively plays Travis Bean guitars. "I've played them since I was 18," he says. "I play other guitars in the studio, but they're just a unique-sounding guitar. To me they sound like how a guitar is supposed to sound. There's something transparent and pure about them in a way. It's not coloured in one way or another. I find that it's quite a blank sound in a good way. The neck is super flat and you can play higher on it, which is probably part of the reason why in the early days loads of our riffs were written above the 12th fret. The other guitar I played quite a bit on the record was a '67 Gretsch Country Gentleman that I picked up in Cincinnati, and it smells like your grandma's house."

Roughing Up

Foals' use of synths increased on *Everything Not Saved...*, with Jimmy Smith and Yannis Philippakis sometimes abandoning their guitars in favour of keyboards, to supplement the main electronic role in the band played by Edwin Congreave. Much use was made of hardware synths, including 123's Roland Juno-60 and the band's Korg MS-10 and Minimoog Voyager. Philippakis meanwhile experimented with a Rhodes Chroma Polaris, which he'd picked up while on tour in the States. "I had some MIDI instruments on Logic for the track 'Café D'Athens' — a marimba part and a vibraphone part — and I sent the MIDI into the Chroma Polaris. It's just an amazing synth. Very intuitive and it felt like an important synth for us to use on this record."

More synths were employed when the sessions moved temporarily to Studio La Marquise in Paris, where Foals decided to "maul the tracks" with producer and Air keyboard player Vincent Taurelle. "He had some great synths," says Philippakis. "He had a bunch of Air's synths he was looking after — a Yamaha CS-50 and a Memorymoog that's on a lot of the tracks. Working with Vincent, it's spontaneous and it's aggressive. The way that he works is not sonically polite. He's oversaturating a lot of things: tape, his desk, reverb chains. He likes to drive things really hard and likes to experiment and he's got a bunch of modular gear that's great.

"We'd been working for a long time in 123 with Brett and we'd got into a kind of groove of how to work, which is great. But sometimes that needs to be challenged. The songs needed a bit of roughing up and Vincent's the guy to do it with. He has access to an amazing percussionist and there's vibraphones and marimbas and all sort of other instruments. So I definitely feel like, going

Everything Not Saved Will Be Lost, Part 1 is Foals' first album as a four-piece. Clockwise from left: Jimmy Smith, Jack Bevan, Edwin Congreave and Yannis Philippakis.

Photo: Alex Knowles

in there, it's like a child going into a sweet shop and he's Willy Wonka!"

Another addition to the team in the later stages of recording was James Ford, the Arctic Monkeys/Gorillaz producer who had helmed Foals' previous album *What Went Down*. Ford was brought on board both as a consultant and as someone who could provide additional production. "James added lots of bits and pieces and ideas," says Shaw. "We kept a running dialogue with him throughout the album. Sometimes we'd have tracks where we'd need some advice and we'd reach out to James and say, 'What d'you think is the best way to go about this?' And he'd send back some ideas or say, y'know, 'Try this and that.' It was kind of useful to have that to bounce off."

"All the way through I was sending him tracks," says Philippakis. "There were some tracks that we thought would benefit from letting James squeeze the extra 10 percent out of them or just shine a different light on them. So he did that on 'In Degrees' and then there's some tracks on Part 2 where he helped massively."

Inhabiting The Songs

Mixing duties for both Parts 1 and 2 of *Everything Not Saved Will Be Lost* were handed over to Mark 'Spike' Stent. "He mixed the final versions," says Shaw, "and everything just got a bit more beefy and bigger-sounding."

"I thought that we had fully inhabited the songs for long enough," says Philippakis. "It was quite possible that we'd lost all sense of perspective. It was probably a good time to have an experienced outside pair of ears. He sent me mixes every few days and I just sent him reams of notes."

Eventually, Foals completed over 20 songs, and it was decided that the tracks should be released as two albums. The result is the most ambitious and experimental music that the band have yet produced, which at the same time maintains their commercial appeal. Ask them to define the two albums and, while cagey about the second, Philippakis will say, "I think the guitars are more emphasised on album two, basically. And in some ways, it works as a response to album one."

"The first one I think is a lot more groove-based," says Shaw. "There's probably more experimenting, and the second one I think is a lot more direct. There's one track on there that is probably Foals' hardest, heaviest song. But it sounds great. It's one of those that the fans in the mosh pit are going to love."

Studio File

MOSH,
SÃO PAULO

HANNES BIEGER

The largest country in South America, Brazil boasts a musical history like no other, spanning classical music, heavy metal, hip-hop, rock and numerous electronic genres. Some genres have their spiritual homes in particular cities: for instance, it is widely accepted that bossa nova, one of Brazil's most famous musical exports, first emerged from Rio de Janeiro. However, the largest Brazilian city, São Paulo, can be seen as the country's overall cultural and business centre. With a population of over 30 million in its greater metropolitan area, São Paulo is not only the largest city in this continent, in the Western and Southern Hemispheres, but also the 11th largest in the world!

There has long been a thriving music scene in São Paulo, and where there is music, there are recording studios! No surprise, then, that one of the largest and most active studios in Brazil has been operating there for 40 years. MOSH Studios takes its name »

The spectacular Studio A live room occupies over 100 square metres, with four-metre-high ceilings.

The post-production suite houses a collection of rare keyboards and synthesizers, including a vintage Moog System 15.

from the initials of the two founders, Oswaldo Malagutti Jr and Hélio Santisteban. Malagutti recalls setting up the studio in the '70s: "My career as bass player and singer in bands began very early in 1962, playing covers of the Shadows, the Ventures, the Beatles and Stones in several rock bands. After 1972 I had a great success with the band Pholhas. I've always had an interest in electronics, PA systems and audio recording, and I have learned a lot at RCA studios. We were artists on the RCA label where we recorded some Brazilian hits that reached sales of over three million records at the time, and after a while we decided to build our own studio."

Mosh On

This first studio was located in São Paulo's neighbourhood of Pompeia, and openend its doors at the end of 1979. First equipped with a Sound City console and an MCI JH8 tape recorder, MOSH became busy very fast. Co-founder Hélio Santisteban left the company in 1983, but the studio was thriving regardless, and eventually it outgrew its first facility: more space was needed, and there were problems with neighbours. Malagutti found a new place in a former jewellery factory in the Água Branca district of São Paulo, a commercial area that provided everything he needed: soundproofing, easy access, ample parking lots, and space to grow. The current MOSH premises were designed by Jeff Forbes, and were ready for recording in 1988. The initial MOSH Studio A was followed by Studio B in 1990, Studio C in 1992, Studio VIP in 1994 and Studio D in 2000, and today MOSH is a seven-room facility, with spaces ranging from smaller suites for mastering and post-production to the large flagship rooms.

The heart of MOSH is still Studio A, which, bar minor refurbishments, remains in its original condition. With a live room of 100 square metres and a control room a little more than half this size, it offers ceiling heights of over four metres. Beautiful woodwork and bold and unique colour patterns in the live area add to the expansive ambience. Equipped with a 56-channel DDA console and Studer A827 tape recorders, integrated into the digial world with a CLASP 24 system, it also boasts a huge outbard collection.

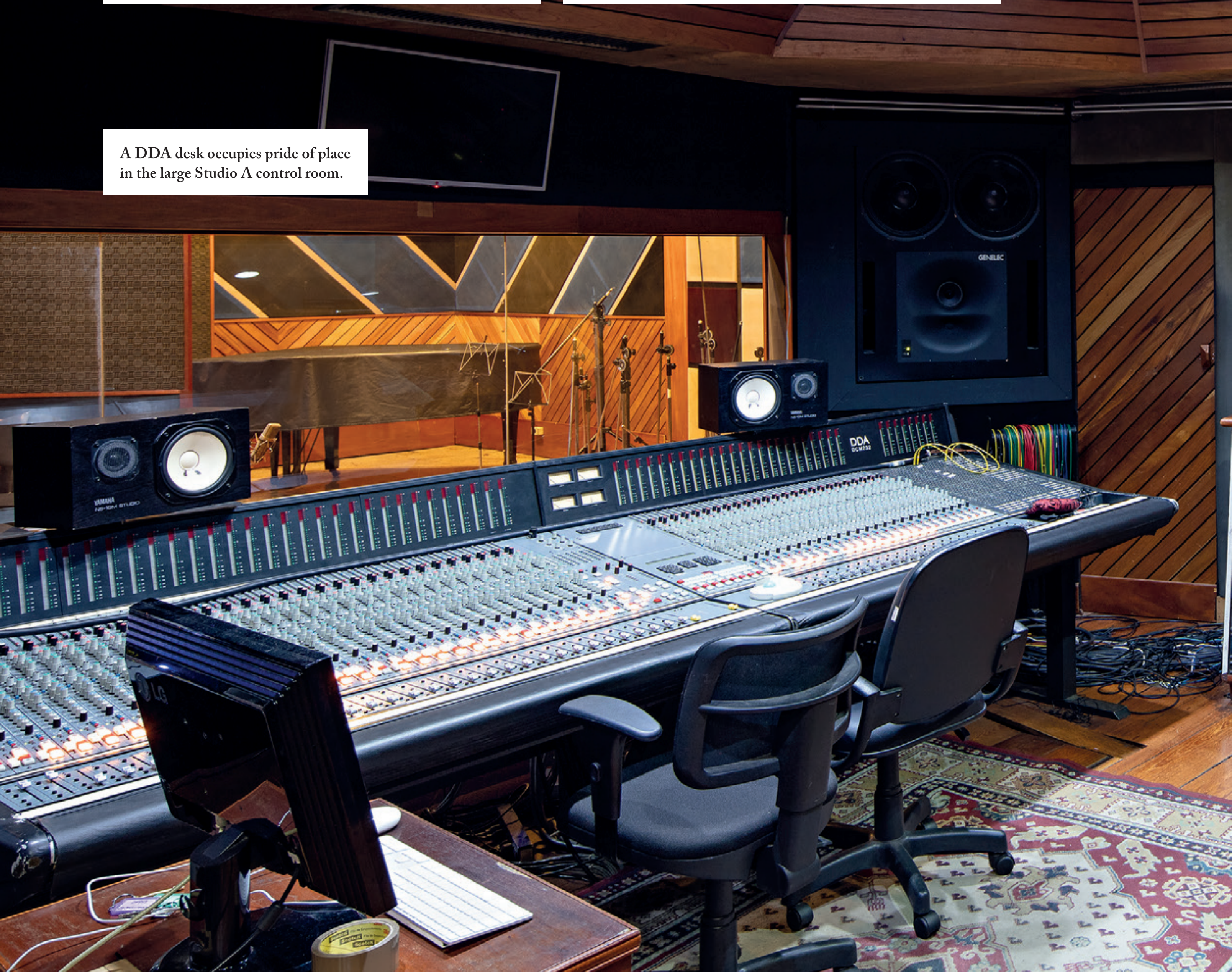
The outboard in Studio A includes rare units such as an Eventide Instant Flanger and a UTC 4-B filter.

The six side racks are occupied by several Pultecs, a DW Fearn VT-1 preamp, Focusrite Red Series units, a Manley Voxbox, seven Amek 9098 channel strips and several UREI LA-4 and Purple Audio MC76 limiters, amongst many other high-end units.

Studio B is also built around a 56-channel DDA; its rooms have smaller proportions, yet the live area is still big enough to host a dozen players at the same time. The largest space of the facility is Studio C, with a live area space of 130 square metres. This multi-functional room serves as recording space for orchestras, but also as a filming stage. Studio D, meanwhile, is based on a 96-channel SSL Axiom digital console and boasts the largest Pro Tools system in the entire facility, again with plentiful analogue outboard. Malagutti's favourite room is Studio VIP, equipped with a vintage Neve V2 console from 1984. "It first belonged to Record Plant Studios in New York, and it came to Mosh in 1993," Malagutti explains. There is a fully equipped mastering room, and also a post-production suite that is now home to an enviable synth and keyboard collection that includes a vintage Moog System 15 modular synth. On top of that, MOSH offers a collection of over 150 microphones.

Star Turns

Over the decades many famous musicians have worked at MOSH, which today also runs a record label and offers artist development and management plans. Famous Brazilian artists who recorded at MOSH include Zezé di Camargo & Luciano, Leandro e Leonardo and Chitãozinho e Xororó, all of whom have sold dozens of millions of records, as well as legends Roberto Carlos, Caetano Veloso, Rita Lee and the heavy metal group Sepultura. International clients include Sérgio Mendes and Black Eyed Peas, amongst many others.



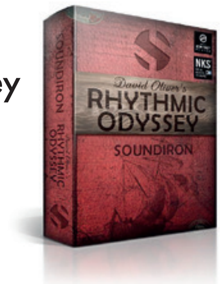
A DDA desk occupies pride of place in the large Studio A control room.

EMT 258, 260 and 261 dynamics processors in Studio D.

SAMPLE LIBRARIES

Soundiron
Rhythmic Odyssey
Kontakt Instrument

★★★★★



Rhythmic Odyssey represents a collaboration between Soundiron and the highly regarded UK-based percussionist David Oliver to develop a virtual instrument built around live percussion performances. The sound sources — and all the performances — are David's and span a huge range of both tuned and untuned instruments. In essence, the library is a collection of over 6600 individual performance loops, all recorded as 24-bit/48kHz WAV files and totalling some 12.5GB once installed. The loop collection is built from 100 rhythmic compositions, each of which features up to 12 layers. Users have access to the raw WAV files but also get the same material presented via a feature-rich Kontakt front-end (v.5.7.3 or later of either full or Player versions are supported).

Let's start with the sound sources. These span conventional bass drums, daikos, a number of different box drums, large hand drums such as congas, bodhrans, and djembes, and small hand drums such as bongos, dhol, djembuka and udus. There are also various metal percussion instruments including agogo bells, cowbells, finger cymbals, standard cymbals and... well, bicycle spokes, tin cans, jerry cans and an oven grill, with miscellaneous extras including vibraslap, various tambourines, shakers, rattles, cabassa, a bucket and plastic engine tubing. This is an impressive array and, thankfully, is matched by equally impressive performances and recording quality. Isolated, or in combination, the loops sound full and fabulous.

The front-end is perhaps a little quirky but comes in three different flavours — stereo mix, multi-track or single-track — allowing you to access the loop collection in different ways, but each provides plenty of scope for manipulating the loops and applying effects. The stereo mix and single-track modes provide pre-mixed loops or single loops to work with, while the multi-track mode (which was where I spent most of my time) allows you to work with up to four single loops in the same instance. In each case, however, you set a set of 'clip slots', each able to house a different loop, and with keyswitching between them. This makes it easy to build plenty of variety into an overall performance without needing multiple instances of Rhythmic Odyssey to be loaded.

That flexibility goes a lot further, though, as each clip slot can also have its own settings for the various slice parameters and performance controls. There are lots of creative options here and, while the untreated loops give you an authentic natural sound, you don't have to stick with it. You can, for example, sequence volume, pan, pitch, ADSR, and filter cutoff/resonance on a per-slice basis for each clip slot. Equally, a whole range of effects are provided including EQ, dynamics, distortion and ambience. With the ability to trigger both the whole loop or individual slices, reverse playback, and options to match the host tempo via stretching or slicing, the sound design options are impressive.

My guess is that Rhythmic Odyssey will most obviously appeal to media composers. In that regard, it is very competitively priced considering just what is on offer. The sounds would not be out of place in productions with the highest of budgets, but the workflow is efficient enough to help with the tightest of deadlines. An excellent choice for those needing a quality source of inspiring hand percussion content. *John Walden*

$149

www.soundiron.com

Modwheel
Bass Banjo
Kontakt Instrument

★★★★

When I am looking for unusual virtual instruments that are simple to use and somewhat unique, Modwheel's website is almost always my first port of call. Their latest offering, Bass Banjo, is based on a bespoke instrument originally built by one of Modwheel's founders, David Donaldson, together with collaborator Stuart Porter, for performance duties with an outdoor marching band. The pair made the instrument by combining a bass drum with a modified double bass fingerboard.

Having thoroughly sampled this curious instrument, Modwheel ended up with a 2.2GB core library of sounds (reduced to 1.07GB using NI's lossless compression facility), including four round-robin variations and multiple articulations. They then created 10 separate Kontakt interfaces to segregate various performance methods. For example, five of the interfaces are home to what Modwheel call Bang-On Sidekick percussion sounds, which were created by striking the drum in various ways rather than plucking the instrument's strings. Generally I prefer Kontakt instruments to have a single interface so that everything is in one place, however Modwheel's approach makes it possible for each interface to be elegantly simple.

The first of the interfaces is built using the basic bass banjo notes, which are not entirely inspiring on their own, but it does provide plenty of ways to process and develop them. Its Oomph control adds a tightness to the bottom end, Heat introduces tape saturation, Note Off controls the level of the finger noise on the fretboard and moving the modulation wheel adjusts the level of the performed slap. There is also a 32-step arpeggiator, a reverb and a delay.

In addition to all this is a set of keyswitch controls that govern the articulations, so that the user can decide whether or not they want sustain, staccato, muted or mildly percussive 'chopstick' notes.

The second interface has been given exactly the same controls at the first, but offers soprano sounds that are nearer the register of a traditional banjo.

Interfaces three to seven serve up a wide variety of percussive sounds and looping patterns. Amongst the octaves are a mixture of string scrapes, dampened drum hit, rattles, tones, taps and grooves, enabling the user to quickly create something approaching the rhythmic arrangements of Tom Waits' *Swordfishtrombone*, or those at the end of Mike Oldfield's *Ommadawn (Part One)*. Once again, there are plenty of sound processing tools to try out, many of which are the same as those given to the first two interfaces, the main difference being that several of the percussive variety include a phaser in place of the keyswitch controls.

»

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» The final three interfaces provide synth sounds which, with names like JamJarre and Bangelis, leave little doubt as to which musicians inspired them. These too include a phaser rather than keyswitches.

Virtual instrument shoppers might not immediately be tempted by the idea of a bass banjo, but this very affordable product has a lot to offer in terms of percussive possibilities, sound processing and synth sounds. My advice is to give it a try and see where it takes you. *Tom Flint*

$39

www.modwheel.co.nz

Unorthodox Audio
St James The Great Organ
Kontakt Instrument

★★★★

Unorthodox Audio are a newcomer to the arena of Kontakt instruments, currently offering two products at opposite ends of the sonic spectrum: Superfly Drums and St James The Great Organ, which, as the name implies, has been built from samples of the pipe organ installed in the church of St James The Great in Snitterfield, Warwickshire. This is a 19th-Century English church organ with dual four-and-a-half octave (C-G) manuals, a two-and-a-half octave pedalboard, and just 16 stops (of which 14 are recreated here), so don't expect the grandiose power of a large cathedral organ. The 14 stops comprise two reeds, five principals, five flutes, and two strings, the longest of which is the 16' Bourdon rank assigned to the bass pedals, and the shortest of which is the single 2' rank. The Celeste (an 8' string) is tuned slightly flat so that it choruses gently when combined with other stops. You can transpose the playback within the instrument itself by ±12 semitones but, to stop munchkinisation, this selects the appropriate samples rather than pitch-shifting those under each key. Alternatively, you can use the transposer within Kontakt to pitch-shift the samples. While this can change the characters of sounds for additional flexibility, it generates noise, so I wouldn't in general advise it.

Unorthodox Audio have included several facilities in the Kontakt instrument that are not present on the original organ. For example, there are individual volume and pan controls for each of the stops, and you can assign MIDI CCs to each of these to automate the registration. There's also a control for a virtual swell box. On pipe organs, the swell box reduces the amount and pressure of air reaching the pipes played from the Swell manual, causing them to speak more slowly and at a lower volume. Here, the swell box reduces the volume but doesn't appear to affect the attacks of the notes. Next we come to the Stereo Width control, which uses the standard Kontakt stereo width algorithm. At its maximum, this spreads the organ across the whole panorama, which is a pleasing sound. In addition, Unorthodox Audio modelled the reverberation within the church and added this to the instrument. You can apply this using the Reverb slider or, of course, utilise an external effect of your choosing if you fancy something with greater intensity and a longer decay.

'Combinations' is another facility that was not found on the original organ but has been added to the Kontakt instrument. There are six of these, and they are, in effect, pipe organ memories within which you can store wanted registrations (combinations of stops). For live performance you can select these almost instantaneously by activating Key Switching, selecting Combinations mode, and then playing the white keys C0 to A0, which lie outside of the musical range. (If you're not in Combinations mode, you can also toggle individual stops on and off by pressing the keys in the range A-1 to Bb0.)

Aside from the lack of 32' stops and the massed ranks that would fill the Royal Albert Hall with musical grandiosity, the most obvious limitation of the St James The Great organ is the fact that it only has octave stops. This means that there are no fifths or other harmonically interesting intervals available, and therefore no Mixtures to create complex sounds. Nonetheless, its voicing can still range from gentle to strident, and it comes with nine example presets that demonstrate this well — but always within the limitations of being a small church organ.

If you choose to load a single instance of the Kontakt instrument and play it from a single keyboard, you'll find that the Swell and Great manuals and, over its more limited range, the pedalboard are permanently coupled, which greatly limits the range of sounds you can obtain and how you can perform with them. It's therefore worth spending some time setting it up correctly, loading three instances within Kontakt, assigning each an independent MIDI channel, and then playing the manuals and pedalboard separately as nature and JS Bach intended. If you do so, you'll find it a much more convincing and characterful instrument than if you layer everything over everything else.

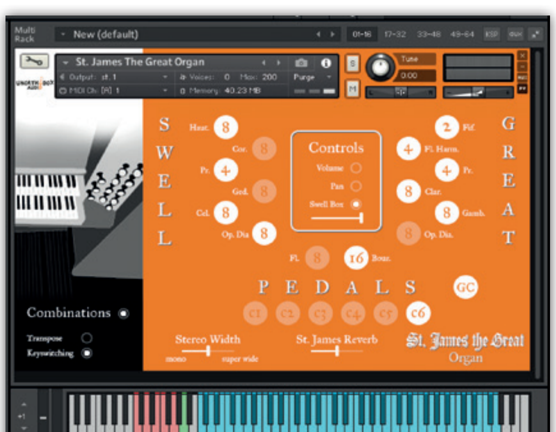
I only encountered one problem during my tests. On very rare occasions that I couldn't recreate but may have been linked to my experiments with pitch-shifting, I was able to disable the loops of some of the samples so that some of the notes under some of the stops played for just a few seconds and then stopped. Closing and reloading the instrument always fixed this, but it will be good when the bug is identified and eradicated.

Small church organs are not mainstream instruments in modern music, but they are nonetheless interesting, and they are able to add another flavour to suitable compositions. The St James The Great Organ is a nice example of the genus. There are six excellent examples of its use on the company's website and, if you've heard these, there's nothing that I need to add. If you haven't, listen to them and, if you like what you hear, it could be worth the small investment.

Finally, it's worth noting that one of the founders of Unorthodox Audio learned to play the organ on the original St James The Great organ, and the company will be donating some of the proceeds from the sales of the Kontakt instrument to keeping it serviced and in good condition. That's another damn good reason to buy it. *Gordon Reid*

£36

www.unorthodoxaudio.com



Audio examples of this month's libraries are available at www.soundonsound.com.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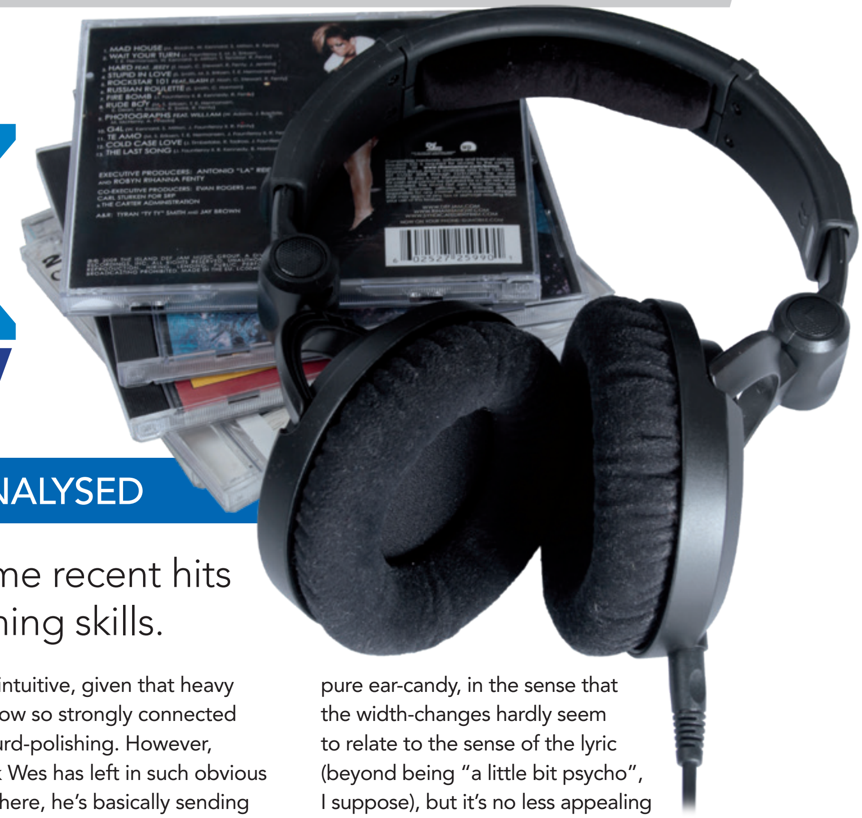
THE MIX REVIEW

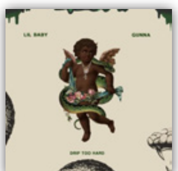
COMMERCIAL PRODUCTIONS ANALYSED



We examine the production of some recent hits to help you brush up on your listening skills.

LIL BABY & GUNNA 'DRIP TOO HARD'

SHECK WES 'MO BAMBA'



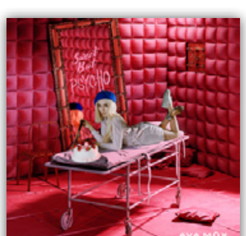
I confess that I'm fed up to the back teeth with hip-hop's blanket abuse of vocal Auto-Tune, a technique which I trust history will smile on as kindly as it has the gated reverb of the '80s. Still, some producers are alleviating the tedium a fraction by tweaking the formula. For example, in 'Drip Too Hard' it sounds like the default chromatic setting has been replaced with something like an A pentatonic scale, which catches the ear by adding an appealingly musical scalar quality to the fall-offs at the ends of many of the vocal phrases.

In 'Mo Bamba', on the other hand, the novelty is the way the vocal melody, for the most part T-Pained into submission, audibly struggles against the Auto-Tune stranglehold at critical moments. For example, many of the E notes are allowed to start off sharp for a fraction of a second before being dragged back into place, almost as if he's singing so forcefully that he's overpowering the processor's correction. (Check out the word "dope" at 0:19 for a particularly strong example.) Elsewhere, there are moments where the correction is clearly mistracking in response to underlying pitch variations in the raw recording, for example "you fuck around and get poled" at 0:49. Bizarrely, this seems to operate almost as a mark of performance authenticity, which might seem rather counterintuitive, given that heavy Auto-Tune is now so strongly connected with general turd-polishing. However, because Sheck Wes has left in such obvious pitch vagaries here, he's basically sending the message: "I'm using Auto-Tune for its sound, not because I'm knee-deep in studio fakery!" And I'll concede that there is a certain swagger in that. (Whether it's actually true, of course, remains a moot point...) *Mike Senior*

AVA MAX 'SWEET BUT PSYCHO'

I'm rather partial to a nice bass dive-bomb, so this production immediately endeared itself to me with the fall-off at 0:17, where the bass pretty much plummets out of the audible band. Even better, it's an integral part of the verse pattern, rather than just a fill, although I did feel just a smidgen nonplussed when it disappeared entirely for the choruses. Still, another low-end rarity soon cheered me up: the lovely little kick-drum roll into the downbeat at 1:17.

The other thing that struck me in this production is how often the vocals do a kind of stereo call-and-response thing, where a central vocal is sporadically joined by wide-panned double-tracks. For example, right from the outset every fourth bar of the main hook section (ie. the trailing "m-m-m-m-m-mine") is widened in this way, a rhythm that changes to every second bar in the verse (0:16), and to the first half of every bar in the pre-chorus (0:30). This is pure ear-candy, in the sense that the width-changes hardly seem to relate to the sense of the lyric (beyond being "a little bit psycho", I suppose), but it's no less appealing for that. *Mike Senior*



CALVIN HARRIS & RAG 'N' BONE MAN 'GIANT'

Finally! I've been waiting for some big hitter in the EDM community to take advantage of Rag'n'Bone Man's extraordinary vocal timbre, and it only adds to my respect for Calvin Harris (of whom I've long been a fan) that he's the one who's stepped up to the plate. And, as I'd hoped, the producer shines plenty of spotlight on his star vocal, during both the verses (0:01 and 1:18) and the breakdown (2:45). Part of me does wish he'd resisted the urge to layer the vocal so heavily elsewhere, though, as this inevitably homogenises the sound to an extent, obscuring some of the grimy details that feel so central to Rag'n'Bone Man's appeal, as I see it. Still, it's a testament to the character of this particular vocalist that so much passion and expression continues to be communicated nonetheless.

What's especially great about this production, though, is that Harris doesn't just sit back and bask in the radiance of his vocalist's unique talent, but instead delivers instrumental material that would arguably have been strong enough to carry the track without any help from a singer. That brass-stab hook at 1:03 is as fat as hell, and I love the way the notes aren't »

» over-short — especially the fourth and fifth ones (both quarter notes), which have great rhythmic punctuation to them, but also enough duration to remain satisfyingly meaty. There are some other lovely nods to classical instrumentation too, for example the reverse piano leading into verse 2 and the cello and double-bass harmonies that follow it. *Mike Senior*

BILLIE EILISH
'BURY A FRIEND'

It was more than 20 years ago, but I still vividly remember walking into my local HMV and being stopped in my tracks by the sound of Fiona Apple's debut single 'Sleep To Dream' coming over the speakers. Since that occasion, no record has demanded quite such immediate attention from me and my wallet, and I'd assumed that was just to do with me becoming a jaded pensioner-in-waiting. But then I heard this single from 'Billie Eilish', and within five seconds the credit card was skipping from my pocket like a spring lamb! Honestly, I haven't heard such a fabulously arresting piece of new chart music for ages.

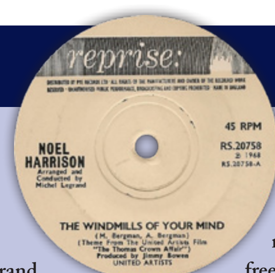
The vocal production is mind-boggling in the variety of vocal deliveries, as well as various cool spot-effects and layers. It's so densely detailed and inventive that it's hard to know what to single out for praise, but personal highlights include the spoken double-tracks at 0:30-0:40 and 1:34-1:44, which progressively increase in intensity; the many claustrophobically dry spoken phrases, such as "come here" (0:24) and "I wanna end me" (0:55); the backing vocals at 0:43, which are then reversed at 0:47; the ominous rattle in the male vocal timbre on "dead by now" (1:50), as well as the purring chorused female "wow" that follows it; and the subtle pitch-dropped layers under "what is it exactly" (0:27) and "what had you expected" (1:31).

Another thing that slays me is how well the rest of the production supports the vocal creepiness. The Foley and atmospheric effects are particularly rich, favourites of mine being the spooky door-hinge squeak at 1:53 and the unnerving dentist-drill whirr at 1:55. Is that a blade swishing through the air at 0:47? Cybernetic rats skittering across the stereo image at 1:05? Some hideous alien tearing through rusty metal at 1:48? A paranoid ringing in our ears at 0:55? The sound-design is so beautifully targeted, fastidiously nuanced, and restlessly mixed that the song comes across as much like a teaser-trailer to some kind of horror-game franchise as it does like a chart single — and I mean that as a compliment, given the extraordinarily high production values you'll frequently find in that part of the audio industry.

But, above all, the sheer bravado of the producers is breathtaking. At 0:57, for example, they deliver possibly the boldest six seconds of production I've ever heard in the charts. That silence between the two low-frequency tones seems like it goes on for ever! In fact, the way pockets of stasis are repeatedly used to generate unease is brilliant, much like those momentary breathless pauses that precede many a cinematic jump scare. In this respect, it's particularly cool that the very last phrase of the song "where do we go" (3:01) trails to silence just as it did at 0:21 and 1:25, leaving you unsure whether the song's actually finished, or whether another of those menacingly up-close male vocal phrases is still waiting to pounce...

Overall, despite the lyric's slightly worrying undertones of glamorising self-harm, I can't recommend this production highly enough to any student of modern production. The more you listen, the more you'll find to appreciate, which is pretty much the definition of great art, as far as I'm concerned. Not to be missed. *Mike Senior*

Classic Mix



NOEL HARRISON
'THE WINDMILLS OF YOUR MIND' (1968)

The recent sad passing of Michel Legrand drew me back to this Oscar-winning song (from his film score to *The Thomas Crown Affair*), which reminds me of something he once said in an interview: "I play with [*music*]. It's a game." The game in question appears to be whether he can out-do 'Autumn Leaves' for capitalising on falling cycles of fifths — a contest in which he triumphs convincingly, in my view!

One of the things that defines any song with a cycle of fifths is where and how it strays beyond the home key. Both songs initially work their way through all seven root notes tonally, though, within the prevailing minor tonality, giving (in Legrand's case) an Eb minor chord sequence of Ebm-Abm-Db-Gb-Cb-Fdim-Bb. But where 'Autumn Leaves' just resolves straight back to the tonic, 'Windmills' breaks the tonal cycle with a glorious diminished seventh chord on A (0:36), thereby intensifying the eventual Bb-Ebm resolution. But where 'Windmills' really sets itself apart is from "keys that jingle in your pocket" at 1:11, where Legrand begins working his way through another two complete cycles of falling fifths, but this time modulating as he goes, passing through Gb major (the relative major), Cb major (its subdominant), Bb minor (the dominant), Ab minor (the subdominant), and Gb major again, before finally allowing the Fdim chord at 1:39 to return us home to Eb minor.

One problem with using a long harmonic sequence as the basis for a song is that it encourages you to follow an overly repetitive melodic contour. Indeed, at first glance, 'Windmills' might appear to justify that criticism, but on closer scrutiny there are some nice features that mitigate it. The first is the way the 'pace' of the melody seems to double as the melody under the last four words of "and the world is like an apple whirling silently in space" is repeated for "like the circles that you find" and "in the windmills of your mind". And once the modulations get going, the melodic contour becomes much freer, culminating in "that the autumn leaves were turning to the colour of her hair", where those characteristic melodic leaps are finally abandoned to considerable emotional effect. (I can't help wondering whether that lyric's a sly wink to Joseph Kosma, too...)

I can't say I've ever warmed to Harrison's vocal performance, though, which feels slightly rushed and lacks real emotional attachment. Of course, given the success of the song, it can be difficult to imagine it any other way. (And the less said about Sting's easy-listening version for the film's 1999 remake the better.) But if you really want to hear this song performed with panache, check out Michel Legrand's own French-language version from 1969, 'Les Moulins De Mon Coeur', and marvel that such a talented pianist, composer, arranger, and conductor was also a tremendous singer. *Mike Senior*

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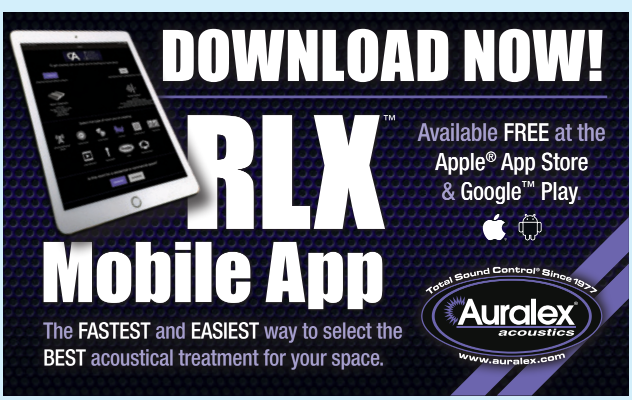

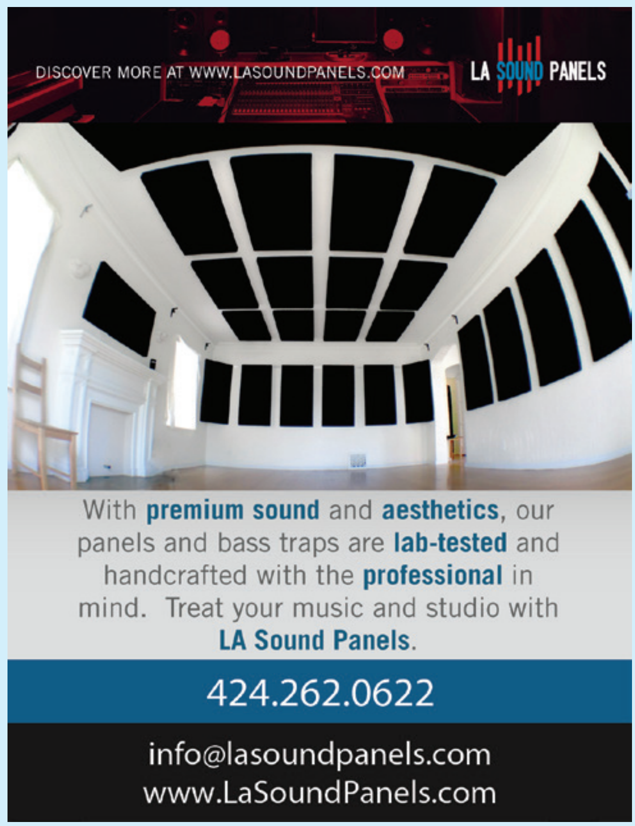

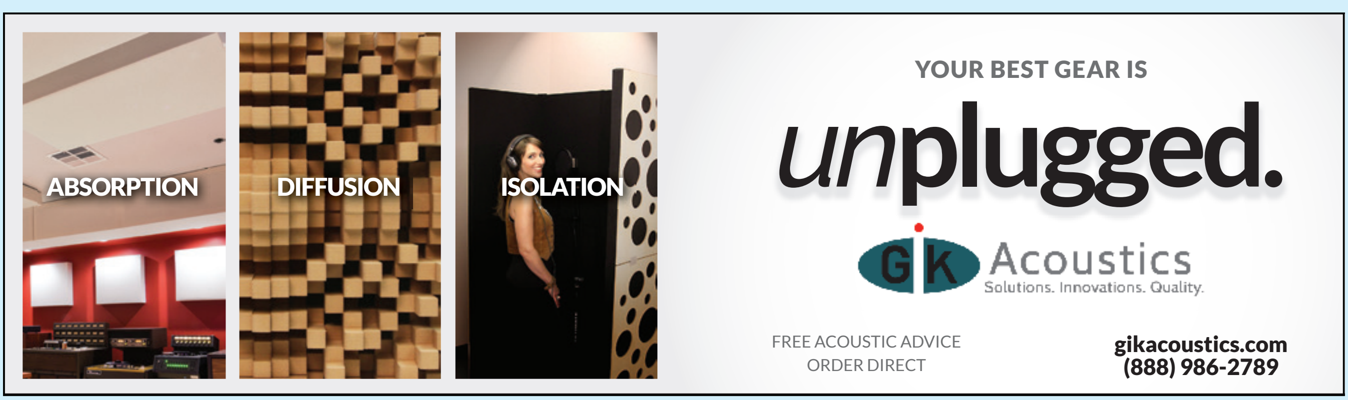

SOS WHY I LOVE...

THE ALBUM

MICHAEL DENNY

Streaming services have revolutionised the consumption of recorded music, with millions of tracks now available at our fingertips. However, I still find purchasing and listening to the physical copy of an album to be a much more immersive, engaging and rewarding experience. As a teenager, I remember receiving the cassette of Oasis' *Be Here Now* for Christmas and saving pocket money for their subsequent album *The Masterplan*. Once I had my first Saturday job I would visit Circa Records every Monday to browse the latest releases to add to my collection, having read about them in the previous week's *NME*. Fast-forward 15 years, and on recent trips to London and Manchester I spent time browsing independent record shops searching for hidden gems, artists recommended to me by friends and releases by seminal artists.

The physical album evokes memories of when I discovered the artist, who suggested I should listen to them, where I bought the record, and moments in life associated with listening to the record. For me, this deeper connection and experience of owning the physical item combines to enhance and enrich the listening process.

From the artist's perspective, the album format gives us the freedom to craft a journey upon which to take the listener. For a creator, a collection of songs can reflect a time period, mindset, the involvement of a collaborator, and in some cases it can be autobiographical. While the same can be said about individual songs, the shorter format limits the scope of this for the writer.

As a listener, it is important to remember that tracks in the context of an album can have a totally different effect in comparison to hearing those tracks in isolation, and some don't always have the same impact when heard as stand-alone works. Nostalgia can also play an important role in the listening process, and often reminds me why I fell in love with music in the first place, thanks to a childhood friend lending me some of his cassettes to listen to. Those were the only records I had access to, so I listened to them on repeat. While the tape could be rewound, the limitation of the playback format encouraged the consumption of the whole album in the way it was intended. Just as it can for a creator, an album can have an association with a time, place or mindset the listener was faced when they first heard it.

As a composer and music producer, listening to an album can inspire my next track or project. Along with the obvious characteristics including genre and instrumentation, I listen for inspiring sounds and textures, chord sequences, rhythms and the way the instruments present are performed and utilised.

While I certainly don't feel the album is becoming a lost art form, instant digital access to almost any record removes some of the magic from purchasing and experiencing an album as a whole. With this in mind, I encourage you to take the time to listen to an album from start to finish, with no distractions. While there is no inherent problem with listening on digital services, perhaps we should be mindful to take the time and effort to listen to albums as they were intended, instead of jumping to the latest playlist or trending track. This mindset and approach to music consumption can enable us to reap the full rewards of the emotions triggered and the overall sensory experience we can have with music.

Michael Denny is a music producer and composer. You can hear his music at www.michaeldennymusic.com.

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