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CREATIVE CUTS

When we think of musical genius, we tend to call to mind people with exceptional powers of planning and control. Whether it's Bach composing fugues, James Brown drilling his band to obsessive levels of perfection or Brian Wilson crafting Pet Sounds over long months in the studio, genius is the person with a special type of command over music and the means of making it.

But there's another type of genius. Exemplified by the likes of Brian Eno, John Cage, Daphne Oram or Miles Davis, this form of genius is all about giving up control. It's about creating conditions where chance can flourish. It's about inviting accident and happenstance into the room, and trusting yourself to recognise magic when it occurs.

When you're working with samples, it's this second form of genius that often comes into its own. The great samplists are those who have been utterly fearless in collecting, juxtaposing and treating samples, and far-sighted enough to identify the seeds of greatness in those experiments. Where most of us would think 'Oh, that doesn't work', the great sample-based producers hear the key to a new sound world.

Genius, by its nature, is given only to a few. In this day and age, though, the tools that genius requires are available to everyone. And when it comes to the creative manipulation of sound, sample slicing is one of the most powerful tools we have. When we load up a loop and assign its individual hits to trigger pads, we usually don't know exactly what we want to do with those sounds. We're not executing a preconceived plan. We're opening the door to chance.

But if chance is to generate magic, we need to give it the opportunity. Genius can issue from experimental, risk-taking boldness only if that boldness is harnessed to technique. Before you can experiment with sample slicing, in other words, you need to know how to do it. You have to become fluent with the process before you can reap the rewards.

When it comes to sample slicing, each platform has its own well-established approach. Hardware options include Akai's historic MPC range, NI's powerhouse Maschine or Elektron's elegant Octatrack; or you could pair a grid-based controller with software such as Ableton Live, Bitwig Studio or Reason. From the perspective of a newcomer, it can seem a daunting choice — which is why we persuaded Simon Sherbourne to write this month's cover feature. Simon has dissected more loops on all of these platforms than I've had hot dinners, and he's the perfect guide for anyone taking their first steps in sample slicing. It's a journey that could lead you absolutely anywhere — and that's the beauty of it!

Sam Inglis
Editor In Chief

"When it comes to the creative manipulation of sound, sample slicing is one of the most powerful tools we have."

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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

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WORLDWIDE EDITIONS

NORTH AMERICA

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EDITORIAL
sos.feedback@soundonsound.com
Editorial Director Dave Lockwood
Executive Editor Paul White
Editor In Chief Sam Inglis
Technical Editor Hugh Robjohns
Reviews Editor David Glasper
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Production Editor Nell Glasper

SUBSCRIPTIONS
subscribe@soundonsound.com
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Circulation Manager Luci Harper
Administrator Nathalie Balzano

PRODUCTION
graphics@soundonsound.com
Production Manager Michael Groves
Head Of Design George Nicholson Hart
Designer Alan Edwards
Designer Andy Baldwin

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

ONLINE
support@soundonsound.com
Digital Media Director Paul Gilby
Design Andy Baldwin
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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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ADVERTISING
paul.dacruz@soundonsound.com
Tel: 707-569-6021
Advertising Manager Paul DaCruz
Regional Sales Associate David Carson

MARKETING
marketing@soundonsound.com
Business Development Manager
Nick Humbert

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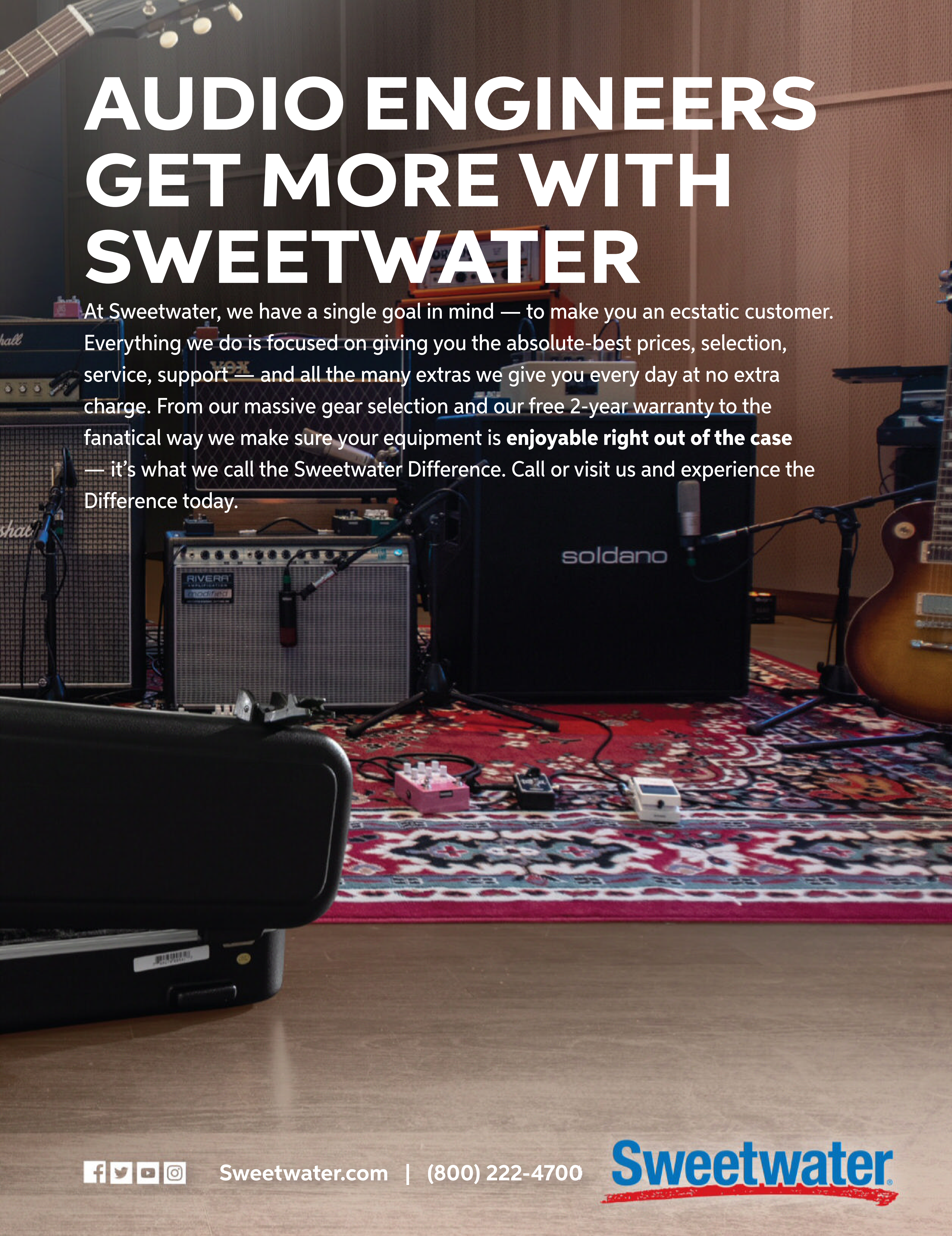
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FEATURES

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ON TEST

WORKSHOPS

M-Audio Oxygen 25

Controller Keyboard

M-Audio's trusty MIDI controller workhorse treads indefatigably onwards.

ROBIN VINCENT

It often feels like these M-Audio Oxygen MIDI Controllers have been around forever. The Oxygen 25 MkV that I have before me definitely has shades of the original MIDIMAN Oxygen8 that first arrived in 2002. They were chunky then and they're chunky now. These are not the stylish controllers that you'd be too scared to take out of your Instagram-curated studio space, no, these are the chunky controllers that get pulled out of rucksacks at Electronic Music Open Mic Nights covered in stickers and graffiti with probably a bit of gaffer tape to keep the sides on. It's quite reassuring that M-Audio haven't gone down the elegance route of so many of their competitors and instead are keeping it real with substantial plastic shells that feel like they're going to survive a chaotic live environment.

The look and the layout of this new version is pretty much identical to the MkIV: same eight knobs, same eight pads, single fader, mod/pitch wheels, three-digit LCD display. They've pulled off all the red accents, turned the grey buttons to black and generally darkened the whole vibe and left all the markings as white. The knobs are still nice and tall and as wobbly as you'd expect from an entry-level controller. However, the knobs themselves feel much better than the MkIV, with a definite white indicator on a shiny cap which protrudes from the edge of the knob so you can detect the position with your fingers.

The keyboard feels good, there's no aftertouch or weighting, it's just a good clean synth action that doesn't make too much noise or feel clunky. The black keys are textured in a slightly matte finish and so are less grippy on the fingers than shiny keys. The velocity response is just fine with a number of different velocity curves or set velocities to choose from. It is a decent playing experience that I would probably be able to say more about on the 49- or 61-keyed version.

New Modes

What is different from the MkIV is all the extra white text above the keys. These refer to the three new creative modes that M-Audio have built into the Oxygen MkV. These are features borrowed from the Oxygen Pro range and are found on most MIDI controllers these days. So, to keep these keyboards competitive it was definitely the right thing to do. »

Included software

The bundle of software is a little on the light side. Usually M-Audio products come with a big bundle of instruments from their family of InMusic products but this time you get MPC Beats with some sound packs, Ableton Live Lite, the MiniGrand and Hybrid 3. The manual says it has presets for many more but they are not actually included. However, once you've jumped through the registration, software manager download, install and iLok activation hoops you do get a couple of half decent things to play with.

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- Hugh Robjohns, Sound On Sound Magazine, June 2020

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» The three modes are Arpeggiator, Chord and Scale with a Note Repeat function thrown in for good measure on the pads. M-Audio haven't added any extra buttons or data sliders to deal with the functionality like they have on the Pro versions, no, for this it's all down to the Shift key and the white text above the keyboard. You select modes by holding Shift and using the Preset Select left/right buttons and a little red dot appears in the display to show the current mode.

The Arpeggiator has all the usual note order modes and a chord mode and a range that expands your held notes up to 2 octaves. There are also controls for Gate length, Time Division and amount of Swing. There are actually two Arp modes, one that latches and one that doesn't. There is a Latch button sitting there looking at you but apparently this only refers to the Note Repeat. Once you're in the Latched mode you can only stop it playing by switching into the non-latched mode or hitting the Panic button. It's a small but annoying workflow feature that a straightforward latching 'Hold/Latch' button would resolve.

Smart Chord mode lets you play chords on a single key. You can set the musical key, voicing and type of chord you wish to play. You have seven different voicing choices plus a random one which is rather fun. For chord type you just get Major and Minor but there is a Custom option where you play a chord of up to six notes and it will use that as the chord, although it won't remember it for next time.

Smart Scale lets you play in a chosen scale so that you never hit the wrong note. You can use the same Key options as Chord mode but now you get a few extra Types including Pentatonic and Harmonic, Dorian and Blues.

All these modes work perfectly well, and they are very welcome things to have in your MIDI controller. However, the Shift+Keyboard workflow is never the greatest way of accessing parameters. Sometimes you want to hold some notes or play while making changes and that's difficult to do, particularly on a very small keyboard where every key has a Shifted function. Some of the options have multiple values which can only be accessed by repeatedly hitting the key to step through them. So, while it's perfectly functional it doesn't offer much in the way of control during a performance. One exception is the Arp Time Division control which can be accessed by holding Note Repeat and using the Preset Left/Right buttons — that at least is a useful thing to be able to change while holding down notes.

I should mention Note Repeat which is a function that enables the pads to do a roll when held. You can either enable it momentarily or engage it with a latch just like the Arp should be able to but doesn't. The Preset Left/Right buttons let you step through various time divisions so you can change the roll on-the-fly.

DAW Control & Auto Mapping

The word Auto-Mapping means something different to me than it does to M-Audio. I absolutely assumed that the controls on the Oxygen 25 would map themselves automatically to whatever virtual instrument or effect plug-in I had loaded. Auto-Mapping has tended to refer to that overlay system that Novation had, or the Nektarine software system from Nektar, or I thought it might be along the lines of the AKAI VIP software that provides a way of saving mapping to instruments within a plug-in wrapper. Sadly not. In this case Auto-Mapping refers to good old Mackie/HUI transport and mixer mapping that we've been using for hundreds of years on all MIDI controllers that have ever possessed some sort of transport controls. I was genuinely intrigued and genuinely disappointed.

However, the Oxygen 25 has DAW control and they've done a decent job of making it all work within the confines of this MIDI controller while letting you map yourself manually to plug-ins at the same time. They make this happen by using a couple of individual MIDI drivers. You have one driver that's for your regular keyboard and MIDI control shenanigans and another that you set up the Mackie DAW control with. The Oxygen 25 has some internal DAW presets that ensure the smoothest possible control experience and you're good to go. There's a button to switch between DAW control and MIDI control (that they call 'Preset') and an LED comes on to show you when DAW control is active.

On the larger versions of this keyboard you've got a bank of faders for mixer control but on the Oxygen 25 you just get the one. This solitary fader will control the level of whatever channel you have selected or the master level depending on your DAW. Otherwise, the eight knobs step up to be the volume and pan on your first eight channels. You can jump to other banks of eight with the Preset Left/Right buttons. The knob functionality is selected by holding Shift and hitting the pad with the relevant choice written underneath. Holding the Shift key will tell you which is currently selected by illuminating the pad. Two of the options 'Device' and 'Sends' appear to be reserved for Ableton Live use only.

Round the back things are as simple as you'd expect, with just a USB port, a sustain pedal socket and a power on/off switch.

To make the mapping feel a little bit more 'Auto' you can edit the control numbers of all the knobs to match various instruments and save them as presets. Unfortunately, at the time of writing the Preset Editor software wasn't available but M-Audio have already set up a number of presets to work with the included virtual instruments.

Conclusion

The MkV is a decent upgrade to the solid Oxygen range of keyboards but does nothing to change the game. All the new features are great but they are already present on several alternative keyboards and even my workflow criticisms are the same on many other compact controllers that use the Shift + Keyboard functionality. There's certainly nothing wrong with it, the hardware is good, the feel of the keyboard and pads is great for this price and the physicality remains satisfyingly chunky. There's no MIDI output port but there is a power switch, which is a huge bonus in my book. So a decent job, well done.

summary

The Oxygen 25 is a solid choice for the less careful or less style-conscious mobile musician that's going to survive a hectic live performance while offering useful tools and easy control of your studio.

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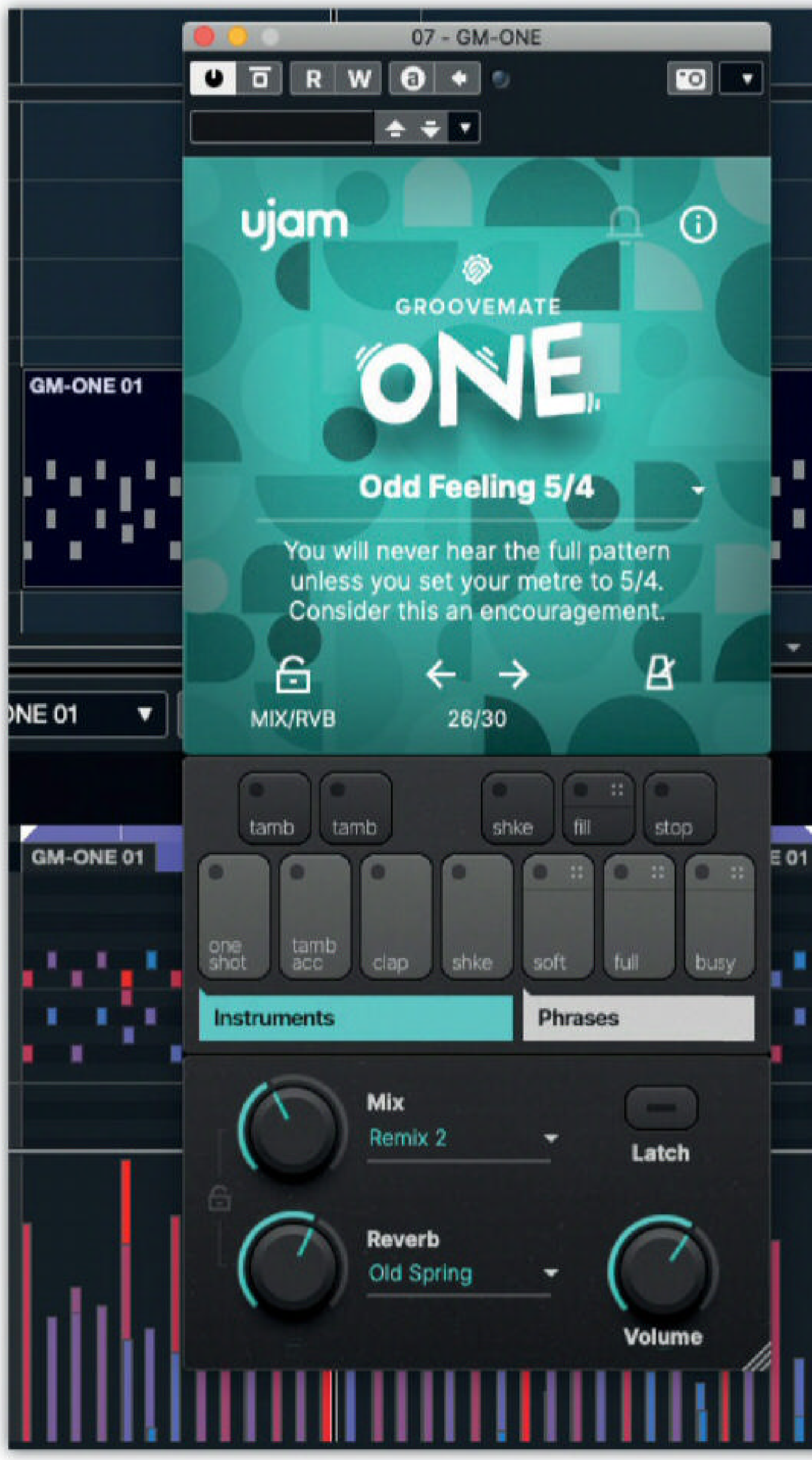
UJAM Groovemate ONE

Percussion Instrument

UJAM's Groovemate ONE is a one-stop shop for pop percussion.



JOHN WALDEN

UJAM specialise in virtual instruments and effects that sound excellent, are super-easy to use and are accessibly priced. It's a winning combination. However, the latest KISS offering is perhaps the cutest example yet; it's called Groovemate ONE. Priced at just $19, Groovemate ONE provides the essentials of pop-based percussion — shaker, tambourine and claps — in a compact format.

Easy Does It

The UI, while instantly familiar to anyone who has tried their virtual guitar, bass or drum instruments, is ultra-streamlined. You can start with any one of 30 pattern presets. For each of these, individual hits and three intensities of the pattern (plus a fill option) are mapped to MIDI keys starting at C3. You can trigger the pattern options to create an instant percussion part and add your own playing (with velocity response) from the individual hits. Two macro controls — Mix and Reverb — allow you to adjust the sounds of the percussion or control the ambience. These each feature a small set of presets offering a good range of options, while the knobs themselves can be used to dial in as much or as little of the effect as desired. The small metronome icon allows you to adjust the pattern triggering resolution and also to add swing, going from straight to swung feels with ease.

The presets include plenty of useful 4/4 patterns but also throw in some more esoteric grooves with 3/4, 5/4 and 6/8 all represented. Even so, pretty much everything here is going to just work in pop or other contemporary music styles and, in my own experimentation, Groovemate ONE quite happily slotted in beside a couple of UJAM's virtual drummers to add that little percussive lift a chorus section can often benefit from. And, if you like a preset pattern, you can drag and drop its MIDI data into your DAW for further manipulation. And as the sounds respond to MIDI velocity, that does provide a workaround for my one minor niggle with the streamlined UI; there is no mixer option to adjust the relative levels of each sound. Maybe that's something that UJAM might add in a future update?

Conclusion

That comment aside, Groovemate ONE is a bit of a gem. There are plenty of excellent upmarket virtual percussion instruments available, but if you just need some basic percussion sounds to add a little groove, and don't want to get sidetracked by too many options, Groovemate ONE is a no-brainer. Sat in a mix, the sounds themselves are perfectly adequate and the Mix and Reverb options provide enough sonic choice without ever being a distraction to your creative workflow.

UJAM have already indicated that ONE is the first product in a new Groovemate line. Here's hoping what follows is as useful as ONE and, mixer option aside, delivers the same brilliantly simple, totally usable and budget-friendly experience. I can only imagine that lots of potential users will find this incredibly useful. If that might be you, then the free 30-day trial is well worth downloading.

summary

Groovemate ONE makes adding simple percussion grooves about as easy as it will ever get. A genuinely useful tool that won't break the bank.

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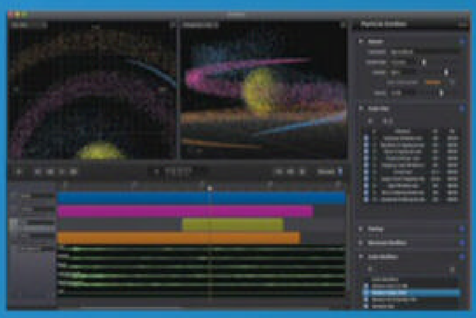

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HEDD Type 07 Mk2

PHIL WARD

Active Monitors

HEDD's impressive Mk2-series speakers let you choose between ported and unported formats. Which works best?

My previous experience of HEDD monitors was the three-way Type 20 Mk1, which I reviewed back in the March 2018 issue of this magazine. I admired the Type 20 greatly, not only for its very high level of both objective and subjective performance, but also for its innovative 'Lineariser' plug-in approach to correcting the time-domain errors inherent to all electro-acoustic transducers. The Lineariser seemed, to my ears, to add an extra polish and finesse to the Type 20's performance, however its plug-in format meant that it was only accessible when the monitors were being driven from a suitable DAW session. Now, with the Mk2 versions of their monitor range, HEDD have fixed that issue by incorporating the technology within the speakers' on-board DSP. But that's not all. The Mk2 versions of the HEDD monitor range introduce quite a few refinements, improvements and new ideas. Read on and I'll tell all.

Knocking On Seven's Door

Rather than simply looking at the Mk2 version of the Type 20, I decided for this review to step one rung down in the HEDD product range and examine the two-way Type 07. Before I get on to the specifics, though, a little refresher regarding the company. HEDD is an acronym for Heinz Electro-Dynamic Designs, and the Heinz in question is Klaus Heinz, who previously played the role of R&D director at ADAM Audio. Heinz, and his mastering engineer and musicologist son Frederik Knop, founded HEDD in Berlin in 2015 specifically to offer a new range of technologically advanced monitoring and headphone options for pro audio applications. The full HEDD range now includes four monitor models ranging from compact nearfield to large-scale midfield; an ambitious main monitor 'tower' system; two subwoofer products; and a headphone. Our editor, Sam Inglis, wrote in glowing terms about the HEDDphones back in the July 2020 issue.

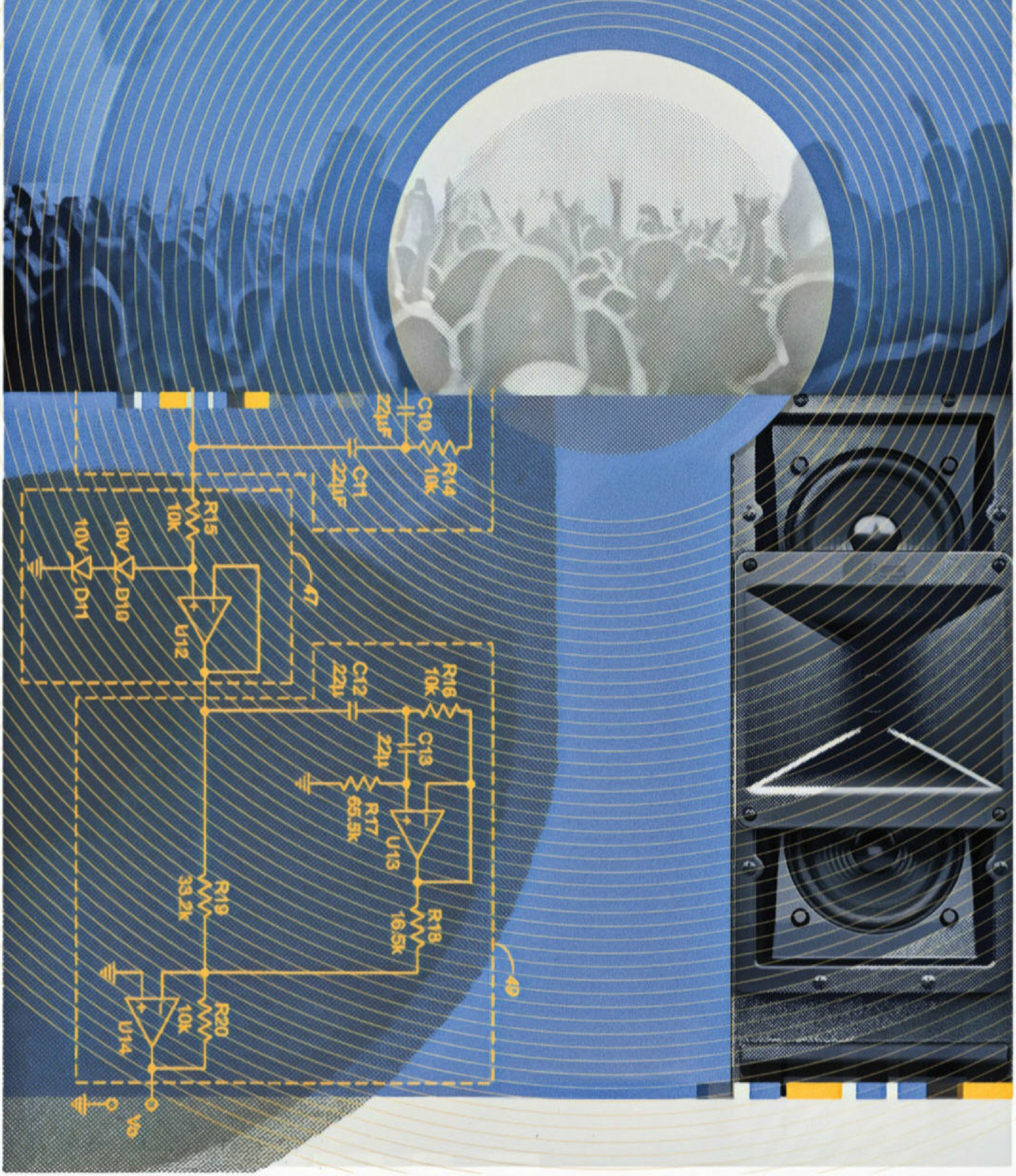
Back to the Type 07 Mk2, at first sight it appears to be a relatively conventional two-way ported active nearfield monitor, with perhaps the only obviously notable »

HEDD Type 07 Mk2
$1798

PROS
- Exceptionally high fundamental electro-acoustic performance.
- Adaptability through EQ and port options.
- Lineariser option.

CONS
- None.

SUMMARY
The HEDD Type 07 Mk2 is a seriously high-performance nearfield monitor that incorporates some genuinely useful and interesting extras. Apart from the other models in the Type Mk2 range, there's really nothing else quite like it.



feature being its use of a Heil AMT-type folded ribbon tweeter (monitor geeks might notice that use of an AMT-type ribbon tweeter is customary in ADAM monitors; this I guess is no coincidence considering Klaus Heinz's previous employ). The Type 07 Mk2 is a convenient size for nearfield installation, and although relatively heavy its weight shouldn't cause any sleepless nights. The MDF cabinet is finished in a silk/matte black paint. Beneath the ribbon tweeter is a nominally 18cm-diameter composite-diaphragm bass/midrange driver, and beneath that live a couple of decently flared reflex ports. It's with the ports that the unusual stuff begins, because the Type 07 Mk2 actually offers ported (reflex loaded) and non-ported (closed-box) operational modes.

There's two aspects to converting the Type 07 Mk2 to and from reflex loaded to closed: the acoustic and the electro-acoustic. The acoustic element involves the physical insertion of plugs, supplied with the monitors, in the ports to seal the enclosures. The plugs are short metal cylinders fitted with foam peripheral sections that provide a comfortable yet tight interference fit in the port diameter. On one face of each metal cylinder is a tapped hole that fits a small, machined screw tool supplied with the monitors. Screwing the tool into a cylinder enables its removal from the port. I can't help thinking HEDD would be well advised to have a tapped hole on both faces of the cylinders because, as things stand, I'm not sure how easy it would be to remove a plug if it were inserted the wrong way. The reason I'm not sure is that I didn't make that mistake, but it was an extremely close run thing.

Before I move on to the electro-acoustic aspect, a bit of background. It is perfectly feasible with a passive monitor (that can't incorporate DSP or even conventional analogue EQ) to contrive a low-frequency system that works in both ported and closed-box modes. In fact, quite a few hi-fi speakers ship with optional foam port bungs. You just have to engineer the various electro-mechanical parameters of the bass driver to produce a workable low-frequency and time-domain response with a fixed enclosure volume for the closed option and a fixed combination of enclosure volume and port tuning frequency for the ported option. Such arrangements can work, however they'll always involve a hint of frequency response compromise — neither ported nor closed modes is likely to be optimal. But with a DSP-enabled active monitor, where the opportunities for equalising in both time and frequency domains are almost unlimited, it's perfectly possible to engineer a speaker that can operate equally optimally both ported and closed (actually it's not quite so clean cut as that; see the 'Ports & Stuff' box). So when the Type 07 Mk2 morphs from ported to closed (or the other way around) through the addition (or removal) of its port plugs, a switch on the rear panel also needs to be moved to select a ported or closed EQ profile. If the optional Lineariser is also engaged, selecting the ported or closed option will also implement alternative time-domain compensation profiles, because not only does adding or removing a port change a monitor's frequency response, it'll change its group delay (frequency-dependent latency) too.

The Type 07 Mk2 can accept either balanced analogue or digital (AES3) signals. All processing — including the crossover, shelving filters, desk EQ options, and the Lineariser — is performed using the speaker's internal DSP.

Reflex Reaction

So that's the nuts and bolts of the ported or closed configurations of the Type 07 Mk2. But, I hear you ask, why would you want the option? The ported versus closed-box question has been mulled over many times in the pages of *Sound On Sound* and, believe me, if you were to rank electro-acoustic topics in terms of hours spent on discussion during a speaker design career, the topic would probably come top. To recap: the point of a reflex port is to extend the bandwidth and/or increase the maximum volume level of a moving-coil speaker by employing the otherwise unemployed acoustic energy that radiates from the rear of the bass driver diaphragm. A port does this by using a tuned resonance to reverse the phase of that rear radiation over a narrow range of frequencies so that it reinforces, rather than cancels, the forward radiation of the driver. There's no free lunch though, so port loading brings with it some snags that many who give this kind of thing a second thought believe are potentially damaging. Firstly, reversing the phase of the driver's rear radiation unavoidably implies a time delay. If, for example, a port is tuned to 40Hz, one complete cycle takes 25ms, so a half cycle flip (180 degrees) implies a delay of 12.5ms. All other things being equal then, when the speaker plays 40Hz, the output will be delayed by a minimum of 12.5ms. Secondly, one fundamental characteristic of a resonant system, such as a reflex port, is that not only does it take a finite time to get going, it takes a finite time to stop too. So back to our 40Hz port, if the driving signal stops suddenly, the output won't. And the higher the Q of the port resonance (and much of the point of a port is lost if you don't maximise its Q), the longer it will take to stop. It's not unusual to find output continuing for multiple cycles after a stimulus signal has ceased. At 40Hz, for example, that could be as much as, say, 75ms. The final

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~ TIM HANDS
(ADR ENGINEER, 4 EMMYS WITH 9 NOMINATIONS FOR GAME OF THRONES & THE LIFE AND DEATH OF PETER SELLERS)

" I hear the tail on the reverb now... "

~ PERCY BADY
(WHITNEY HOUSTON, DIONNE WARWICK, YOLANDA ADAMS...)

" Jones-Scanlon Studio Monitors allow me to hear the essence of the sound at its truest form without color "

~ ALEXIS KING FAKU
(UNIVERSAL MUSIC STUDIOS SOUTH AFRICA)

" I knew I had to have them, the Jones-Scanlon Studio Monitors are hands down my new favorite monitors "

~ KRISH SHARMA
(2 GRAMMYS, ROLLING STONES' RECORDING ENGINEER)

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Ports & Stuff

There's one compromise in optionally ported and closed two-way monitors that can't be managed with DSP, and it arises because each type of monitor ideally demands different quantities and types of internal damping in the enclosure. In a ported enclosure it's important not to stuff the bulk of the cabinet with wadding, because doing so will significantly decrease the reflex port's Q and damp the very resonance it is installed to exploit. So the usual strategy is to line only the internal enclosure surfaces with acoustic foam. In a closed-box enclosure, however, there's no port resonance to worry about so the entire volume of the enclosure can be lightly stuffed with wadding (usually polyester but sometimes natural wool), often to the benefit of reduced midrange coloration. Now, you may have noticed that I qualified this whole discourse by specifying two-way monitors in the first sentence. That's because three-way monitors dodge this particular compromise: the midrange driver enclosure can be bulk filled with wadding to more effectively dissipate the driver's rear radiation and minimise coloration, while the separate bass driver enclosure can be internal surface-lined so that a port, if one is used, isn't compromised.

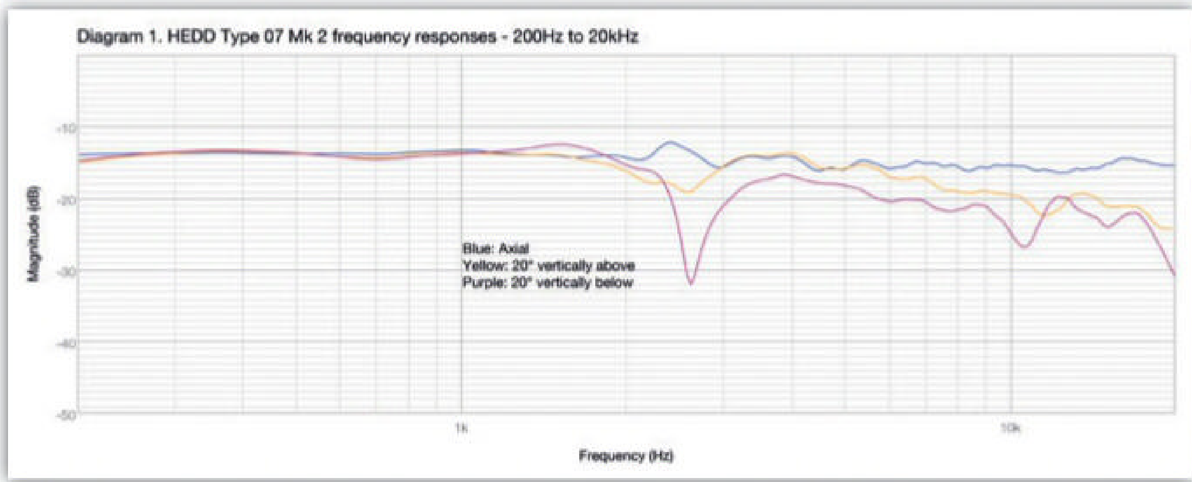

Diagram 1: The Type 07 Mk2's frequency response, measured on-axis (blue trace), 20 degrees above axis (yellow), and 20 degrees below (purple).

» snag introduced by port loading is that port performance is highly sensitive to its mechanical architecture. The Q of the port resonance will only remain suitably high if the airflow in the port remains laminar. If the airflow becomes turbulent (and that's jointly a function of its velocity, the port diameter and length, and degree of exit and entrance flaring), the Q will collapse and, rather than reinforcing low-frequency output, the port will behave more like a leak that contributes nothing but compression and noise. Ports are also prone to 'organ-pipe' resonance, where they effectively become a wind instrument in their own right and, driven by the upper-bass and lower-midrange acoustic energy bouncing around inside the enclosure, generate a high-Q pipe tone that can make an audible contribution to a monitor's midrange.

The supplied port-blocking bungs.

Having read that paragraph of reflex port demolition, you're probably wondering why any monitor designer would ever bother with a port. There are two answers. Firstly, the benefits that ports offer in terms of low-frequency bandwidth and volume level, especially when a monitor has to hit a manufacturing budget that can't afford the extra bass driver costs that a closed-box format of equal bandwidth and volume performance would demand, are too great to ignore. A hole and a plastic tube is a lot less expensive than, say, a bass driver magnet that's twice the size. And secondly, skilful and knowledgeable design can undoubtedly minimise the problems that ports introduce and render them acceptably benign.

Before I start measuring and listening, I'll finish the job of describing the rest of the switches. Firstly, the Type 07 Mk2 is unusual in offering both input sensitivity and gain options. Both are on detented knobs so there's no problem in ensuring that the two monitors of a pair are matched. Having both sensitivity and gain control leaves no stone unturned in the search for optimising monitoring gain structure, but considering how DAW and interface output levels all tend to be very similar these days, I'm not sure how much genuine use it will be to the majority of users. Having said that, I did nudge the input sensitivity up a notch in my system.

Along with input sensitivity and gain, the Type 07 Mk2 includes low- and high-shelving EQ options that offer ±4dB of adjustment in 1dB steps at 200Hz and 3kHz respectively; low-frequency bandwidth options that offer extended, normal and subwoofer (80Hz high-pass filter) settings, and three desk EQ presets. I'll investigate the desk EQ options a little further down the page.

While I'm still on the rear panel, as far as inputs are concerned the Type 07 Mk2 offers balanced analogue and AES3 connections. Internally the Type 07 Mk2 signal flow is digital, running at 32-bit/96kHz, and at the downstream end of that signal flow, the Type 07 Mk2's Class-D amplification is rated at 100 Watts for both the bass/mid driver and tweeter. The crossover frequency sits at a conventional 2.3kHz.

Measuring Up

And so to the promised FuzzMeasure data. Diagram 1 shows the Type 07 Mk2's forward axial frequency response from 200Hz to 20kHz, along with measurements taken 20 degrees above and below. The axial response is suitably flat and tidy, and the vertical dispersion is also well managed. The sharp suck-out at around 2.5kHz on the lower axis curve (purple) is typical of vertically arranged two-way speaker systems, and is caused by destructive interference between the two drivers in the crossover overlap band. A vertical off-axis crossover suck-out is all but unavoidable in two-way speakers, however designers do have some freedom (through choices in crossover filter design) to decide in detail how it is managed and whether it is better engineered into the above- or below-axis frequency response. The usual and sensible decision — which the Type 07 Mk2 clearly adheres to — is that it's better not to compromise the above-axis response, as a below axis listening position is less likely. Back to the Diagram 1 off-axis curves, the gentle off-axis droop at high frequencies is typical of ribbon drivers, however it is relatively mild on the Type 07 Mk2 and, especially in the context of nearfield listening, unlikely to be a major issue.

The impulse response plots of Diagram 2 illustrate the result of engaging and »

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ALTERNATIVES

The Type 07 Mk2 finds itself in a competitive niche where there are some very capable alternatives. For example, the **Focal Shape Twin**, **Genelec 8040A**, **Reproducer Audio Epic 5** and **Dynaudio Lyd 8** can all be had for around the same price.

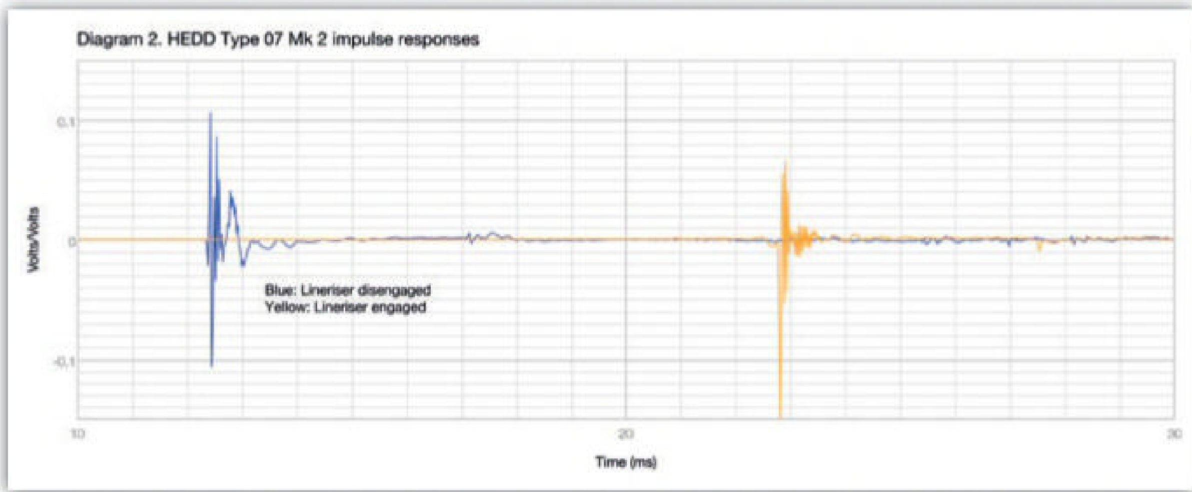

Diagram 2: The Type 07 Mk2's impulse response, with the Lineariser processing off and on (blue and yellow traces, respectively).

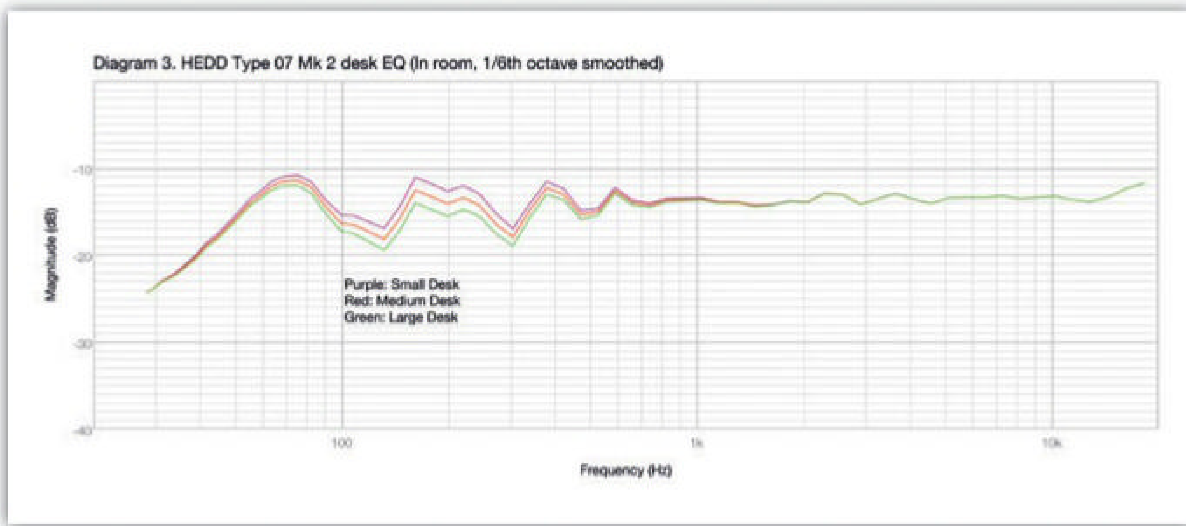

Diagram 3: The desk EQ options for a small, medium and large desk, in purple, red and green respectively.

» disengaging the Lineariser processing. The most obvious result is that the Lineariser adds around 10.5ms of extra latency to the system, but it also clearly tightens the impulse response shape, making it more compact and, well, more impulsive. The 12.5ms latency of the non-Linearised impulse, by the way, reflects that of my measurement signal chain and the signal 'flight time' from monitor to measuring mic.

Finally, Diagram 3 illustrates the action of the Type 07 Mk2's desk EQ feature. The desk EQ introduces a gentle suck-out centred on 180Hz that increases in depth from a subtle 1dB or so to a more significant 4dB. I've found similar desk EQ options quite useful in the past and in combination with the shelf EQ options there really ought to be few listening spaces that the Type 07 Mk2 can't be adapted to suit. And that brings me to one of the significant positives of the Type 07 Mk2. The multiple EQ options, ported or closed format, and linear or 'natural' time-domain response mean a huge variety of possibilities for adapting the monitoring character to meet the needs of personal preference, room acoustics and programme material. The Type 07 Mk2 is a genuinely impressive proposition in that respect.

Listening In

Monitoring adaptability would of course mean nothing without great fundamental performance, but the Type 07 Mk2 has that too. I remember being immensely impressed with the core electro-acoustic character of the original Type 20 in terms of its tonal balance, imaging and detail portrayal, and lack of coloration. The Type 07 Mk2 pulls off the same trick in that there's something immediately satisfying about the way it presents material: everything is fundamentally in the right place, with the appropriate tonal character, image focus, pitch and dynamics. Vocals in particular sound properly focused and convincing with a 'high-end monitoring' level of quality. For example, I found myself listening to Jesca Hoop's, 'The Lost Sky', a track I know well (and love), on repeat and becoming fascinated anew by how it's put together in mix and arrangement terms, while simultaneously being completely seduced by the vocal performance.

For what it's worth (because your mileage will vary), the EQ options I settled on in my studio space were 1dB of desk EQ combined with 2dB of LF shelving attenuation and 1dB of HF shelving attenuation. But I guess the elephant questions in the room are, "ported or closed" and "Linearised or natural"? My preference on the second of those questions was for Linearised. As with my previous experience of the Type 20, the Lineariser seemed to add an extra degree of image focus and precision. Mix elements appear to live somehow more separately in their own space. It's subtle, but once you become tuned into the effect, I think it's worthwhile.

The question of ported or closed is a more complex one. The subjective difference between the two options is significant, and my preference was generally for the closed option — if forced to choose I've always preferred precision and accuracy of bass over bandwidth. Monitors are tools of a trade, however, and personal preference is an unaffordable luxury if it impedes getting a job done. So in use there will be times when the extra low-frequency bandwidth and maximum volume level offered by the ported option will be of crucial importance because, on a very basic level, it could render audible some mix elements that the restrictions of the closed-box option cause to remain hidden. And even then, it's not as if the quality of bass that the Type 07 Mk2 is capable of in ported mode isn't extremely high, because it is.

My time with the HEDD Type 07 Mk2 was characterised by increasing respect. To begin with I felt it was predominantly a very well-engineered but conventional two-way nearfield monitor. But as I used the Type 07 Mk2 more and experimented with its many EQ and format options, my admiration grew significantly. And now, at the end of the review period, I'll be sorry to see the back of them. As is my usual practice, when I get towards the end of a monitor review, I do a bit of research to remind myself of the price, and when I did that with the Type 07 Mk2 I had to check I wasn't looking at the cost of a single monitor rather than a pair, because I'd imagined it to be significantly more expensive than it is in reality. So bearing in mind just how much I came to admire the Type 07 Mk2, I think it's a steal.

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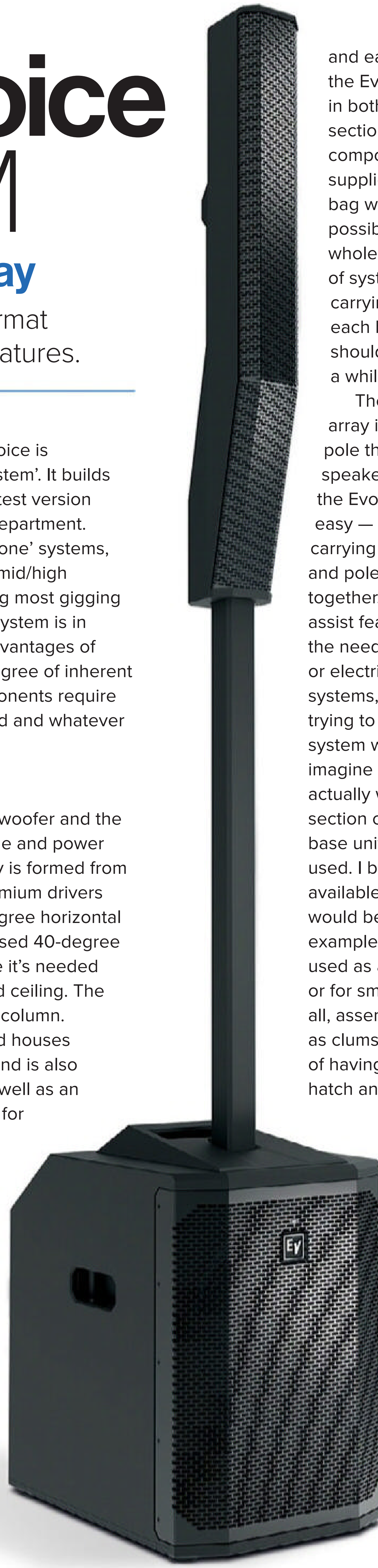
Electro-Voice Evolve 50M

Compact Active Line Array

EV's latest rig takes the column PA format and adds a host of thoughtful extra features.

MIKE CROFTS



The Evolve 50M from respected makers Electro-Voice is described as a 'compact column loudspeaker system'. It builds upon an earlier model (the Evolve 50), but this latest version incorporates a few new tricks in the mixer and control department. Various manufacturers produce outwardly similar 'all-in-one' systems, and the convenience of having a single unit containing mid/high speakers, subwoofer, power amp and mixer is something most gigging performers appreciate. The top section of the speaker system is in the form of a vertical array, and that brings with it the advantages of compact dimensions, very wide coverage and a high degree of inherent resistance to acoustic feedback. The Evolve 50M components require no interconnecting cables — all you need is a mains lead and whatever is being plugged into the input stages.

The Parts

The Evolve 50M is made up of two active parts, the subwoofer and the vertical speaker array; the sub contains the mixer module and power amp stages for both speaker sections. The column array is formed from a composite material and contains eight 3.5-inch neodymium drivers mounted on waveguides that result in very wide 120-degree horizontal coverage, and the curve helps it to deliver a tightly focused 40-degree vertical coverage, which aims the acoustic output where it's needed (the audience) while not wasting energy on the floor and ceiling. The overall tuning is assisted by four ports at the rear of the column.

The subwoofer section is made from 15mm wood and houses a single 12-inch driver mounted in a vented enclosure, and is also home to a Class-D amplifier module rated at 1000W, as well as an eight-channel programmable digital mixer that provides for various combinations of mics, line inputs, stereo inputs, high-impedance and Bluetooth sources — we'll take a brief look at some of these later, but all the fine detail is available on EV's product page (better still, download the user manual and read at your leisure).

Both speaker sections are well-balanced and easy to lift and carry, and the array module has an integral handle that's large enough for both horizontal carrying and vertical placement or removal. Black steel grilles finish the package off nicely and should protect the speaker components against most pre-apocalyptic hazards.

The Setup

No matter how good they sound, compact all-in-one systems are a bit off the mark unless they are neat and unobtrusive when in use, and quick and easy to assemble and pack away; the Evolve 50M scores very highly in both these categories. The two sections are designed to travel as two components and the speaker array is supplied in a neat, high-quality carrying bag with a shoulder strap, so it's quite possible for one person to carry the whole system. EV supplied a pair of systems for this review, so I tried carrying both by myself, with a sub in each hand and a 'top' bag over each shoulder. It is indeed possible... for a while!

The carry bag houses the speaker array itself and a spacing section or pole that sits between sub and top speakers. Putting the three pieces of the Evolve 50M together is extremely easy — it takes longer to unzip the carrying case and remove the array and pole than it does to connect them together. I particularly like the magnetic assist feature, mainly because it obviated the need for any heavy-duty mechanical or electrical coupling which, on some systems, can present problems when trying to pull the pieces apart. The EV system works really well and it's hard to imagine an easier method that would actually work. As far as I can tell, the array section can't be coupled directly to the base unit, so the spacer pole has to be used. I believe that a shorter section is available as an optional extra, and this would be useful in some applications, for example if the Evolve 50M were being used as a close-up monitor for say, piano, or for small-group AV applications. All in all, assembling the Evolve 50M is about as clumsy-proof and easy as it gets, short of having a built-in robot pop out of a little hatch and do it for you.

The Mixer

The mixer and control section of the Evolve 50M fits neatly into the back panel of the subwoofer and contains everything you'd need for a modest combo performing live. The mixer has eight inputs: four mono mic/line channels plus two further stereo channels, making up the eight. Channels 1 to 4 have standard balanced XLR or TRS inputs with phantom power permanently enabled, and these have sensing and protection features to detect

when anything not requiring power is plugged in. Channels 5 and 6 operate as a stereo pair with a choice of balanced XLR, TRS, unbalanced RCA and even stereo 3.5mm mini-jack input connectors, so all common analogue devices should be well covered. Inputs 7 and 8 are reserved for Bluetooth and are accessed by any compatible paired device. This type of input can be really handy in any application using backing tracks, as playback can be controlled by someone off stage, a feature I've found useful at numerous school events where the teacher can be given the job of starting and stopping the tracks without the hassle of trailing cables.

As if this wasn't enough, channel 4 has an additional physical input in the form of a high-impedance TS jack intended for direct connection of instruments, without the need to use a separate DI box. There's also a guitarist-friendly footswitch jack for switching the user-selectable DSP functions.

The Evolve 50M mixer surface, which bears the Dynacord name and logo as testament to the design lineage, is clearly and cleanly laid out to present a simple and functionally obvious control panel. There is only one rotary control, whose default function is master volume, but this also functions as an adjuster/encoder for other functions. Each channel's parameters are accessed via a row of buttons above the input connectors, and whichever channel is selected can be adjusted at either a basic or more advanced level. There are several pages of mixer functions in the manual, and these describe in detail what can be adjusted — suffice it to say here that this is indeed a full-function digital mixer, and can of course be controlled either on the panel or, with a lot more facility, via EV's QuickSmart app. I found both the panel controls and the app to be easy to use, and adjustments are quick and smooth. Many users of this system will be looking after their own sound balance from the stage, so this is an important aspect of a system such as this.

All the expected complement of dynamics processors, detailed EQ, two full effects engines and so on are included, and I think that the mixer section alone would probably find an appreciative market if it were available as a standalone item. The best way of controlling the adjustable parameters is by means of the QuickSmart app, but the little LCD display does a good job of summarising the main mixer functions and shows a neat all-channel meter view so you can see what's going on with your input levels with a quick glance at the rear of the unit. Just as assembling the Evolve 50M is impossible to get wrong, the built-in protection circuits should keep all but the relentlessly determined user within the unit's performance boundaries.

The subwoofer unit also houses a digital mixer, with eight analogue inputs plus a stereo Bluetooth channel.

The Great Link

One major new feature built into the Evolve 50M is the ability to connect two systems together. A few years back I remember getting my hands on a portable column-over-sub system when the idea was fairly new, and the first and (to me) obvious question that came to mind was 'Could you use two to make a nice little stereo array-type PA system?'

Of course that's always been possible to an extent, but what EV have done with the Evolve 50M is, firstly, make a system that truly does produce a respectab e output level for live gigs, and secondly, make pairing two systems as a left-right PA really easy. Not only that, but when two systems are linked together using the QuickSmart digital link, they are automatically configured as you desire (for example as left and right mains, if that's what you fancy), and the two mixer modules act together to provide twice the input count, a l properly integrated into a single mix. In short, a pair of Evolve 50Ms can act as a front-of-house stereo rig, with a 16-input digital mixer, and two auxiliary mix outputs into the bargain, all with one single Cat-5 cable running between the two.

Linking two systems is simply a matter of connecting them together with such a cable plugged into the RJ45 sockets labelled 'QuickSmart Link'. Although these look like standard network ports they don't work that way — the interconnection protocol is very much EV's own and they are dedicated to the Link function. With two systems hooked up you can then, as mentioned, make use of all 16 inputs, and control both units as a single sound system from one instance of the QuickSmart app, although »

Electro-Voice Evolve 50M

$1999

PROS

- Powerful performance in a small and easy-to-use package.
- Lots of onboard DSP and good control from the app.
- Easy linking between systems with the ability to combine mixer inputs.
- Can be used without the app if necessary.

CONS

- Nothing significant, though I would like to see level meters alongside the app on-screen faders.
- Unless I've missed it, I'd like a 'skip firmware update until next time' option.

SUMMARY

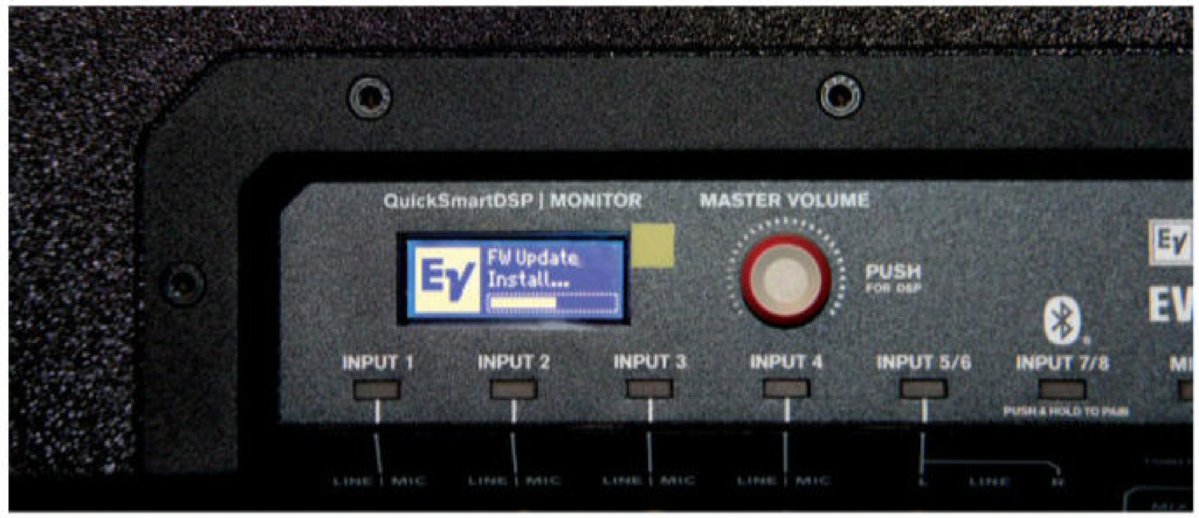
A portable yet powerful and high-quality column system that boasts some useful extras, including the ability to run two as a pair with all their mixer inputs combined.

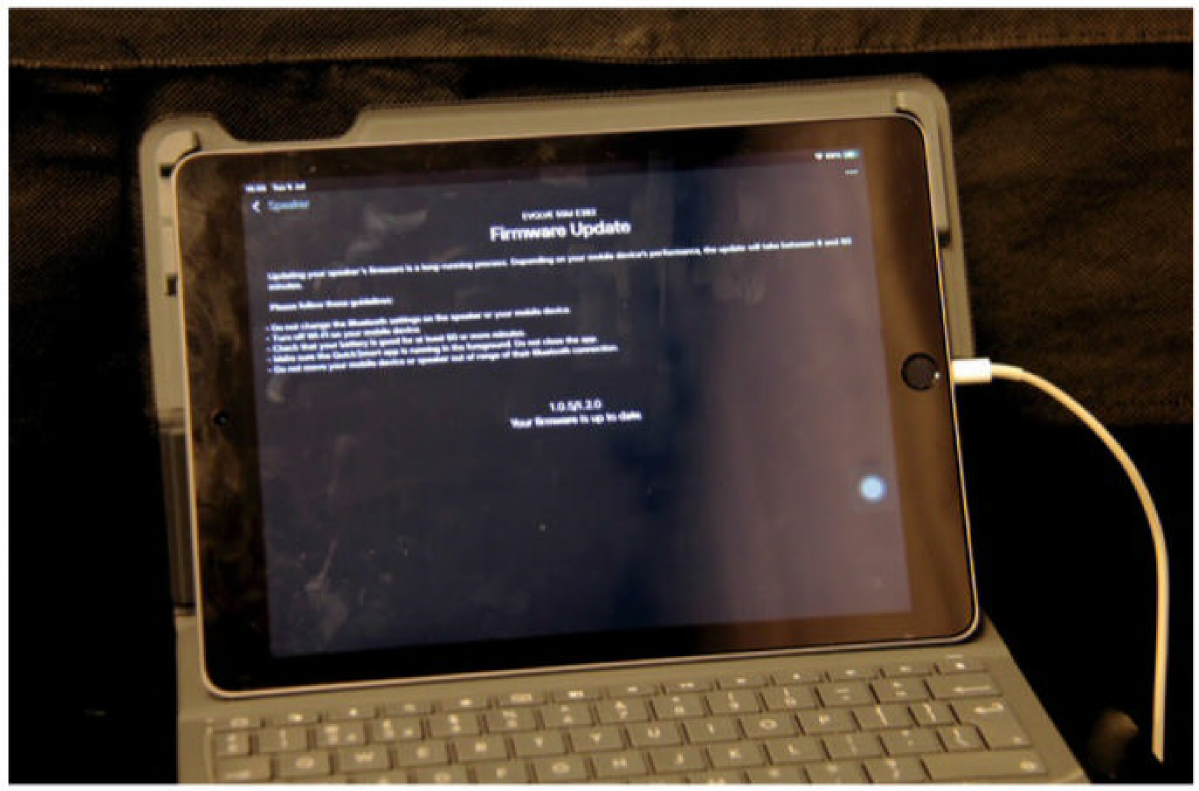
you do have to identify which system you're accessing — I didn't find a way of swiping across both sets of inputs within a single layer view, but the app layout is very clear and free of screen clutter, and it's obvious which set of inputs are in focus at any time. When the Cat-5 cable is connected, the unit will automatically configure itself depending on your chosen mode, and will either send to or receive control parameters from the other Evolve unit, presenting options for running the two units together in mono or stereo, as well as offering various send or receive parameters. I set the two demo systems up with 'house left' designated as 'stereo left', as I would normally mix with the app from a front-of-house position if not actually playing on stage. Once linked, all control parameters are duplicated across to the 'receive' unit and most functions will then be linked, although individual master volume levels can be controlled on a per-speaker basis for room balance.

When the control device (phone or tablet) is connected via Bluetooth to any Evolve system, an automatic firmware update check is carried out, and if an update is available it must be installed to bring the system up to date before Bluetooth control can be established. The manual says that this can take up to 50 minutes (!) and advises against doing this just before a performance! When I first got the devices and systems paired up the app informed me that a firmware update was indeed required, however it was all done and fully operational in just over four minutes. When control is acquired everything apart from physically plugging into the unit can be done from the app, and it's as well to check that the input and/or master levels are down before applying audio as the Evolve 50M can generate a respectable output level — a lot more than you'd expect from its modest size.

The Sound

So, what about the sound from this very portable rig? It's crisp and clear and, as you'd expect from a vertical array like this, horizontal coverage is wide and even. In my studio live room test I had to stand almost directly to the side of the array before any significant lack of top end became noticeable. I had initially only rigged up one of the Evolve systems, and fed it with a selection of mono test tracks just to get used to the sound and then find out where it would run out of steam. As I said, I have used these types of mini-array speakers in the past on many corporate gigs and band rehearsals, and they're perfectly fine so long as you don't need to go too loud, but these EVs have been crammed with lots of power and really do deliver a respectable output, all with the trademark full, rounded sound we expect from this maker. The spec sheet gives the maximum output as 127dB SPL at 1 metre, and the quality and focus is very impressive.





Although the manual warns that firmware updates can take close to an hour, in practice the process only lasts a few minutes.

When I added the second system and ran them as a stereo rig, I was convinced that I'd easily have enough level to run an average band gig in a club or pub. Unfortunately because of the remaining Covid restrictions at the time of testing I wasn't able to find a live event for a proper road test, but I did take the two Evolve 50M units outside and run them at almost full chat for a while with a wide range of recorded material. The mids and top end carried well (over a mixture of hard standing and grass), and the balance stayed fairly constant over a reasonable distance. Even the two 12-inch subs managed to keep up and pushed out a decent level, certainly more than I'd have expected given their size and weight, and the sub output level can be controlled independently for achieving the best sound balance. The only way to test PA systems is to use them for real, and I would have been very happy to take these along to a smaller-scale gig and run a live band through them, but for the special circumstances.

The Evolve 50M has far too many settings, features and user options to go into here, but I did experiment with the overall system EQ presets (the 'Mode' setting), which offer a choice between Live, Speech, Club and the default Music mode. No prizes for guessing roughly what these do, but my personal default mode would always be Live as this is the 'neutral' setting.

Practical PA

You would expect any product of this type, brand and price to be well designed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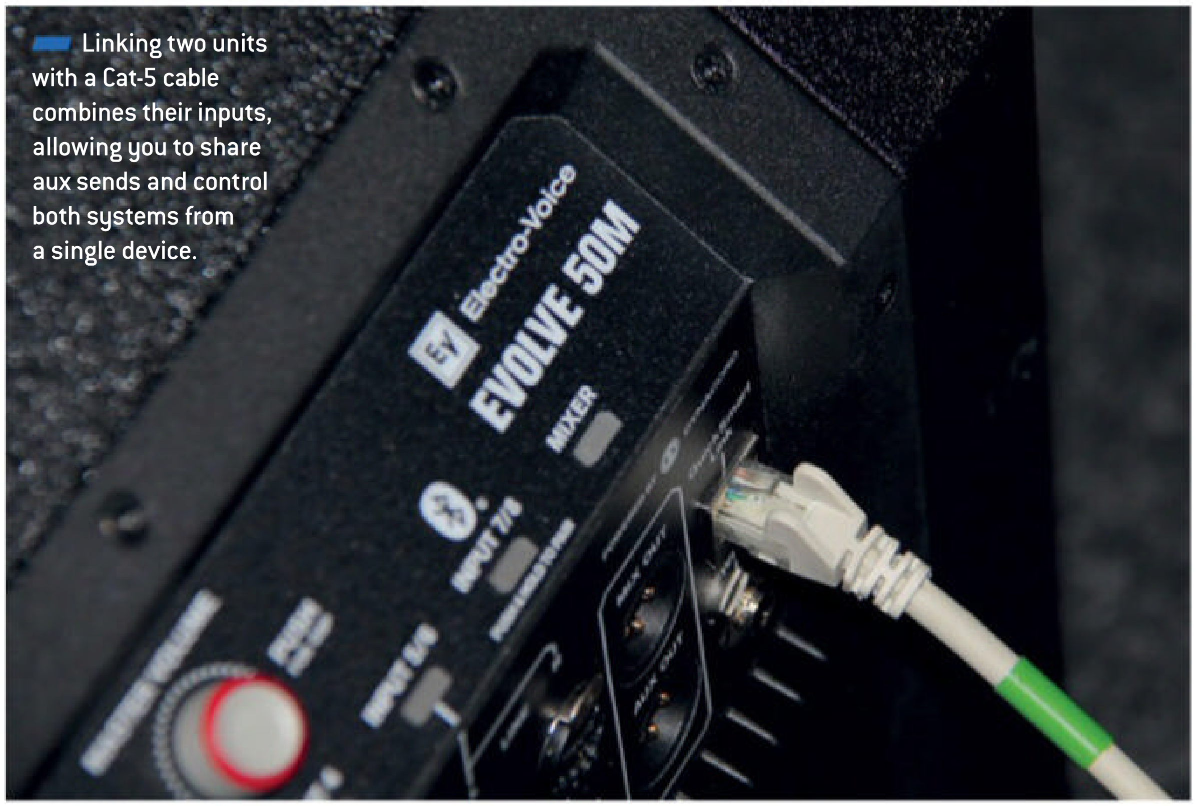
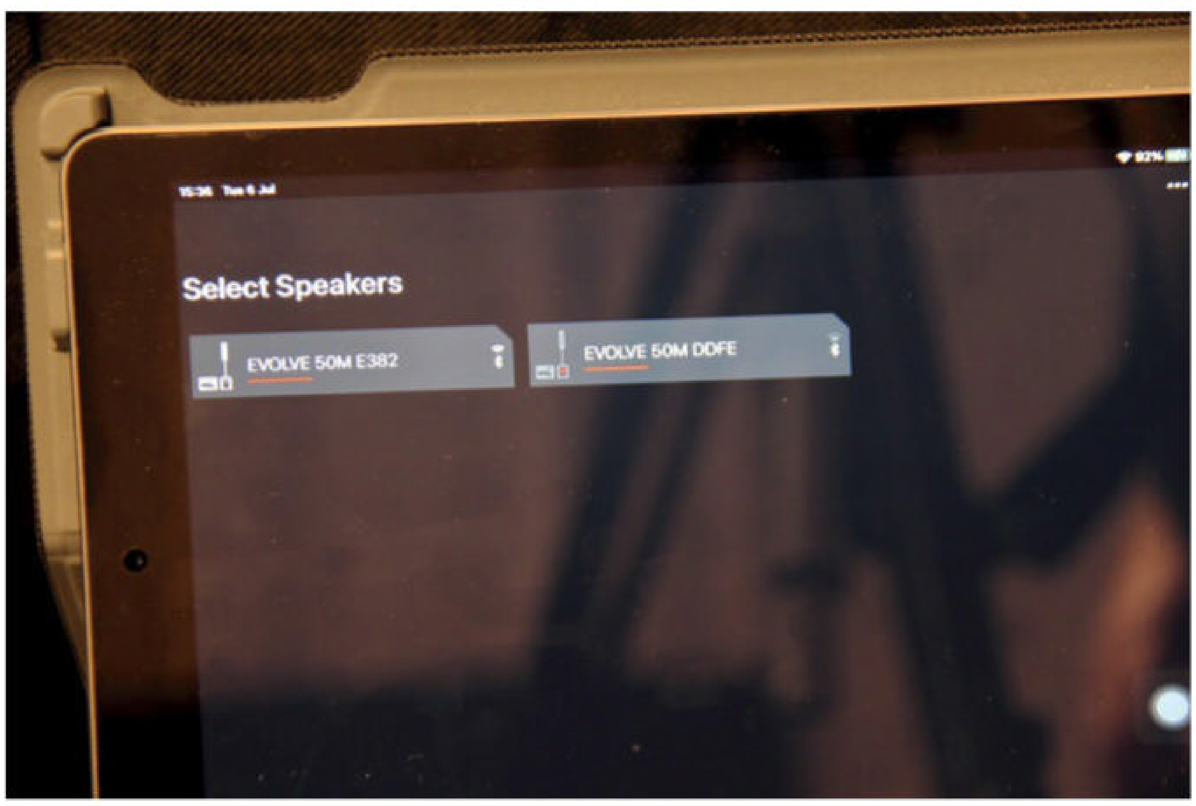
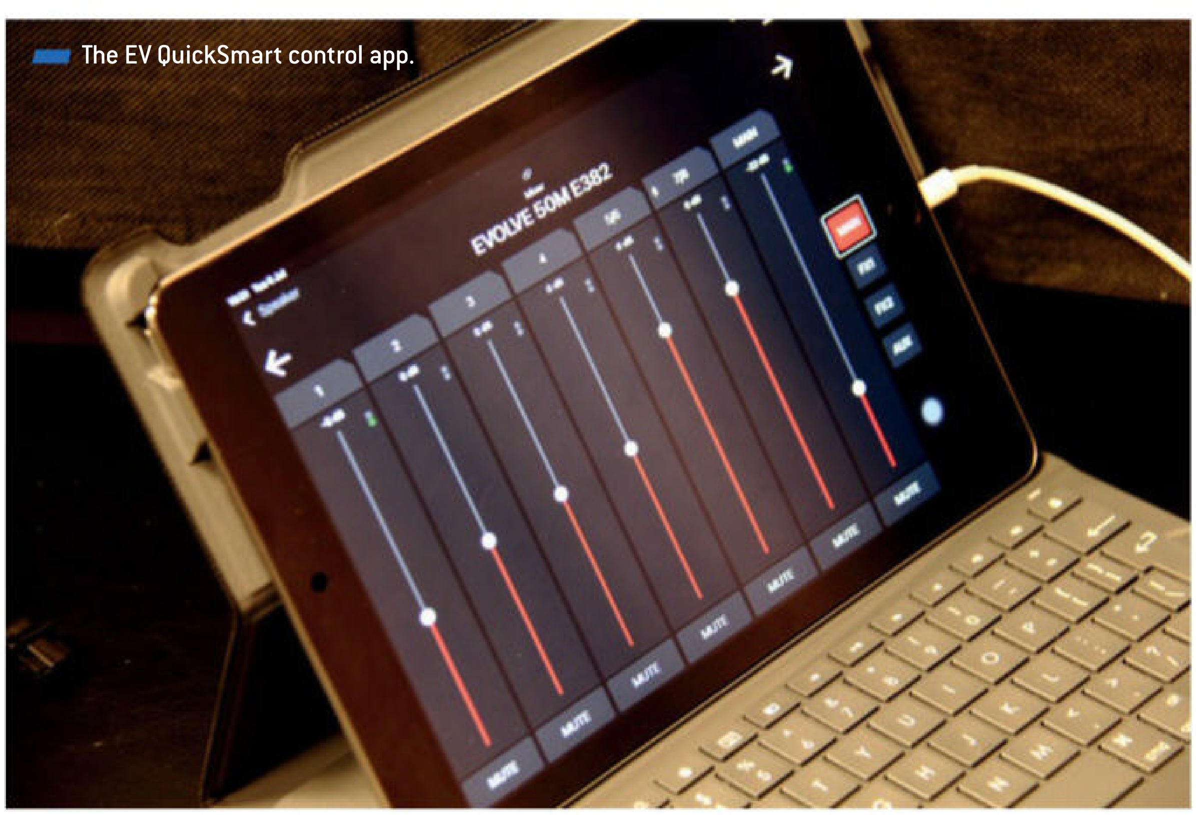


Linking two units with a Cat-5 cable combines their inputs, allowing you to share aux sends and control both systems from a single device.





The EV QuickSmart control app.

ALTERNATIVES

In terms of quality and price, **HK Audio**'s **Polar** range probably comes closest, but while those systems do include digital mixing facilities, they lack the Evolve 50M's linking ability when used in pairs.

» and to sound great for its intended purpose, which it does, but often it's the little practical things that can make the difference. I didn't find any aspect of the Evolve 50M impractical, annoying, or disappointing — this is a system that delivers all that is promised. In use, it is easy to use and will keep producing output if app control is lost for any reason. If two units are SmartLinked and the 'master' system loses power, the 'downstream' one will be muted, and I found that when power is restored the second system remained muted until re-linked. They can of course be linked with an XLR cable if you don't trust these new-fashioned digital things, but the mix output is still an active connection as it's under the control of the first system's mix settings. When powering on and off I could not produce any thumps or pops from the speakers even by pulling the mains plugs straight out — these are well-behaved speakers.

Final Thoughts

Time for a quick summary, then. There is truly a lot to say about the Evolve 50M, and I have only touched on the standout aspects as I saw and heard them, however I believe this neat little system has finally stepped up to the 'real band PA' level, and offers many of the advantages of a line array in a compact, portable and virtually user-proof package. It is surprisingly loud for such a small set of boxes, flexible in that you can use one or two depending on the application (or indeed use the Evolve 50M together with other systems), and provides an all-in-one sound solution that can reduce extra gear and cabling whilst still offering all the bells and gizmos. As a single 'stick PA' it has that 'fill the room' quality, while as a pair at front of house it's a real miniature line-array rig with a clear, powerful, focused sound, with good resistance to mic feedback. I hope I get to try one when the diary fills up again!

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ZP Microphones ZP800G

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Can a solid-state microphone really give you the sound of Sony's classic valve C800G?

ZP Custom Microphones are an established vintage microphone repairer based in Belgium, and who are now a boutique manufacturer, too. Currently the company offer a single limited-edition product, the ZP800G, and the microphone's name provides a strong hint of its intention. This is a high-quality vocal microphone designed to have a similar sound character to the much revered, but scarily expensive and exclusive, Sony C800G microphone.

Surprisingly, given its aims, most aspects of this new Belgian microphone are substantially different from Sony's classic. For example, the Japanese original uses a triode-connected 6AU6 pentode valve for its impedance converter, kept cool by a uniquely idiosyncratic cooling system extending to the rear of the mic body. In contrast, the ZP microphone features a completely conventional cylindrical body with solid-state FET-based circuitry.

One thing both mics do have in common, though, is the use of a customised 34mm, centre-terminated, Neumann K67-style dual-diaphragm capsule made in China. However, whereas the Sony flagship microphone offers selectable omni and cardioid polar patterns, the ZP mic has a fixed cardioid-only pattern.

ZP Microphones ZP800G

$2654

PROS

- Delivers a beautifully crafted, modern, crisp vocal sound.
- Well-built and solid.
- Supplied with an effective shockmount.
- Cardioid response and close working help reduce ambient sound.
- Ready to go the moment it's plugged in!

CONS

- The shockmount can't be stored in the mic case.
- The looks don't 'wow' like the sound does.
- Two grand is still a lot of money, even when it's much less than 10 grand!

SUMMARY

A conventional phantom-powered studio vocal mic cleverly tuned and optimised to emulate the distinctive vocal sound of Sony's flagship C800G.

The most significant difference of all between these two mics is that while the Sony C800G costs more than £10,500 in the UK, ZP Custom Microphones' ZP800G microphone is available for less than a fifth of that!

Given such radical technical differences between these two mics it might seem surprising that company CEO and designer Gregory de Mee claims the "sonic footprint and audio quality... are identical". While that was the primary goal, he also wanted the mic to be better suited to modern recording practices, meaning smaller, easier to rig, and more convenient to use — which was why he eschewed the delicate and slow-to-warm-up valve and its associated heavy external power supply, and adopted robust and instantly functioning phantom-powered FET circuitry.

The decision to provide only a cardioid polar pattern, when most vocal recordings with the C800G are probably made with the omni setting, might seem more perplexing, but de Mee's reasoning is that project studio acoustics aren't as good as those of professional studios and that a cardioid polar pickup is more forgiving in that context, as it excludes more ambient sound. Another way to reduce the ambient acoustic's contribution is to place the mic closer to the source, and so de Mee has optimised the capsule and circuitry for a relatively close working distance of 100mm (four inches). That's probably around half the distance that most vocalists in a professional studio would typically use »

» the C800G. Obviously, working a cardioid pattern mic this closely invokes the proximity effect, but the 'ZPK8A' capsule is built to ZP Microphones' own specs and the elements assembled and fine-tuned in the company's Belgian workshop to achieve the required tonal balance at that close distance.

One benefit of using a dual-diaphragm capsule in cardioid mode is that it is inherently less noisy than it would be in its omni mode — the reason being that only one side of the capsule contributes electrically to the microphone's output. Combining that aspect with high-quality solid-state electronics allows the ZP800G to claim a self-noise figure of 12dB SPL (A-weighted), whereas Sony specify the C800G in cardioid mode at 18dBSPL (and nearly 21dB SPL in omni mode).

The Package

The ZP800G ships with a hard-shell zippered case with internal padding. A net pocket in the lid behind a protective flap stores a hard plastic stand adapter and an 'info card' which is actually a flip-out 4GB USB thumb drive. Sadly, though, none of my PCs were able to access it so I've no idea what information might have been supplied upon it. A little faux-leather purse is provided to protect the microphone's grille when not in use.

The stand mount screws onto a threaded section around the XLR connector at the base of the mic, but most users will probably prefer to use the substantial shockmount which is also supplied. Frustratingly, this is too large to store inside the mic's case as it comprises a 120mm-diameter stainless-steel disc with attached stand adaptor. A stainless-steel cylinder is suspended inside the disc on four cleverly laced elastic cord loops, and a threaded collar at the centre of the cylinder attaches to the base of the microphone. The design is similar to the plate shockmount supplied with some Oktava mics, and that's not too surprising as ZP Microphones apparently worked closely with Oktava to develop the ZP800G's body hardware and accessories.

The microphone body measures 192mm in length, 50mm in diameter, and weighs 405g. Finished in a polished black paint, the ZP logo is stencilled on the front just below the multilayer chromed wire-mesh grille, while the 800G model name is at the rear near the base. Connection is via a standard XLR socket at the mic's base, and 48V phantom power is required (the current consumption is not specified). Sensitivity is quoted as -33dB/Pa, and assuming ZP meant dB ref 1V/Pa that would be equivalent to a healthy 22mV/Pa — just 1dB lower than the Sony C800G. The maximum level the mic can tolerate is given as 130dB SPL, but no distortion threshold is provided. Nevertheless, it should accommodate even the loudest vocalist without complaint.

In Use

I was unable to disassemble the mic to evaluate its construction, but ZP Microphones proudly states it uses Wima resistors and Nichicon and Russian NOS 'military-grade' capacitors, with hand-wiring and "top-notch quality control". The mic certainly feels solid and well-built, although the shockmount rattled on arrival as the stand attachment was loose. This was easily corrected with a screwdriver, but if it was my own mic I'd apply a thread-locking compound to prevent it loosening again with use. Nevertheless, when installed in the shockmount the mic feels stable and secure, and it can be mounted upright or suspended from above equally safely. The mic is not too sensitive to plosive blasting, but I would always recommend using a good pop screen anyway — if for no other reason than to keep the expensive mic clean and dry!

The capsule is optimised for working with very close sources. The sound is quite lean when the vocalist is six inches or more away, but at three to four inches the ZP800G delivers a very nice, modern, mix-ready vocal sound which has a slightly compressed midrange character with plenty of energy that delivers a full-bodied quality. This is balanced against a distinct upper-mid presence which has been shaped to give a nice clarity and clean diction without over-emphasising the sibilance region. Above that is an open and airy top end that brings a silky smoothness to breaths, and the overall impression is of a larger-than life voice with a well-focused, almost 3D quality. Within mixes I found the vocals needed almost no further processing beyond some dynamic control. Every part of the voice was present and correct — body, weight, clarity, crispness, breathy airiness, all ideally balanced from the outset.

ALTERNATIVES

Ignoring the Sony C800G itself, the two obvious alternatives to the ZP800G are the **Golden Age Premier GA-800G** and the **Advanced Audio CM800T**. Both of these mics follow the same core design of the Sony flagship, and both are valve mics, but the GAP GA-800G is a very close clone, even retaining the Peltier cooling extension. So it looks very similar to the Sony original, but costs around a third as much, making it around $1000 more than the ZP product.

The **Canadian Advanced Audio CM-800T** takes a rather simpler approach, repurposing parts from some of its other models and a different NOS military valve to come in at under $1000. The overall tonal character is very similar, though, and it has the added benefit of providing three polar patterns (omni, cardioid and fig-8).

I've only used the Sony C800G a few times, but from memory I'd say ZP Microphones have captured its sonic essence pretty well here. Regardless of its notional accuracy, though, the ZP800 is a very appealing vocal mic in its own right and, despite its rather understated looks, it delivers such a flattering and complete sound that every vocalist will be impressed from the moment they don their headphones. And as we all know, happy talent delivers great performances!

From a technological point of view, there is little in this microphone that is out of the ordinary: it uses a replica capsule with pretty standard FET circuitry in a familiar tube-shaped body. But as is always the case, it's the little details that really make the difference. The specific machining precision of the capsule backplates, the hand-assembly and careful tensioning of the diaphragms, the selection of the passive and active components... The complete product is so much more than the sum of its parts, and that's why its high price can be justified. It is a very high-quality vocal microphone with no caveats, limitations or provisos. And if viewed as a legitimate alternative to the legendary C800G — which I think it probably is — it is clearly an enormous bargain. Either way, it's certainly a mic worth adding to the shortlist for an audition if your budget extends this far.

$ $2654.91
E zpmicrophones@gmail.com
W www.zpmicrophones.com

NEIL ROGERS

Bad Dogs P1

500-series Microphone Preamplifier

Ever fancied trying different output transformers in your mic preamp?

Bad Dogs were founded in 2020 by Italian electrical engineer Francesco Canacci and recently sent us their debut product, the P1 500-series mic preamp, for review. A high-quality but cost-effective preamp, it offers an interesting twist that should help it stand out in what is a crowded market place: the user can choose to add or swap output transformers, to tailor the sound to taste.

On Test

The P1 is a simple, tasteful-looking design that includes all the standard features you'd expect of a good preamp: 48V power, switchable polarity inverter, 20dB pad and 80Hz high-pass filter, and a front-panel quarter-inch TS jack instrument input. The mic preamp can apply between +12 and +65 dB of gain and has an input impedance of 1.5kΩ, while the instrument input offers -5 to +45 dB and a 1MΩ input impedance.

Both can handle pretty hot sources and are within -3dB of a flat frequency response from 4Hz to 50kHz when at their maximum gain.

The review model was supplied with two cost-option output transformers: a Jensen JT-11-HFMPC and a Lundhal LL1517. The P1 doesn't need these to operate, but they do introduce subtly different characters, should that be desired. The idea is that the user fits or swaps out the transformers. Following the steps described in the manual, this was a really easy, straightforward process: there's no soldering or wiring involved; you just flick a small jumper on the circuit board and carefully insert the transformer in its socket.

The P1 arrived at an opportune time, as I already had several preamps in the studio for testing. It's often surprising just how small the audible differences can be between models, even those with very different asking prices, but there are subtle differences nonetheless, particularly if you drive them harder. Putting the P1 up against some well-known and expensive preamps, the P1 held its own very well indeed; and these comparisons included close kick and snare drum duties as well as DI'd bass guitar and acoustic guitar recorded with a small-diaphragm condenser mic.

Bad Dogs describe the P1 as a high-bandwidth design and my perception, with no transformer fitted, was that it captured the full range of an instrument without imparting a very noticeable character — reminiscent of the clean-sounding preamps on my Audient 8024 console. Bringing the transformers into play made a meaningful difference to the sound, albeit a small one, and pushed the P1 more in the direction of, say, my Neve 1073LB modules. Assessing the difference between the Jensen and the Lundahl transformer sounds felt very much like splitting hairs. There are discernibly different characters, but it's all very subtle; I'm not sure my own audio geekery has quite reached the level where I have a distinct preference but when A/B comparing recordings I made with them, on kick and snare in particular, I felt that the Lundhal revealed a little more in the low-end, whereas the Jensen seemed a touch brighter sounding on the snare.

When it comes to preamps, 500-series users are spoiled for choice but the P1's price/performance ratio makes it very good value. It's a preamp that I'm very happy to use alongside or instead of the various big-name preamps I have here that cost more than twice its price, and the optional transformers are not merely a gimmick: both add a familiar, pleasing thickness that works well on most sources. If you just want a good preamp, then, the P1 is well worth considering, but if you enjoy shooting out equipment and diving deep into fine-tuning your recording chains, it could prove particularly attractive.

summary

A classy 500-series preamp that's good value for money and offers some useful tonal options courtesy of swappable output transformers.

$ P1 standard €369 (about $436). With one transformer €459 ($542). With both transformers €549 ($649).
E info@baddogsaudio.com
W www.baddogsaudio.com

Sound Particles Brightness Panner

Auto-panning Plug-in

This stereo/surround frequency-based panner is dripping with creative potential.

PAUL WHITE

Sound Particles' Brightness Panner has much in common with their Energy Panner, which Hugh Robjohns reviewed in the August 2021 issue of *Sound On Sound*, so to avoid excessive duplication, I'd suggest that you read that in conjunction with this review. The panning engines seem pretty much identical, so the difference between the two plug-ins is really in the way the panning is controlled. While Energy Panner responds to the amplitude of the incoming signal (which can be mono, stereo, immersive or surround) to trigger panning, Brightness Panner can be triggered by note pitch (with user-adjustable high and low note ranges), by tonal brightness via a variable filter, or via MIDI. The plug-in supports VST 2 & 3, AU, AUv3 and AAX native on recent Apple and Windows operating systems; Logic users will need to instantiate the MIDI-controlled version of the plug-in via an Instrument track in the usual way if MIDI control is required.

The panning modes include stereo X-Y and M-S as well as binaural, with a choice of discrete surround formats up to 7.1. The resizable user interface is very similar to that of Energy panner other than the three triggering modes. There's still the large, circular display representing a three-dimensional dome-like space within which the input signals are panned. The start and end points of each panning motion are shown as points with a dotted line indicating the panning path, and with speaker positions shown around the circumference. As with Energy Panner, there are two main panning modes: Pan and Sliding. In Pan mode, when the input signal falls outside the trigger range, the pan direction reverses so the sound moves back towards its starting point. In Sliding mode, for which the GUI colour changes, the panning stops when the trigger disappears but then continues from where it left off when a new trigger is detected.

The Sliding mode, in which panning pauses at its current position when there's no trigger signal.

There are four movement settings: to Speakers or to End Point, with a choice of clockwise or anti-clockwise motion. Depending on how you set it up, the stereo image can widen when triggered, the left and right channels can swap places or the virtual sound source can move around inside the dome, either in a circular path or in more mysterious ways that depend on the movement setting and where you place the start and end points. I'm never entirely sure how some of the movements are calculated, as some motions seem to do their own thing to an extent, with just passing nods towards your start and end point settings. But the result is always interesting and there are controls to adjust the rate of movement. A side-chain input, for example, enables panning of one sound source according to the characteristics of another track, and there's a wet/dry balance control to allow the panning effect to be diluted.

I'm already using Energy Panner quite a lot when working on my more ambient mixes, usually in binaural mode for scattering ear candy around the soundstage. But as with Energy Panner, while Brightness Panner does sound more dramatic when used in surround sound and immersive mixing formats, it works perfectly well with good old-fashioned stereo. You

The Pan mode: when the input signal is beyond the trigger range, the sound moves back towards its starting point.

can set it up to keep low frequencies relatively narrow, while giving the higher pitches more leeway to wander. And if you put a swept filter before Brightness Panner, pad sounds can take on a new sense of movement, as the motion follows its timbre. MIDI control can be used to create rhythmically tight pan patterns too. Then there's the option to have the pan position follow pitch, so that a musical scale plays from left to the right or vice versa. I've found that setting a very fast pan speed can sometimes give slightly lumpy results, but at more moderate rates the result is smooth. Given their relatively low cost and ease of use, there's a lot to commend both Energy Panner and Brightness Panner, even if, with so much in common, part of me wonders why they didn't combine both options in a single plug-in.

summary

Supporting both stereo and surround panning, the Brightness Panner can inject interest and movement into a range of mono, stereo or surround sources.

$ Individual price $49.
Panner Collection (Brightness Panner and Energy Panner) $69.
T Sound Particles +351 244 859 465.
E info@soundparticles.com
W https://soundparticles.com

ROLI Equator2

Software Synthesizer

ROLI's Equator2 is less an update than a full rebuild.

NICK ROTHWELL

Equator2 is a refresh of the original Equator software instrument from ROLI. That first version actually ran natively inside the ROLI Seaboard Grand as well as being available as a soft synth. Equator2 is an update which extends and generalises Equator1, and improves its interface.

Installation is via the ROLI Connect app, the fiddliness of which I think I grumbled about a bit in my review of ROLI's LUMI keyboard, and installing Equator2 itself is not enough: 5GB of sample library needs to be fetched separately, otherwise most of the presets of this sample-oriented synth will be silent. You probably want to wheel in the original Equator1 material too — there are some useful single-cycle waveforms in there, for a start.

First Impressions

Visually, Equator2 has a bit of the Bang & Olufsen about it: grey controls sit against a grey background with very subtle shadows, and a bit of colour is used for sparse (but important) highlighting.

Equator2's window layout is very much a game of two halves: oscillators (or, as Equator2 calls them, 'sources'), routing and effects share the top half, selectable by tabs, whilst the lower half is completely dedicated to modulation.

Nothing special jumps out from the synth's rather austere appearance, but it is subtle, deep and layered. For a start, a voice (by which I mean what's heard when a single key is pressed) comprises six parallel sources, each of which can be configured as a PPG-style wavetable player, a sample player, a granular source or a noise generator. FM and ring modulation are provided for the synth purists. There are two multimode filters per voice, but each source also has a dedicated filter, making eight in all. Modulation is very well catered for, with five envelopes, five LFOs, two multistep modulation sources and some 'maths' functions for side-band modulation, randomisation and the like. Conventional and MPE gestures can be programmed as modulation, and there's a macro control panel for broad-stroke control.

Sound Sources

A central oscilloscope display is flanked by the six panels for the sources, three on each side. Click on a source panel, and the central area switches to a detail view of more controls for that source. Single-cycle waveform displays are animated — useful to see 'symmetry' distortion and wavetable sweeps — but there's little visual indication for sample or granular playback, or filter shape. In general, Equator2 is pretty miserly with

The top half of the Equator2 controls, showing sources and filters.

visual feedback, which is a bit of a drawback.

For a source, Wavetable mode really means single-cycle waveforms of some kind. There are some waves (pulse, sawtooth, sine, square, triangle) which are procedurally generated

An expanded view of one wavetable source, with filter.

for maximum accuracy — important for techniques like FM — and a couple of hundred sampled-based sweepable PPG-style tables, plus a handful of single-cycle samples. These can all be shaped by the 'symmetry' control, akin to PWM. Waveforms can be stacked and detuned within a single source, leading to the sort of chorusing effects beloved by PPG fans.

Any sources in this mode can participate in FM synthesis: a source can have its frequency modulated by two others. If you want to arrange sources into an FM stack you can mute those intended as modulators, or just not mix them to the outputs. The various FM INIT presets display a clean, crisp FM tonality, even when running with just two or three sine-wave sources. Other sources could add to the FM or layer it with samples, taking things into classic Yamaha SY territory, especially with some tastefully programmed FM dynamics. For even more harmonic fun, sources can be ring modulated too.

In Sampler mode a source plays samples, as the name suggests. The controls are minimal: apart from the standard pitch, pan and level controls there's a start offset knob and a switchable 12dB boost, and that's about it. Most of the named samples are actually multisample sets, but this is only really apparent if you dig around in the file system to find them. Some samples are one-shot, some are looped: there's no indication which is which and no display of loop or crossfade points. It probably goes without saying that there's no actual sampling functionality in the sampler, but you can drag and drop multisample sets on to the sample chooser to add your own into a personal library. Even this modest task might be a challenge depending on your DAW, if it insists on hiding plug-in windows when backgrounded. (In the end I launched Equator2 as a standalone application when I wanted to add samples that I could use in my DAW.) None of the preset sample sets is fantastically big (the baby grand piano is 340MB, rather small by today's standards), but the material is varied and decent quality.

Some of the disappointment of the sampler is assuaged by the next source mode, Granular. Pretty much everything you'd expect from a granular synthesizer is here. You can set, and modulate, the starting sample position for the grains, the scan rate through the sample, size and creation rate of grains, grain shape, playback direction, pitch randomisation »

ROLI Equator2

$249

PROS

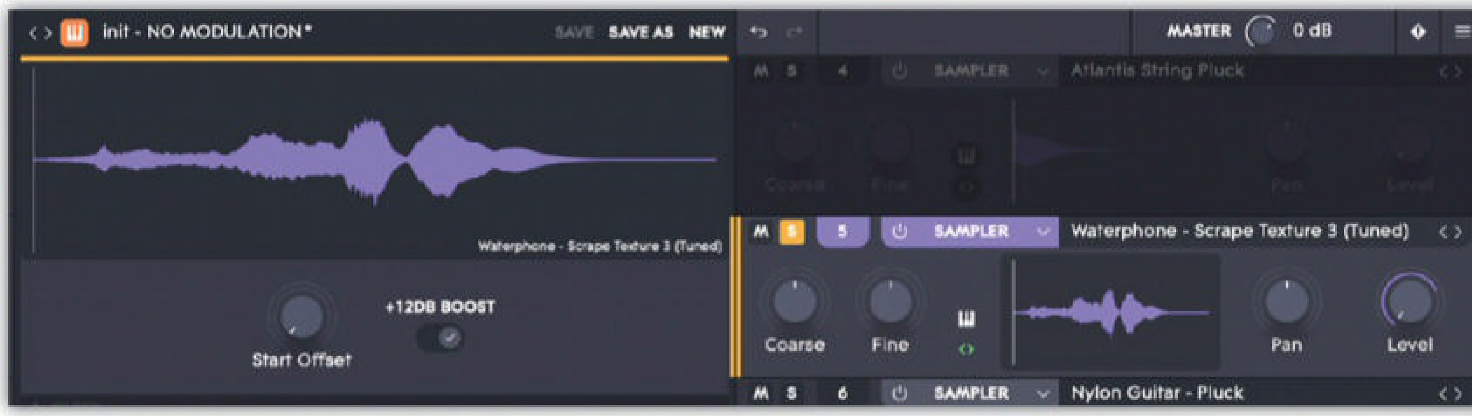
- Great-sounding, multi-layered instrument.
- Mix of FM, wavetable and granular synthesis possible within a single preset.
- Powerful but accessible modulation matrix.
- Some nice-sounding effects algorithms.

CONS

- A lack of visual feedback of audio signals makes editing a bit opaque.
- The flat greyness of the interface causes some controls to get a bit lost.
- Sample playback is rather under-powered.
- Distinction between MPE and 'standard' presets can be a little inconvenient.

SUMMARY

Equator2 is a flagship soft synth from ROLI, and is oriented towards, although not reliant on, controllers with MPE support. Although predominantly sample-based it offers a wealth of synthesis and modulation options in an easy-to-use package with no serious downsides, although the user interface could provide a few more clues as to what's going on when it's playing. For responsive, dynamic cinematic soundscapes with accessible controls it deserves a serious look.



The rather sparse sampler source.

and stereo width. Rather unusually, grain creation can be quantised to the clock, allowing for rhythmic effects (there are some interesting drum rhythm presets which use this technique). Sample selection works in one of two modes: in 'note' mode the multisampling machinery is still in play, so that you get the appropriate sample for the pitch you're playing. In 'index' mode this is switched off, and instead you get to choose which of the samples to use. This isn't all that useful for the stock acoustic samples, since you're taking them way out of their natural root pitch, but there are some rather good 'granular' sample sets where each sample is a distinct texture. Sweep the index and position with some LFOs and you get some truly dynamic and engaging textures: cleverly, Equator2 lets you dynamically modulate which sample you're using for grains within a single held note.

The final source type is Noise, with the choice of white, pink or a sparser 'crackle'. A density control can thin the noise effect out. Noise is not particularly exciting on its own, but for thickening up textures or building drum sounds it does the job.

The voice architecture provides two global multimode filters per voice — each including a tone control — and each source also has a dedicated filter, so that's eight in total per voice. Unless you're an avid reader of manuals, you might well not spot them all: the source filter is only shown in the detail view and is disabled by default, visually fading into the grey background.

Routing

Downstream from the six sound sources and the global filters there's an effects section, but to engage this we have to go into routing mode first, via a tab top-left, though in fact 'mixing' is a more accurate term for what is effectively an audio matrix. Each of the six sources, and the ring modulation output, has five level knobs for sending signal into the two global filters, two effects chains, and/or directly to the audio outputs. These five levels can be modulated to vary each source's downstream processing over time, or automated for overall control. The filters can be arranged in parallel or series with various routing options, and their output can be routed into the effects and/or dry to the main output.

The routing matrix is not fantastically complicated but is subtle enough that I found myself wishing for some kind of VU metering to get a sense of what was going where or to explain what I was hearing. (Or why, sometimes, I wasn't hearing anything at all.)

Effects

Equator2 supports up to 12 effects in a preset — which should be enough for anyone — accessed by a third tab. The available effects are basically the staples (chorus, flanger, filter, gate, distortion/bit-crushing, EQ, various kinds of delays, a couple of reverbs). There are serial and parallel (six plus six) configurations for the effects block.

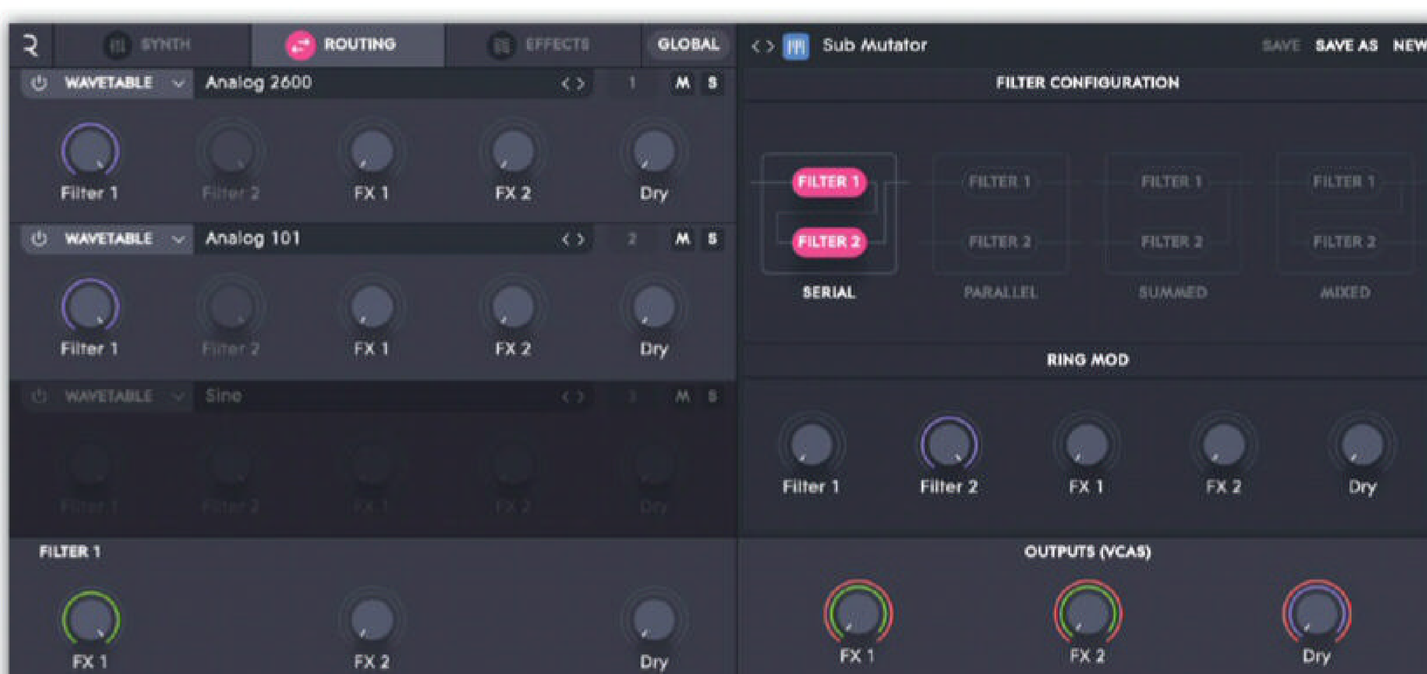
The effects controls are pretty plain, and there's no visual animation or level indication in any of them. Frustratingly, the compressor effect even sports a graphic of a VU dial but no needle to make use of it. So, as ever, you'll have to use your ears.

Having slightly dissed the effects in general, I'm going to make an honourable mention of the grain delay, which can be pressed into all sorts of textural pitch-varying services, and the modelled distortion effect which supports a range of algorithms and sounds rich and evocative. Handily, each of the effects algorithms comes with a menu of its own mini-presets.

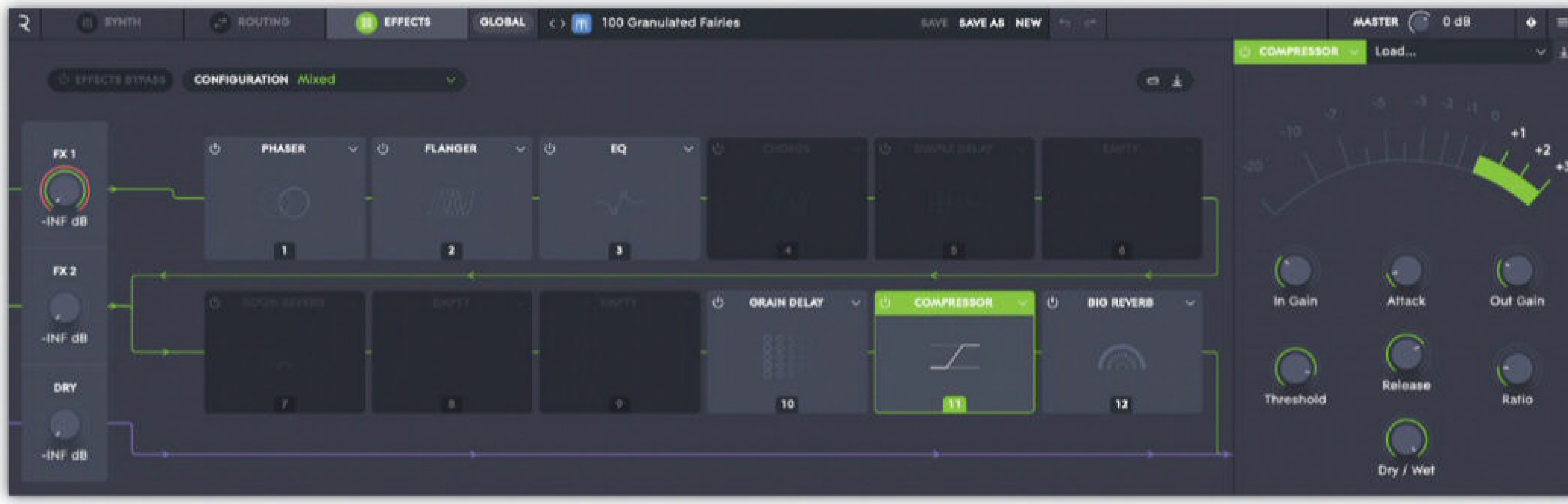
Modulation

Equator2 is big on modulation, which is why it devotes half of its window to it. There's a full modulation matrix behind the scenes, complete with transfer (shaping) functions, macros, mathematics and MPE gestural support.

We'll start by looking at the visual cues for modulation. Any parameter knob which can be modulated is enclosed by two thin concentric rings. The inner ring shows the current value, which by default will match up with the marker on the knob, while the outer ring shows the



The routing view, routing sources into filters and effects.



Lots of effects slots, though the effects are rather plain.

extent of the programmed modulation on the parameter. In the nearby screenshot of Strike, the filter cutoff and envelope level knobs show actual values in purple and red respectively; both values are higher than the knob position because the parameters are being modulated by key strike which is highlighted in the centre. The blue of the strike modulator is reflected in the blue outer rings around the knobs. Click a different modulator (they're colour-coded in groups) and the outer rings will update to show that modulator's influence. Click and drag the outer ring area to change the modulation amount; the inner ring will update in response.

As is usual, some modulation sources are per-voice (envelopes, LFOs and MPE gestures), others are for the entire preset (macros, conventional MIDI controls like mod wheel). Per-voice modulation sources can be applied to preset-level effects parameters — there's a setting to specify which voice supplies the modulation, or you can apply an average of all sounding voices.

For a more detailed view, you can right-click a modulator's title tab to see a table of the controls it is attached to, together with amounts and transfer functions — entries can be added, edited and deleted here. Right-click on a control knob and you can see all the modulation sources which affect it. You can also pull up a global mod matrix table for the entire preset.

Looking at the modulators themselves, there are five identical envelopes, although the first 'AMP ENV' by convention controls voice allocation. Envelope modes include 'ADADR' which loops the attack-decay portion while a note is held, and 'ADS-PR', which provides a pluck-release section on note-off. (The attack decay is still present, so there seems to be no way to make an envelope 'pluck only', as it were.) Most segments have adjustable curves. Segment times range from sub-millisecond (at least according to the control readout) to 32 seconds maximum, and there's a sync mode to use metrical times including dotted and triplet values.

There are five general-purpose LFOs with multiple waveforms, and two 'multi-mod' units which are essentially loopable grid-based envelopes, the closest thing Equator2 has to a step sequencer. Once again, I found myself wishing for a bit of live visual feedback here: it would be good to see the current playback position of a multi-mod (per voice, ideally), especially since the playback rate can be modulated, and multi-mods can free-run without notes being held. As ever, you'll have to use your ears.

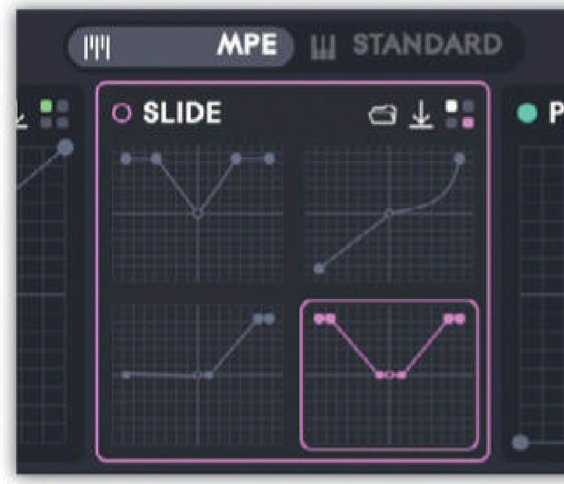
Also in the modulation arsenal is keytracking — again editable as a multistage curve. Unless you click on a tiny icon (or read the manual!) you may completely miss the fact that there are four independent keytracking sources. Keytracking is not applied to a parameter directly as a modulator: instead it influences the other modulators already on that parameter, by shifting, limiting or scaling their effect.

A collection of math modulators take one, or two, input modulation sources and process them in some manner: lag,

Lots of options on the table in the modelled distortion space.

Strike (aka key velocity) as modulator.

Table of modulation destinations.



The slide keyboard gesture, available with four different response curves.

quantise, add, multiply, threshold, etc. In the same group is a set of random sources which are sampled at note-on, and a 'flip-flop' which alternates between minimum and maximum on each played note.

Last, but by no means least, we come to the performance modulators associated with keyboard gestures, dominating the lower centre of the window. Equator2 believes in two kinds of keyboard controller: those which support MPE, and standard ones which don't. Every preset is either configured for MPE or not. In MPE mode the five modulators are for the so-called 'five dimensions' of gesture (strike, glide, slide, press, lift), available as per-voice modulation sources. Switch to standard mode and the modulators switch to their traditional equivalents (velocity, pitch-bend, mod wheel, etc).

The rest of the preset isn't directly aware of whether modulation is coming from the MPE sources or not: switch to standard and mod wheel will have much the same affect as slide, albeit monophonically. This led me to wonder why Equator2 ships with many of its presets essentially duplicated, with a standard and an MPE version of each. According to ROLI, there are enough articulation differences between MPE and standard presets, even for the 'same' sound, that it makes sense to customise them.

As with keytracking, each of the five types of performance modulation is available as four distinct sources, with individual editable response curves. There's a selection of preset curves and new ones can be saved into a library.

Wrapping up the tour of modulation sources we have five macro controls, which just so happen to resemble the five left-hand controllers on a ROLI Seaboard. You don't need to own a Seaboard to make use of them: they are MIDI-mapped, and you can automate them from your DAW. Two of the five controls are presented as X and Y axes on a grid — mimicking the Seaboard hardware — which led me to reflect on how useful it might have been to have more X-Y controls inside Equator2, especially for the routing and mixing of the six sound sources into the filters and effects. As a Korg Wavestation fan, perhaps I just want to vector-mix everything.

Sounds

Equator2 comes with a shade under 1500 presets, of which about 40 percent are MPE-capable, plus 360 legacy presets from Equator1. In addition, ROLI are putting out paid-for soundpacks, as first seen in their Studio Player. Since I own Equator1 and Studio Player, the compatible soundpacks from both products appeared inside Equator2. As I write, four new Equator2 packs are available for purchase and download, and by the time you read this a free pack called Motion Waves should be part of the Equator2 bundle.

There's a fair range of good-sounding material on offer, more than I have space to describe here. Good use is made of Equator2's modulation features, with a few rhythmic patterns that would fool you into thinking that you had some kind of arpeggiator on board. I was not that taken with vintage synth and SynthWave sounds, but only because my personal philosophy is that you're better off crafting material like that for yourself. Orchestral fare is pretty thin on the ground, too. On the other hand, the cinematic soundscapes and 'action sequence' presets sound contemporary, rich and expressive. It's the MPE presets which really shine in terms of expressiveness and control, so if you're a soundtrack maker you should think about investing in a Lightpad or Seaboard Block as an input device.

Conclusion

I'm going to come clean and confess that I was initially slightly biased against Equator2. I can have a bit of an issue with sample-based synths in general, probably dating back to my days with a Korg M1, and my personal take on the original Equator version 1 is that it was tailored to making a Seaboard sound good — which it does! — rather than being a fully rounded instrument in its own right.

With Equator2, it's clear that there's been a rethink of the design to make it more rounded: six general-purpose sources per voice, granular synthesis (which will always win me over to a sampler-player), powerful audio routing, lashings of effects and a top-notch modulation system. It is only let down slightly by the feature-challenged sample player. Visually the controls are perhaps a little too understated — it's a bit hard to see what's enabled and what's not at a glance — and such a powerful device would benefit from more animated feedback of audio levels and real-time modulation. But overall there are no sharp edges or gotchas, accessibility is good given the complexity of the device, and it sounds great. If you have a ROLI Seaboard or Lightpad I'd say it's a no-brainer; even if not I strongly recommend it.

$ $249, upgrade $149.
W www.roli.com

Test Spec

Equator2 version 2.2.0.
Apple MacBook Pro (Mid 2014), Mac OS Catalina 10.15.7.
Bitwig Studio 4.0.1.
Ableton Live Suite 11.0.5.

Trinnov D-MON & La Remote

Monitor Controller With Listening-system Correction

Trinnov's pioneering range of room-correction systems now has a dedicated monitor controller.

HUGH ROBJOHNS

My first experience of Trinnov's uncompromising approach to digital speaker (and room) correction was 15 years ago. There really was nothing else on the commercial market at that time to match its astonishing capabilities in such a practical and comprehensive manner and, although other manufacturers have narrowed the technology gap over the last decade or so, Trinnov's current hardware and software implementations have raised the benchmark even higher. Back then there was just the Optimizer but today Trinnov offer a variety of hardware processors, each optimised for a different specific market: home-theatre, hi-fi, pro-audio and professional cinema.

Trinnov D-MON & La Remote

From $8724

PROS

- Unparalleled room-correction capabilities.
- Professional monitoring sound quality.
- Compact and interactive USB remote-control interface.
- Remarkably easy installation, calibration and customisation.
- Versatile I/O connectivity.
- Aligns and controls multiple independent monitoring systems.

CONS

- Some monitoring features have not yet been implemented.

SUMMARY

The Trinnov D-MON system combined with La Remote hardware control interface presents a unique combination of a sophisticated and customisable monitor controller with class-leading room-correction technology that's capable of handling multiple separate speaker systems.

Overview

Phil Ward reviewed the ST2-Pro stereo model for professional studio applications in *SOS* July 2019 and was very impressed. As his informative review is available for free in the *SOS* online archive (www.soundonsound.com/reviews/trinnov-st2-pro) I recommend reading it for more in-depth information about what Trinnov's Optimizer Toolbox system does and how it does it; there's little point me re-treading that ground here. Suffice it to say that Trinnov's systems correct both the amplitude and phase responses of the loudspeakers, with the aim of recreating the most accurate transient response at the listening position. This allows the most precise rendition of time-dependent auditory cues, and the benefit can be heard as a much more focused and stable stereo image, and more realistic reverberant spaces.

The system can also rectify physical misalignment of speakers (particularly in multichannel arrays) by introducing compensatory delays and remapping the various channel signals between speakers, where necessary to generate the format's intended sound field. It also corrects some room peaks, cancellations and primary reflections, but is clever enough not to make the matter worse by going beyond what is reasonable. Sophisticated facilities for manual system/room EQ are also included, along with comprehensive parameters to fine-tune the application of calculated optimisation settings within the physical loudspeakers' capabilities and to suit personal preferences.

System analysis is performed using a bespoke four-mic tetrahedral array, which is included with most Trinnov systems. The spacing between capsules allows the software to determine the directional information needed to work out the precise locations of each speaker, and any strong acoustic reflections, in three-dimensional space.

As you might expect, this level of sophisticated signal analysis and real-time processing is substantially more demanding than can be delivered by typical digital speakers with integrated DSP, or even by room-correction DAW plug-ins — hence the need for the powerful dedicated hardware platform. Trinnov's solutions are all based on custom (silent) PCs in an elegant 2U chassis, with rack-ears for the professional models. They all run a Linux operating system, with the number crunching performed in 64-bit floating point. A specially designed Trinnov Audio Core (TAC) DSP card handles communications between the PC and the mastering-grade analogue I/O, as well as the digital audio, clocking and ancillary interfaces. AES3 digital I/O is also included as standard, while the largest systems use MADI, Dante or Ravenna interfaces.

The hardware processor can be controlled through a local mouse/

screen setup, but more usually Optimizer installations are controlled either by VNC or a web browser — the computer contains both VNC and Web servers. With Internet access, the hardware can 'phone home' to Trinnov's servers, from where it can be configured remotely by Trinnov boffins. That's an extremely useful feature in such a sophisticated machine!

D-MON

The majority of Trinnov's systems are intended for use with a single loudspeaker array: to correct a standard stereo setup, a 5.1 surround system, or even a multichannel Dolby Atmos setup (the largest system can process up to 64 speakers in a single acoustic array). However, most professional studio environments employ two or three independent sets of monitors, possibly in different formats, too. Historically that would have entailed a monitor controller feeding separate Trinnov correction systems for each different speaker set.

The impracticality of this is not lost on Trinnov, though, and the subject of this review, the D-MON, is specifically designed to work with several independent loudspeaker configurations in the same room, at the same time. It has been designed to control and switch between them just like a conventional monitor controller and, in fact, it's probably best to think of the D-MON as a fully configurable digital monitor controller first and foremost, that just happens to have sophisticated integrated speaker/room-correction processing facilities built-in!

Currently available in two versions, the D-MON|6 and the D-MON|12 provide either 6 or 12 Optimizer signal-processing channels, respectively. So the D-MON|6 can be used to control a single 5.1 array or three stereo sets of monitors, for example, while the D-MON|12 can handle a 7.1.4-format Atmos system (with three front speakers, two pairs of side and rear speakers, one sub, and four height speakers), or a 5.1 array and three stereo pairs... or any other combination that fits within the unit's processing capabilities. »

The D-MON units have plentiful I/O on the rear, and can cater not only for monitor correction and switching but also artist cue mixes.

All the speakers in the various monitoring arrays are normally analysed and aligned in one process, to ensure consistency at the listening position, but after that they can be controlled completely independently.

Beyond their six or 12 Optimzer DSP channels, both D-MON hardware units include additional 'uncorrected' audio I/O channels. These are provided for additional input sources and for feeding separate (unprocessed) outputs for artist headphone cue mixes (with talkback, if required) or other ancillary signal paths. Consequently, the total I/O count of both units is significantly greater than the DSP channel count suggests. For example, the D-MON|6 has four AES3 inputs and outputs (eight channels in each direction), eight analogue line inputs, and 12 analogue line outputs. The larger D-MON|12 has eight AES3 inputs and outputs (16 channels), eight analogue line inputs, and 16 analogue line outputs.

In both cases, the hardware incorporates a full routing matrix, allowing any physical input to be routed to any physical output (with level changes, if required), as well as via the Optimizer processing. There's also a versatile summing matrix (ie. a mixer) for generating artist cue mixes, monitoring downmixes, and so on.

I mentioned the D-MON's mastering-grade converters earlier, and Trinnov's specifications quote dynamic ranges of 119 and 118 dB (A-wtd), for the A-D and D-A respectively, with the ability to work up to 96kHz sample rates. I ran my usual Audio Precision system measurements to confirm these specs, which place the D-MON just outside the top 10 of my converter review tests to date. Similarly performing converters and interfaces include the Prism Lyra 2, Apogee Symphony, Cranborne 500R8, Mytek Brooklyn DAC+, the UAD Apollo and Crookwood's M1 mastering console. So it's fair to say that the Trinnov's analogue signal path is at least as good as most mastering studios would require. Wordclock is provided on the hardware platform for sample rate synchronisation, of course, and there is also comprehensive interfacing for GPIO and MIDI, as well as integration for ICON and EUCON remote control protocols.

La Remote

Although the D-MON's monitor controller features can be operated remotely via a web browser, or from an Avid console via Eucon, Trinnov have recently introduced a dedicated physical desktop controller called La Remote. This new addition makes the Trinnov D-MON a really practical proposition for any mix or mastering room, not just those equipped with Avid hardware.

This USB bus-powered controller is recognised as soon as it is plugged into the hardware unit but its configuration data is stored (and programmed) in the main hardware platform as part of the Optimizer software and uploaded to the La Remote during bootup.

With La Remote sitting on the desk or console, the D-MON really becomes a 'proper' monitor controller, and once configured the speaker/room-correction aspect becomes as transparent as the sound from the speakers themselves; the operational focus is entirely on the versatile La Remote which can be customised to suit each installation's specific requirements. (For those with Trinnov ST2 or MC platforms, the La Remote unit can be used as a more basic monitor controller, with fewer facilities and customisation options than are available when connected to a D-MON.)

Physically, La Remote is a pleasingly compact but surprisingly heavy wedge-shaped unit, dominated by a monochrome LCD screen. Four illuminated buttons run down each side, and seven of these buttons are user-assignable; the one at the bottom left always provides a master Mute function. An eighth assignable button, usually configured for either Dim or talkback functions, is located just below the large rotary encoder which serves as the main volume control.

The large rotary encoder is unusual in featuring a magnetic clutch system which makes it feel like a high-quality rotary switch, with reassuring tactile feedback. A user-selectable acceleration mode allows a fast spin to change parameter values quickly over a large range, while slower moves provide much finer resolution, and this all works very well in practice. The volume level of the currently selected speaker set is displayed in a banner across the top of the screen, either in a relative dB scale (gain/attenuation) or as an absolute dB C acoustic SPL value.

A second, smaller encoder, at the top right of the panel, selects different menu layers, with the default selection accessing basic monitoring facilities, output metering, configuration presets, networking parameters, and so on. However, pretty much everything is configurable and extra layers can be added as necessary. A talkback microphone is built in at the top left of the controller, and a USB B-type socket on the rear is the only connection to the Trinnov hardware unit, which makes for a very neat installation.

For such a versatile system, configuring the La Remote is remarkably straightforward. A web browser is used to access the La Remote configuration page in the main hardware unit. Any number of menu Layers can then be defined in addition to (or instead of) the default set, and a simple drag-and-drop paradigm is used to place the available functions from

For those without a La Remote controller, the D-MON units can all be controlled by a browser-based app: this is the main one of several different screens.

a list on the left of the screen directly onto the appropriate soft buttons in each layer.

There are far too many options to list them comprehensively here, but monitoring functions include mono sum, left-right swap, and stereo difference options, as well as various surround downmixes with customisable contribution levels. A calibration reference level can also be established, sources can be selected exclusively or summed with others, output speaker sets and/or headphone outputs can be selected, individual speakers muted or soloed, and talkback and listen-back activated to/from different destinations.

An unusually comprehensive bass-management system is available when working with multichannel arrays, and Presets allow completely different setup configurations to be loaded almost instantly. One application of this is to load different speaker/room-alignment parameters, optimising the reproduced sound for a sofa at the back of the room, say, instead of at the mix position, so you can impress the clients! And the cherry on top is the ability to load a (monochrome) company logo for display on the La Remote screen.

In Use

I was very pleasantly surprised with the ease of installation and configuration, assisted greatly by the way it can be controlled remotely by the Trinnov gurus. The hardware is well-built with plentiful mastering-quality audio interfacing, and the highly optimised software is surprisingly straightforward to use; I rarely needed to dip into the comprehensive manual.

Most Trinnov systems ship with the dedicated, battery-powered, multichannel calibration microphone, with its four miniature capsules emerging on stalks from the top. Its outputs are at line level and plug directly into the analogue line inputs of the Trinnov system when performing a speaker/room calibration, and when the mic's serial number is entered into the calibration menu the Trinnov system automatically locates the relevant calibration file to ensure a precise alignment. Simple on-screen instructions guide the user to perform various tasks in the alignment process (such as turning on and off the microphone at appropriate moments to avoid howl-rounds)

As I was reviewing this unit during a lockdown and while in isolation, the remote-control feature proved very useful indeed: I was able to converse with Trinnov's UK agent over a Zoom call while he took control of my specific installation remotely, demonstrating the features and facilities in real time. He was also able to work through a first speaker/room alignment with me, explaining the pros and cons of the various customisation settings and options in the Trinnov Optimizer software as we progressed.

All I had to do was plug the D-MON into my studio monitoring chain (in place of my usual Crookwood system) which involved re-plugging four XLRs on a patchbay, and it was all completely configured and calibrated in well under an hour! I subsequently performed a couple of other room calibrations of my own, just for experimental purposes, and was very impressed with both the ease and flexibility to tweak and fine-tune settings to suit personal requirements or speaker capabilities.

My 'raw' studio/monitor acoustic frequency response is pleasingly good, having a response comfortably within ±4dB above 200Hz, but with a narrow 6dB notch around 100Hz before a slow roll-off reaching -6dB at 50Hz. With the Optimizer running that improved to ±0.75dB above 50Hz on the left channel and the -6dB point moved down to 30Hz. The right channel was virtually identical except for a narrow 4dB notch at 100Hz, something I eventually tracked down to the acoustic effect of the wooden side of a rack facing that monitor. A deep acoustic panel has been ordered to fit over it...

Due to the physics of analogue crossover filters, the phase response of my all-analogue three-way Neumann KH310s monitors involves a couple of trips around the phase circle, but the Optimzer software resolved that to within ±15 degrees above 100Hz, with a gentle rise below that to around +165 degrees (this being a practical compromise to avoid excessive latency through the DSP). The benefits of these corrections were also evident in the direct impulse response which tightened up dramatically, as did the first reflection from the desktop.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The La Remote's encoder wheel's resistance is controlled with a magnetic clutch that ensures good tactile feedback.

» Given the reasonably good acoustic starting point I wasn't expecting miracles from the Trinnov, but it didn't take much listening to appreciate its significant benefits. The low end was noticeably smoother and usefully more extended, but also more precise in its timing and control. Stereo imaging was even sharper and more stable than before, and reverberation seemed more 3D and with greater depth. Bypassing the Optimizer didn't leave me wondering how I could cope without it, but I definitely preferred to mix and manipulate sound with the processing active!

As a monitoring controller, the La Remote works very well indeed and I particularly liked the ability to place functions on different layers to suit my own ergonomic preferences and workflow. Physically, the unit is practical and easy to use, the magnetic clutch on the large encoder giving a very pleasing tactile response. However, while the operation and configurability are perhaps 90 percent of the way to perfection, there are some important omissions in the present software that preclude my unrestrained praise.

For example, in a stereo monitoring system it's possible to create downmixes to generate 'mono on left only', or 'on both' speakers, and even to create a stereo difference signal and to swap left and right channels. But I was unable to configure a simple right-channel polarity inversion and, while it is technically possible to introduce a static level change to any input signal via the hardware configuration menus, I couldn't find a way of doing that in real time from La Remote to allow a level trim to be applied to a source. This kind of facility is essential when comparing a reference signal to a DAW mix to rule out preferences due to overall loudness differences, and a common feature of high-end monitor controllers. I raised these points with the Trinnov team and was told that they are already on the scheduled development list. Hopefully, they will be implemented fairly soon in a future software update as I see no physical reason why they can't be introduced fairly easily.

Lasting Impression

With those updates and extra features in place La Remote would match the functionality and capability of the best high-end stereo and surround/multichannel monitor controllers currently on the market, and that presents an intriguing quandary. While the D-MON system might initially appear expensive, the introduction of the La Remote controller and its comprehensive monitor control features actually tips the balance quite positively in Trinnovs's favour.

For example, a D-MON|6 with La Remote and calibration mic costs around £7740 (including VAT) here in the UK. In contrast, the high-end Cranesong Avocet Surround monitor controller, which has broadly similar monitor controller functionality and audio quality, costs around £7200. However, the latter has no speaker/room-correction capability, of course, whereas the Trinnov system does, for just an extra £540. Adding Trinnov's superbly effective and sophisticated speaker/room-correction processing to a wonderfully capable and configurable monitor controller for that little feels something of a bargain to me!

Unlike typical single-capsule measurement mics, the microphone used in the Trinnov system is a tetrahedral array which allows the speaker positions to be calculated in 3D space.

I enjoyed using the D-MON and La Remote system. It was simple to integrate into my monitoring chain, relatively fast to configure (with the benefits of remote interaction if necessary), remarkably versatile and customisable, and it improved the accuracy and resolution of my already pretty good monitoring chain significantly. This is a highly impressive system which I can recommend equally highly for anyone wanting to extract the last few percent of sonic perfection in a high-end monitoring system.

$ La Remote only $990. D-MON bundles with 3D mic and La Remote: D-MON|6 $8724.40; D-Mon|12 $13,200.

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Roland TD-50KV2

Electronic Drum Kit

Is Roland's new flagship TD-50KV2 the best electronic drum kit you can buy?

MARK GORDON

Roland have certainly been busy these last few months with what seems like a continuous stream of new electronic drum kits, featuring a variety of innovative technologies. The latest addition to the range is the very impressive TD-50KV2, the new flagship in their electronic drum range.

Weighing in at a hefty $7750, the TD-50KV2 brings together Roland's existing digital products (the PD-140DS digital snare and CY-18DR digital ride) and combines them with a brand-new digital hi-hat, all-new rack system and upgraded version of the top-of-the-range TD-50 module. Does electronic drumming get any better than this? Let's dive in and find out...

New Kit In Town

The kit comprises two 10-inch PD-108 BC rack toms, two 12-inch PD-128 BC floor toms, the KD-180 bass drum and the PD-140DS digital snare. Cymbals are provided in the form of two 16-inch CY-16R-T thin crash cymbals, the 18-inch CY-18DR digital ride and the brand-new VH-14D digital hi-hat. Pads and cymbals are mounted on the new MDS-STG2 system, which is an absolute beast of a rack that includes a range of heavyweight chrome clamps, tom arms and cymbal stands, plus an internal wiring system for the loom. The final piece of the jigsaw is the TD-50x module, an upgraded version of the highly regarded TD-50 that has been at the heart of Roland's flagship kit for the last five years and now includes a host of new functions and features.

The TD-50KV2 is almost a Frankenstein kit, all its parts except for the new VH-14D digital hi-hats and MDS-STG2 rack already being included in kits within the current Roland range. The PD-108 and PD-128 tom pads have been around since 2012 and are currently part of the TD-30KV kit but are still great-looking, top-of-the-range pads that include a new rim sensor to accurately detect the depth of rim shots. All four tom pads sport two-ply mesh heads that can be fully tensioned via the usual six tension lugs and come with a very cool black-chrome wrap.

The KD-180 18-inch kick drum was released in 2018 and is the smallest of the 'full size' bass drums Roland offer. It features a standard all-birch acoustic drum shell with what is essentially the trigger part of a Roland KD10 drum pad mounted in the centre of the batter head and is also finished in the same matching black-chrome wrap as the toms. The CY-16RT crash cymbals are part of the new thin range (denoted by the T) and feature dual triggering via the bow or edge, as well as edge sensors, enabling the cymbal to choke when grabbed. They are 40-percent thinner than previous models, which enables them to flex more and gives them a more natural feel when struck.

The Digital Divide

With all the analogue pads covered, we can move on to the digital elements of the kit. The PD-140DS digital snare and CY-18DR digital ride are included in the VAD506 kit I recently reviewed in *SOS* (November 2020) and have also been part of the existing TD-50 kit range, but their innovative and ground-breaking features certainly qualify them for another mention.

Dispensing with the standard quarter-inch jack connectors, the digital pads connect to the control module via USB cables, which not only power the on-board processing of each pad but also enable them to transmit far more information about when, where, and how hard they have been hit.

The PD-140DS snare looks, to all intents and purposes, like a standard 14-inch metal snare. Its tubular lugs and polished chrome finish wouldn't look out of place on a high-end acoustic kit. It's also incredibly heavy! Under the three-ply mesh head are four cone triggers that supply the module not only with data on how hard the drum is being hit, but also with positional information. This allows the drum to respond more like an acoustic

Roland TD-50KV2

$7749

PROS

- Stunning new sounds.
- Almost endless editing options.
- Digital Hi-Hat is the closest thing to playing an acoustic hi-hat so far.
- Extra USB audio channels.

CONS

- Very expensive.
- No hi-hat bell zone.
- Hi-hat and snare stand not included.

SUMMARY

Electronic drumming doesn't get much better than this. The TD-50KV2 combines existing top-end features with new and innovative technology that sets the bar higher than ever before.

snare, and to vary the sound depending on where the drum is struck.

Also impressive, and very usable, is the PD-140's method for dealing with cross sticks — a sound achieved by placing your hand on the drum head and hitting only the rim with the stick, producing that classic 'click'. On an acoustic drum, this happens naturally, but on a drum that is triggering samples it's not quite so simple. Similarly to the way your finger works on a touchscreen, the head of the PD-140S responds to the static electricity in your hand when it's placed on the head of the drum, instantaneously switching the rim sound to a cross-stick sample. This really is a revelation for an electronic drum. Previously, the options were limited to selecting a dedicated cross-stick preset (forfeiting a rim-shot option), pressing a modifying button or playing a dedicated part of the rim.

In theory, it's also possible to use brushes by selecting the appropriate option in the TD-50x module. I found that brushes worked well for striking the drum, but the 'stirring' techniques associated with more jazzy styles weren't interpreted as well. There may be something I'm missing, but this was also something I noted using the PD-140DS with the TD-27 module. I added the caveat, in my review, that if you primarily play in that style I'm not sure you'd be choosing an electronic kit, but I had hoped a top-of-the-range snare/control module combination like the PD-140DS and TD-50x, with all its digital functionality, would be a little more impressive in this department.

At 18 inches across, the CY-18DR isn't far off the size of a typical acoustic ride cymbal, which really enhances the playing experience. Being digital, the cymbal incorporates multiple sensors, including three bow sensors, a bell sensor, an edge sensor and a touch sensor. The three bow sensors allow for a degree of positional sensitivity that means you can play across the cymbal, from bell to edge, and the sound will change in tone and pitch accordingly. The sensor in the bell

The TD-50x measures a substantial 330 x 255 x 118 mm...

is able to differentiate between the tip of the stick and the shoulder, which means that you can employ the same playing techniques you would with an acoustic ride cymbal. In addition to the choking feature found on the two crash cymbals, the CY-18DR enables the same trick the PD-140S snare is capable of, allowing you to deaden the cymbal by simply placing your hand, or even a finger, on its surface. You can also use this feature to 'play' a deadened cymbal sound, allowing for yet more acoustic techniques to be incorporated into your playing.

The new kid on the digital block is the VH-14D digital hi-hat. Utilising a two-pad design to mimic the top and bottom cymbals of an acoustic hi-hat, the 14-inch pads fit on a regular hi-hat stand (not provided as part of the kit) and function exactly like their acoustic counterparts. Made of thin rubber and almost 2 inches larger than the analogue VH-13 hats, the VH-14D feels very natural to play. Similar to the CY-18DR ride, the VH-14D includes multiple bow and edge sensors in the top cymbal, to accurately reflect where the pad is struck. There is no dedicated bell sensor but the bow sensors allow for the sound to change as you play across the cymbal, with the bell area offering a darker tone. The additional information available in a digital unit allows for greater resolution in terms of the open and closed position of the hi-hat and vertical motion of the cymbals when using the hi-hat pedal. This makes the hi-hat incredibly articulate to play and extremely sensitive to any foot 'splashes' from the pedal. As you squeeze the hats tightly closed, the pitch of the cymbals increases, exactly as you would expect from an acoustic hi-hat. The muting facility featured on the CY-18DR by placing your hand on the surface of the cymbal is also offered by the VH-14D, and can be used to great effect to recreate jazz hi-hat playing techniques. The size of the hi-hat and its jaw-dropping realism make this probably the closest thing to playing an acoustic hi-hat you could imagine.

Control Centre

With everything set up, we can move on to what is the meat in the drum sandwich — the TD-50x control module. The TD-50 has been Roland's top-of-the-range module for some time now and the new TD-50x looks remarkably similar. In fact, it looks identical... and that's because physically they are the same module. All the new stuff is hidden inside, which happily means that existing TD-50 owners can upgrade to the TD-50x feature set for $199 through Roland Cloud Manager, the software-based interface for Roland Cloud content.

In stark contrast to the minimalist look of the TD-27 module included with Roland's most recent VAD506 offering, the TD-50x isn't short of knobs, buttons and sockets, which gives it an almost retro look. Having dedicated buttons for most of the functions does cut out a lot of menu scrolling, which is great for quick access in, for example, a live situation.

The large LCD display is surrounded by five familiar 'soft' buttons whose function is dependent on what is displayed on screen. The five buttons are augmented by three illuminated rotary knobs below the LCD for changing values in the lower

Rack 'Em Up

As I mentioned in my introduction, the MDS-STG2 rack is a substantial piece of kit and looks more than capable of withstanding the rigours of live touring... or pretty much anything else you could throw at it. The curved horizontal pipes of the three-sided rack are attached to the legs using a selection of heavy-duty clamps, and everything, including the clamps, is finished in chrome. The loom for connecting the pads to the module is integrated into the horizontal poles to hide it from view and preserve the clean lines of the kit. Even the cables themselves have a silver external coating, so that they blend in visually. The cymbal boom arms and tom holders can either be attached to the rack via clamps or directly into the legs of the stand which, in combination with the ball joints featured on both, allows for maximum flexibility when setting up the kit. Roland still do not supply a hi-hat stand or snare drum stand with the TD-50-KV2, which surprises me considering the cost of the package, and the fact that absolutely everything else you need, including a drum key, is supplied.

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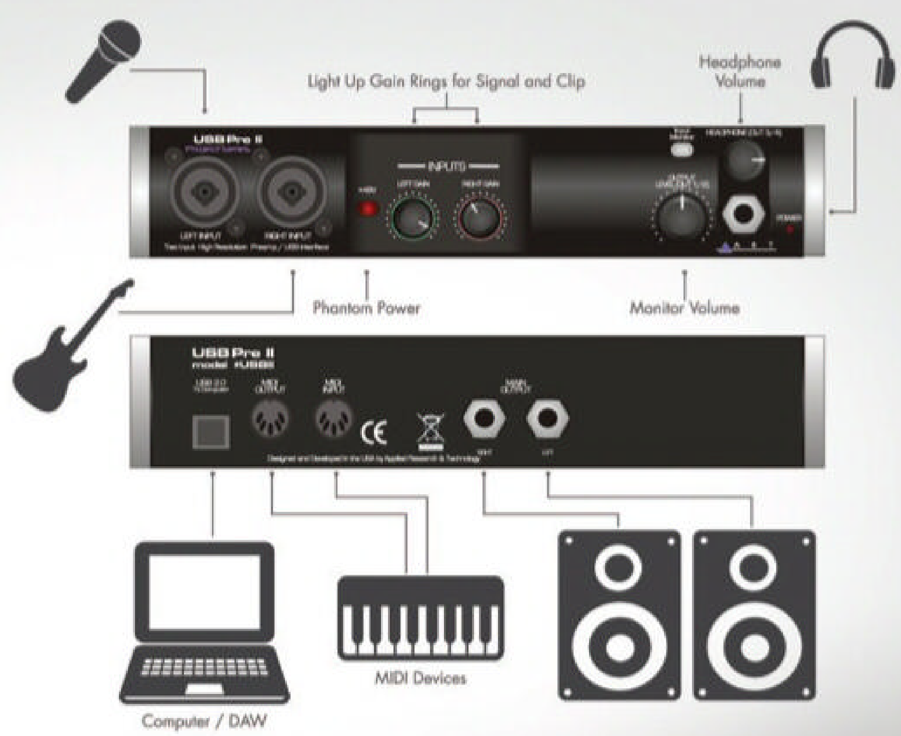

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... which is good because it needs to fit all of this on the back panel.

part of the display. Along with the page up and down buttons, the four-way cursor buttons and large data-entry knob are also used to navigate the display and change parameters. Two large +/- buttons can be used to increment through the preset kits or for data entry, and a fully velocity-sensitive Preview button auditions sounds. This can be used in conjunction with the Lock, Rim and Pad Select buttons to easily configure kits and edit instruments directly from the module, with no pads attached.

Below the display are eight faders dedicated to kick, snare, toms, hi-hat, crash, ride and percussion (Aux) instrument levels, plus the overall ambience level. To the right of the display, Instrument, Ambience and Mixer buttons take you straight to their respective screens. There are also individual buttons to access the SD card, Trigger settings and Set Up pages. Should you get lost in a menu somewhere, a large white Kit button takes you back to the main kit screen and illuminates to confirm its status. Finally, there are individual controls for master output levels and phones, along with separate volume knobs to control the Click level, Song playback and level of external devices connected to the Mix inputs.

The rear of the module is almost overflowing with sockets. Fourteen trigger input jacks are provided for analogue pads and cymbals, and are augmented by three USB sockets for connection to the digital snare, ride and hi-hats. There are enough analogue trigger inputs to accommodate several additional pads and triggers if required — over and above those provided in the standard TD-50KV2 kit. As this is very much a professional device, the stereo Master output is provided in the form of balanced XLR and unbalanced quarter-inch jack sockets. A further eight balanced jacks add dedicated outputs for routing discrete instrument signals to an external mixer, typically a front-of-house desk in a live setup. A quarter-inch jack Mix In socket can be used to bring in external sound sources such as an in-ear monitor feed or output from another module. This is mirrored by a 3.5mm mini-jack socket on the front of the unit that is suitable for MP3 players. The input level from both sockets is controlled by the Mix knob on the front panel.

The TD-50x features two headphone sockets (quarter-inch jack and 3.5mm mini-jack), MIDI In and Out sockets, and a USB port for connection directly to a Mac or PC. Next to the USB socket is an SD card slot for importing user samples into the TD-50x, and for saving kits and exporting song data created using the internal recording feature. I've yet to work out Roland's logic on whether they choose SD card support or USB thumb drive support on their various modules — there seems to be a fairly random mix of the two across the range of available electronic drum products.

More USB Than You Can Shake A Stick At

The TD-50 module supports 10 audio channels over USB, while the new TD-27 offers 28 channels. The TD-50x, being the flagship, increases this to a massive 32 channels, which should be enough for any situation. What this means in practical terms is that if you connect the TD-50x to your Mac or PC via USB you'll be presented with 32 discrete audio channels in your DAW. The TD-50x's routing options enable you to direct all of the individual instruments to their own audio track via USB, and also to route the ambience independently, should you want to record that on its own track. The high channel count and routing also allows for playback of audio in your DAW to any of the direct outputs, meaning that the TD-50x can work as an eight-output USB interface. MIDI is also supported via USB, so with a single connection you can record and play back an entire performance in both MIDI and audio form.

Bang Up To Date

Although I've not used the original TD-50 module, I have reviewed the previous flagship TD-30 and the new TD-27 module, so I had a pretty good idea what to expect from the TD-50x.

The headline feature for the new module (when compared to the original TD-50) is the increased number of internal sounds, up from 422 to 961 — more than double the number of sounds, so a pretty impressive upgrade. There are now 40 acoustic kick drums, 95 'processed' kicks, 35 acoustic snares, 215 percussion sounds... and, as you can imagine, far too many more to list here. However, of particular interest to existing TD-50 owners is the increase from only two ride cymbals to the much more interesting total of nine.

If you're familiar with any of the Roland drum modules, you'll know that the level of editing possible reaches quite fantastic levels of detail. When you can adjust everything from head type, muffling and drum size to mic position, number of snare wires and cymbal thickness (plus a lot more), even 400 instruments would present almost endless options, so increasing the palette this much really does give you an incredible resource to work with.

As a result of this expansion, the TD-50x now includes 70 preset kits (up from 55), which perfectly showcase the

new sounds and additional functionality of the module. The level of realism and playability is incredible, from the solid and warm sound of 'Acoustic #1' through to 'Poppin Walnut', 'Swing Jazz' and the classic electronic sounds of 'FAT808' and 'Hybrid909'.

In truth, it's difficult to highlight particular presets, as they are all absolutely stunning, and, more importantly, very usable. This is in part due to the new Ambience algorithms, whose parameters can be accessed directly from the Ambience button and controlled by the Ambience fader. This aspect of the TD-50x now combines the Room and Overhead Mic settings to create an overall ambience that can be applied to the whole kit, or elements within it via individual sends. Again, the level of editability is extensive and detailed, including mic type, mic position, compression and EQ. Room types include Small and Large Studios, Halls and Stages, in addition to favourites from the TD-50 such as Garage, Beach and Gymnasium. Regular reverb is also catered for and now includes presets from the classic Roland SRV2000. A handy preview button lets you listen to only the ambience as you edit, should you want to isolate that element.

Roland drum modules have always been big on effects, and the TD-50x increases the number of on-board effects available in the three effects processors from 30 to 38. New additions include a speaker simulator and guitar amp simulator plus virtual recreations of the classic Roland SBF-325 Flanger and SDD-320 Dimension D. A Time Control Delay has also been added that enables you to change the delay setting over a specified time period. Delays, flangers and speaker simulators are possibly not the most inspiring enhancements the TD-50x has to offer but when you already have 30 effects at your disposal, what else do you need? Well, the answer may be hidden away in the Master Effects section, where the very tiny addition of a Mix parameter in the compressor settings means that you can create parallel compression, a popular and very handy production technique enabling you to blend compressed and uncompressed signals together to get exactly the sound you're seeking. The Master EQ can also now be switched to Mid/Side processing, which facilitates another quite specific engineering technique, used to process the sounds at the centre of a stereo field differently from the left and right stereo extremes. Perhaps neither of these are going to be at the top of most people's wish lists, but they really highlight Roland's phenomenal attention to detail and the processing power of the TD-50x.

I've mainly focused on what's new or different in the TD-50x compared to the TD-50, but of course everything that was available in the TD-50 is still there, such as the ability to import your own samples and layer them with existing instruments, and the facility to import songs as MP3 or WAV files for backing-track playback. You can still edit any instrument to within an inch of its life and tweak transients and mic positions until the cows come home. If you've played with a Roland drum module or read any of my previous reviews, you know you'll not be short of parameters to adjust — I could literally take up the whole magazine going through the various options. I will specifically mention one, however: the snare drums in the TD-50 have always boasted a huge number of editable parameters, such as three different snare wire types and their tightness on the drum, from choked to loose and buzzing. The TD-50x now gives you the ability to turn the snare wires off completely, which adds a new character to all the acoustic snares.

A number of other small, 'under the hood' changes in the TD-50x, in areas such as layering and sample import, have simplified the workflow but, handy as they are, they are most likely to only be appreciated by a minority of power users.

It's About Time

No matter what playing level you're at, a bit of training will always help sharpen you up. For me, Yamaha are the masters of built-in coaching features, but Roland have upped their game in the TD-50x by adding two new rhythm training routines in addition to the Quiet Count mode from the TD-50. Quiet Count will play a click for a specified number of bars, then mute the click for a specified number of bars, helping you to develop your own internal sense of time, as the aim is to still be playing in time with the click when it is unmuted.

The new Time Check mode is designed to analyse whether you're playing behind, ahead of or on the beat. You can select two pads to be 'active' (typically kick and snare or hi-hat and snare) and, as you play to the click, the TD-50x will display whether your strikes are hitting the spot. You can specify the number of measures over which you'll be tested, and also the strictness of the scoring.

Finally, Warm Ups combines three exercises: Change Up, where the play-along rhythm type changes every two bars; Auto Up/Down, where the tempo increases by 1bpm per minute and a revisiting of the Time Check mode. Over five, 10 or 15 minutes you run through each of the three exercises successively and receive a final score at the end.

Hit Me With Your Best Shot

Having reviewed a number of Roland electronic drum kits over the years, I think it's safe to say they are masters of the craft. This is their absolute top-of-the-range kit, featuring the latest digital pad technology, and anyone who buys it is going to end up with a setup that is hard to beat on almost every level. The TD-50 was already an exceptional control module, and the TD-50x adds a host of very useful and usable features — as well as doubling the number of instruments. The level of editing and sound manipulation possibilities border on the intimidating, but the kits also sound incredible straight out of the box, so I guess that's the best of both worlds! The fact that this is also a downloadable upgrade for existing TD-50 owners is a real bonus.

The PD-140DS digital snare and 18-inch CY-18DR digital ride have been around for a while now, but still manage to impress, and the addition of the VH-14D digital hi-hat ups the game another notch. At around $1000 (if bought separately), it does cost more than any regular hi-hats I've ever seen but, having said that, it sounds and plays incredibly well and is the closest thing to playing acoustic hi-hats that I'm aware of. Having said that, if you're buying the best digital hi-hat in the world, a bell sensor might have been a nice addition.

At this price, the TD-50KV2 won't be for everybody — in fact, I imagine it won't be for the vast majority of people. However, if you're in the market for what is pretty much the best of the best in electronic drum kits, from the industrial-quality rack system through to ground-breaking digital hi-hats, the TD-50KV2 sets a very high bar in electronic drum excellence.

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Analogue Solutions Leipzig V3

Analogue Synthesizer

Analogue Solutions have taken a perfectly good synth... and made it better.

WILLIAM STOKES

Back in May 2012, Paul Nagle asked in his *SOS* review of Analogue Solutions' Leipzig-S if it was the British company's best synth yet. The twin oscillator mono synth reminded him, he said, of a Moog Rogue, "but a Rogue force-fed on burgers and lard before being squeezed into a rack." Tasty. Now Analogue Solutions return with a third synth in the Leipzig dynasty, the V3. Gone are the rack ears, new features have been added and a few modifications have been made under the hood to improve reliability and performance. Conspicuous additions since the Leipzig-S include a headphone output and a line of 3.5mm inputs and outputs down the left-hand side, for integration with a modular system or external audio sources. Longtime Leipziggers will be pleased to know that amid all of this — plot spoiler ahead — it retains that signature filthy Leipzig character, which Analogue Solutions describe accurately as a "wonderful, angry, pure analogue sound". Or, as Paul wrote with equal accuracy, "a brash bulldozer of a synth." For Paul to float such an accolade about the Leipzig-S is noteworthy: Analogue Solutions may not have the corporate clout of some synth companies, but their creations go toe-to-toe with even the most ambitious designs from any company in the world. Just look at their Colossus: an astonishing £26,000 12-oscillator cross between a synth and a piece of furniture, with twin on-board pin matrixes and two spring reverbs. The Leipzig V3, then, is a synth of considerable pedigree, and this only bolsters its appeal as a compact workhorse of a synth.

Frills, And The Lack Thereof

It might actually be worth holding the aforementioned Moog Rogue comparison in the back of your mind for this review — that synth, like this one, represents something I respect greatly: a developer's decision to ditch the frills and create something simple and powerful. Like its predecessor, the Leipzig V3 ditches more frills than most in this respect: it has no screen, no presets, no auto-tuning,

no USB compatibility and no MIDI out. Each oscillator toggles between just two waveforms, saw or pulse-width-variable square, there's a lovely Moog-inspired ladder filter, a simple modulation matrix, a pair of ADSR envelopes and a VCA. It is, suffice to say, very analogue indeed.

I'll say at this point that the raw sound of the Leipzig V3's oscillators are simply a joy to behold. Rich and full of character across their entire pitch range and drivable at high levels (along with the filter) I could achieve anything from deep and clear bass stabs to reedy '70s fusion jazz leads without so much as touching an envelope or modulation generator. Many of the V3's circuits are indeed designs dating back to that era, so there is — dare I say it — a sense of authenticity in its sound. The other side of the coin here, of course, is that there are no CPUs on board to stabilise the V3's circuitry, so the oscillators are naturally prone to drifting, though in this case not badly. Another all-analogue quirk concerns note priority: the V3's signal path prioritises the last note played, but will forget the previous note. In other words, if you hold one note and press another momentarily the VCA will shut when the last note is released. In practice this can mean the V3's sound can cut out unexpectedly if you happen catch an accidental note while playing, though this is not so much a fault as part of the wondrous territory of a fully analogue signal path.

One frill I could have done with is slightly better build quality on the knobs, which feel a little plasticky and light to turn, unlike the weighty and smooth pots one might expect to find on a synth carrying the V3's price tag. It's not just a tactile preference: I'm of the view that heavier knobs make for better parameter control, especially in a live situation, and also tend to last longer. This said, the overall build quality of the V3 can't really be faulted. Its steel and aluminium chassis is certainly — as manufacturers love to say — 'rugged', and judging by the physical heft of the original Leipzig, it's clearly a fruit of Analogue Solutions' labour to have compacted it all into such a neat desktop format.

In Use

It didn't take me long to deduce that despite appearing rudimentary on first glance, the V3's circuits have been carefully designed to interact with each other in very interesting and occasionally wonderful ways. VCO 2, for example, can be toggled to hard-sync to VCO 1, it can be used as a modulation source for some intense FM-style sounds, or it can do the complete opposite and latch to one pitch regardless of what note messages the synth is receiving. In this instance the Leipzig either becomes a kind of quasi-paraphonic synth, with a steady note held beneath whatever melody is being played, or it opens up a fixed-frequency, audio rate modulation source for even more tone shaping. Add some oscillator-specific modulation into the mix, maybe from an envelope, and all sorts of sonic possibilities present themselves. In the modulation matrix, each oscillator and the filter have a selection of different modulation sources beyond the conventional option of the LFO's two waveforms, so there's no shortage of room for creativity.

This sort of thing goes way beyond just the tonal, of course. The Leipzig's snappy envelope, rasping oscillator drive and miaowing pulse-width modulation make for a highly capable percussion instrument, aided further by an on-board noise generator — which can also toggle to receive an incoming external signal and feed it through the V3's VCA. I found it fairly intuitive to create some very aggressive, industrial-sounding beats; normally I'd fine-tune sounds one by one and build a sample bank; while this is always an option, the sheer amount of modulation possibilities on the Leipzig mean that masses of variation can be achieved across one running sequence.

The Sequencer

Speaking of which, if there is a function with which the V3 chooses to really push the envelope, it's the sequencer. Like the Leipzig-S, the on-board eight-step sequencer can be dialled in to influence either of the oscillators or the filter via dedicated knobs, but while the -S offered a wide range of sync options, here those have been reduced down to a choice of the LFO as clock, MIDI sync or an external CV clock via the patch panel. No complaint there. Simplicity is bliss. With the three destination knobs turned up it's a recipe for utter analogue wildness, and compounded with — you guessed it — any of the V3's other modulation sources it's a voyage.

»

Analogue Solutions
Leipzig V3
$1199

PROS
- Incredible-sounding oscillators, capable of a huge sound.
- Highly innovative use of the twin-oscillator monosynth format.
- Compact footprint.
- Easy to integrate with other instruments via MIDI or CV.

CONS
- The sequencer takes a little getting used to, particularly the 'hidden' 16-step function.
- Build quality on the knobs could be better.

SUMMARY
The Leipzig V3 is a worthy development from the Leipzig-S, offering a hugely versatile and powerful analogue synth in a compact package.

Along with a number of subtle front panel adjustments, the main difference between the Leipzig S and the V3 are the loss of the rack ears and the addition of patch points on the left-hand side.

A curious feature beneath the sequencer's tactile eight steps is a 'hidden' 16-step sequencer, which automatically remembers the last 16 notes played manually into the V3 and plays them back when the transport is started — even if all three of the destination knobs are on zero. This, I have to say, could use an off switch. I can imagine plenty of instances where such a function would be a great asset, but it would only increase the usability of the sequencer if it could run with no consequence until one or more of the destination knobs are turned up, as happens with the LFO. It's not possible, for instance, just to sequence the movement of the filter while manually playing a melody, nor is it particularly easy to program or adjust a melody while the sequencer is running. With the sequencer stopped, however, inputting notes step-by-step isn't too difficult.

For my qualms, I should point out that the V3's manual encourages users to think of the sequencer as a modulation source that can also be used for simple melodies, and not the other way around. It also really does come alive with an external clock source, in turn freeing up the LFO for further delights. Overall it's fair to say the sequencer just isn't designed to be used simultaneously with manual playing or for precise melodies, instead demonstrating a totally different side to this synth and offering something like two instruments in one. It is, as the V3's manual describes it, 'the fun bit'.

A novel function on the sequencer is the 'VCO rhythm' button: another very clever quasi-paraphonic feature which allows VCO 2 to be ducked from specified steps in the sequence. Simple in theory, but in practice something of a master stroke. You could choose to accentuate certain steps by adding another oscillator, or — with VCO 2 set to modulate VCO 1 — mangle particular steps beyond recognition. You could overlay a simple cross-rhythm, create dynamic percussive sequences and much more. Excellent stuff.

"The Leipzig V3 shines not because of how many different circuits it boasts, but because of how it chooses to use those circuits."

Conclusion

The Leipzig V3 shines not because of how many different circuits it boasts, but because of how it chooses to use those circuits. In this way it is a synth capable of far more than the sum of its parts, pulling no punches in squeezing the most out of its compact footprint and promising almost limitless applicability. With such variety in the analogue synth market nowadays, as well as countless reboots and clones of older models coming out every year, it's a challenge to make a splash in the monosynth department in 2021. Analogue Solutions have doubled down on the integrity of their Leipzig series, choosing not to add legions of mod cons but instead make careful adjustments with the sole purpose of creating a better instrument that does very well what it is designed to do. In this endeavour they have succeeded.

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Soniccouture Threnody Strings

Sample Library

Soniccouture's new library steps into a strange new world of sci-fi and horror.

DAVE STEWART

The modern media composer can no longer get by with a palette of standard sampled orchestral articulations. Aleatoric effects, sound design and non-traditional playing techniques now play a major part in TV and film soundtracks, and sample libraries have evolved to keep pace. Consequently, most orchestral collections now include unconventional musical effects and processed sounds as a matter of course. Having been in the vanguard of innovative sampled instruments since 2005, UK company Soniccouture are well placed to get in on the act, and their new strings library looks set to become a significant player in the field.

Named after Penderecki's nerve-shredding lament for the victims of Hiroshima, Threnody features the Budapest Art Orchestra's full-blown string section of 16 first violins, 14 second violins, 12 violas, 10 cellos and eight double basses performing 26 avant-garde articulations devised by Soniccouture's Dan Powell with assistance from composer and orchestrator Felipe Téllez. The 60 players were recorded across their entire range using over 30 microphones which were subsequently mixed down to main (Decca tree), spot and ambient stereo pairs.

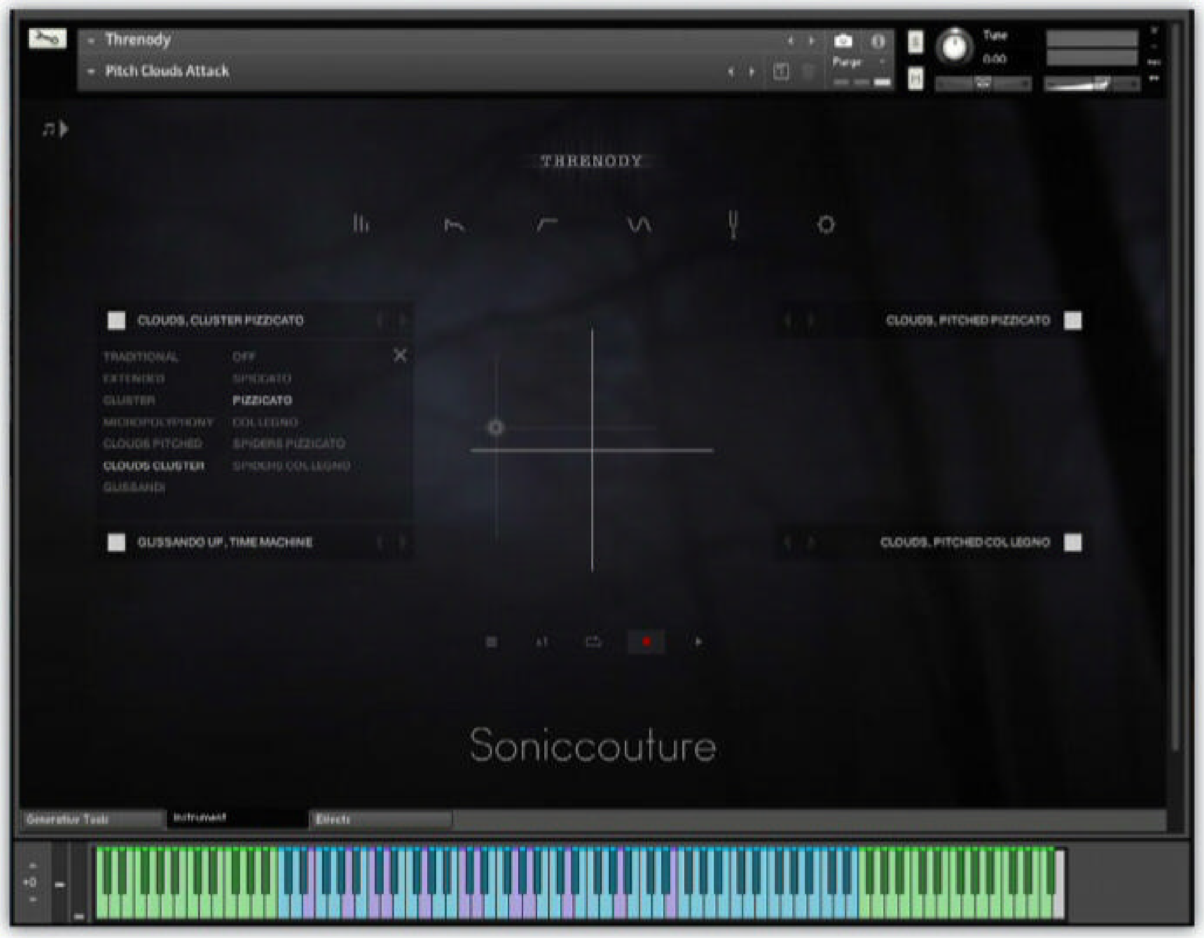
Also included are 35 sound design presets, a cluster generator, a micro-tuning module and the entertaining creative tools Weaver, Jammer and Phraser, the latter now updated with new features (details below). The 12GB library (6.5GB installed with NCW lossless compression) runs exclusively on Kontakt 6 or Kontakt 6 Player version 6.2 or later, with Kontakt Player software available free from Native Instruments' website.

New Traditional

All articulations are presented as single full-strings presets with instruments mapped according to range over six octaves from C1 to E7. The sections are overlapped and nicely blended so there's no tonal discontinuity between registers. You can access additional high and low notes stretching far beyond the instruments' real-life limits, or stick within the safety zone by selecting 'authentic range' in the 'options' edit tab.

Before veering off into uncharted realms of madness and horror, Threnody provides a few conventional articulations suitable for general music production. The looped 'simple sustain' long notes are a chance to hear the Budapest musicians playing it straight: utilising a restrained vibrato, they create a strong, bright and positive section sound which leans more towards tough and heroic than romantically lush — unsurprising given the astringent tonal territory this library aspires to.

An interesting long-note variant, 'diagonal bowing' has the players randomly moving their bows up and down between normal and sul tasto positions.



Threnody's Articulation page, showing the 'XY Pad' four-dimensional mixer. The blue keyboard notes show the string orchestra's authentic range, with green notes indicating stretched samples extending far beyond the instruments' real-life range limits. Purple notes mark the instruments' open strings, useful to know for certain articulations which utilise open-string harmonics.

Soniccouture
Threnody Strings
£220

PROS
- An adventurous collection of atmospheric avant-garde string articulations.
- Styles range from subtle tension-builders to demented horror soundscapes.
- Played by a powerful 60-piece contemporary string ensemble.
- Also contains sound design presets, a powerful X-Y mixing pad, a cluster generator and a trio of generative tools.

CONS
- Not much good for scoring a tender love scene.

SUMMARY

Threnody explores the dark side of orchestral strings with an imaginative selection of nervy avant-garde articulations. Performed by 60 string players recorded in a Budapest scoring stage from three mic positions, the library's myriad creative features include a programmable four-way artic mixer, a cluster generator and several entertaining and inspirational generative tools. File under 'Sci-fi and Horror'.

Played with a slightly stronger attack than the simple sustains, this style adds timbral motion, extra overtones and a slight roughness to the sound. Conventional tremolos make the intense, dramatic shuddering we know and love, while the 'chaotic tremolo' artic features the players randomly changing their tremolo speeds, producing a superbly rich and vibrant ensemble timbre.

The close-to-the-bridge 'molto sul ponticello' bowing creates a fabulous thin, icy texture in the violins' upper register. At the opposite end of the pitch spectrum, playing chords in the basses' bottom octave with the 'ponticello tremolo' preset sets off a cataclysmic earthquake-like rumble, a scary foretaste of the disquieting material that lies ahead.

Fever Pitch

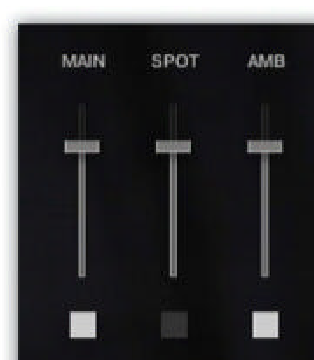
Threnody begins to show its true colours with the 'micropolyphony' artic. This replicates a compositional device developed by Ligeti, in which similar melodic lines played at different tempi overlap to create (in the composer's words) "a kind of impenetrable texture, something like a very densely woven cobweb." In Threnody the overlapping lines are atonal, unsynchronised and improvised from a fixed note selection within a small pitch range, producing a massed, concentrated flurry strongly reminiscent of the sound of buzzing insects. This unsettling style is thus a shoo-in for the soundtrack of any horror film with 'Swarm', 'Attack of' or 'Killer Bugs' in its title.

The sense of unease continues with the 'unstable pitch' preset, a nasty, queasy, menacing musical effect in which the musicians use an over-wide slow vibrato and play out of tune with each other. With its 'behind the bridge' performances, Threnody steps into a pit of horror: the basic arpeggiated style sounds like a screechy, industrial-strength version of the spooky waterphone effects heard in sound libraries 10 years ago, while the tremolo variant veers off into utter madness, a deranged, hellish cacophony calculated to cause consternation amongst listeners. Do not use these presets on a romantic ballad.

Gliss Time

'Harmonic glissandi' provide a welcome break from the insanity. Nothing new here — this effect (created by running a finger lightly up and down the bowed string) appeared in Stravinsky's *The Firebird* in 1910, but its eerie, disembodied slithering still sounds fresh and works splendidly for drones. The 'seagulls' version is a single, determined-sounding glissando slide down through the harmonic series of the string — ideal for scenes of alien winged creatures swooping from the night sky. Originally played on the open strings of the instruments, these artics are transposed to all pitches.

Threnody also contains conventionally bowed glissandi unlike any I've heard before: rather than the fast risers and downers found in other libraries, these glisses travel slowly and steadily through the entire range of the string orchestra, with instruments imperceptibly joining in or dropping out as the massed unison glissando passes through their pitch range like a non-stopping slow train. It takes around 70 seconds to rise from Middle C to the top violin note of E7, and just over 80 to descend from C5 (one octave above Middle C) to the basses' bottom note of C1. Astonishing, a tremendous tension-building dramatic effect. If you



The string players were recorded using over 30 microphones which were mixed down to main (Decca tree), spot and ambient stereo pairs.

want faster rises or falls you can engage Kontakt's Time Machine in the 'options' tab and speed up the glisses by up to 800 percent, though with an attendant drop in sound quality.

Clusters & Clouds

While cluster chords are a fairly standard orchestral library fixture, Soniccouture have dreamed up some cool variations. Microtonal tuning makes the familiar jarring groan of adjacent semitones that bit more discordant, and when you play two-handed chords on the 'cluster sustain' preset, the resulting obliterative, dissonant sheet of noise sounds less like music and more like the end of the world. The tremolo version adds a panicky agitation, while the cluster trills are a study in supernatural terror.

The library includes a suite of Kontakt effects.

So-called 'clouds' feature the orchestra playing rapid unsynchronised unison notes, creating a nominal dense cloud of activity. 'Pitched clouds' played in three different styles represent a return to normality: the spiccato version's energetic tremolo-like fast reiterations sound grandiose when played chordally, the pleasant thrumming of the pizzicato

Generative Tools

Soniccouture followers will welcome the inclusion of three creative tools developed by the company over the years. The 'Jammer' arpeggiator generates user-configurable, tempo-sync'ed patterns in a variety of styles, ranges and user-configurable scales. It can constantly generate new random note data in evolving sequences, or simply loop the last set of notes it created. Suffice it to say that within minutes of activating Jammer I was improvising keyboard solos over an urgent spiccato strings pattern it created, confirmation that this feature lives up to its name.

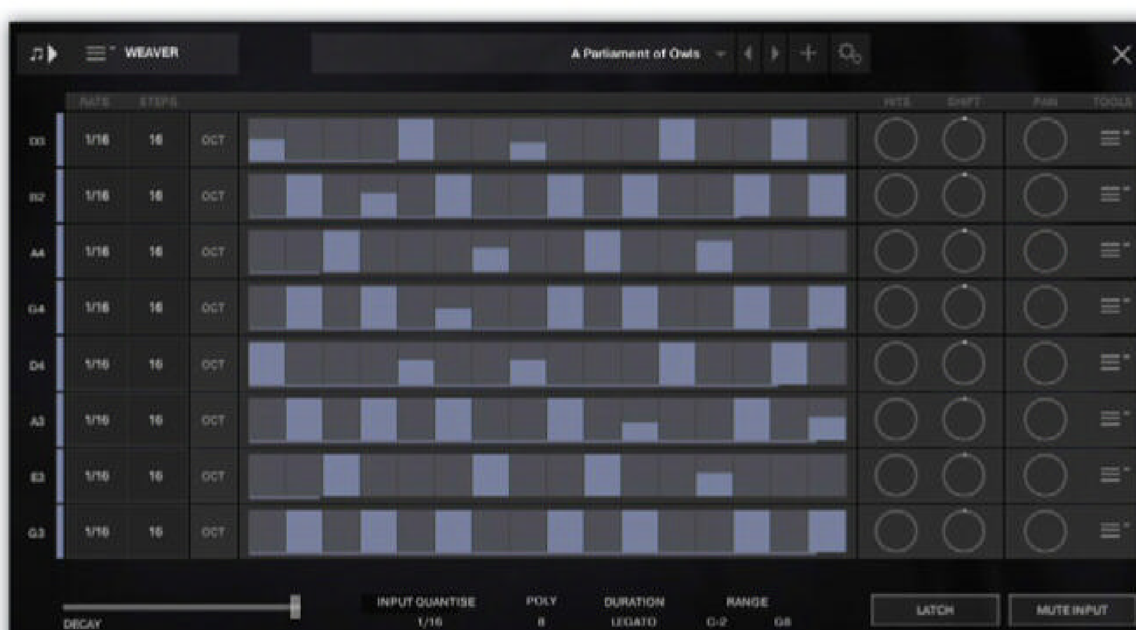
'Weaver' allows you to define a repetitive rhythm sequence which can be triggered by a single note or played chord. It can create DDL effects, pulses and user-defined polyrhythms of up to eight independent parts, and you can set the velocity and octave of each rhythmic step within the part. The display looks a little daunting, but don't let that put you off — once I'd figured out the visuals I managed to create some great, exciting rhythmic ostinatos.

Originally created back in 2008, 'Phraser' has been given a makeover. The idea is that you create a phrase by recording a series of single notes, then trigger playback with a MIDI note. Playing Middle C (MIDI note 60) triggers the original phrase; playing any other note transposes the phrase up or down, while playing a chord creates an instant parallel harmonisation. You can alter the playback rate, phrase start point and length, and reverse or randomise the note order. Lots of fun, and a good way of generating new musical angles from a simple string of notes.

Only one caveat: such generative tools generally work best with short, staccato source material, and Threnody's long, evolving samples are anything but that. However, I found that drastically shortening the sustain and decay of the 'pitched cloud' presets reduced the samples to single short events which sounded excellent when phrased, jammed or woven!

The 'Jammer' arpeggiator generates user-configurable, tempo-sync'ed patterns in a variety of styles, ranges and scales.

Soniccouture's classic 'Phraser' feature lets you create real-time phrases by recording a series of single notes, then triggering playback with a MIDI note.



Soniccouture's 'Weaver' feature can be used to generate polyrhythms of up to eight independent parts, a great way of creating exciting rhythmic ostinatos.

artic evokes the sound of massed tremolo mandolins, while the more percussive col legnos skitter like a shower of pitched raindrops in the high register. 'Cluster clouds' replicate the three artics with the pitches arranged in tight clusters as described above, adding a worrisome, neurotic edge to proceedings.

The final clouds effect is 'Spiders'. Ticking the 'aleatoric' box, each musician plays freely within a two-octave range in a random and unsynchronised manner. There are two types: played at high pitch, the 'spiders pizzicato' artic resembles the chatter of excited children, while down at the bottom end it sounds like an approaching avalanche. Switch to the col legno version, and the orchestra's cellos and basses now sound like the thunderous clacking of giant looms in some infernal Victorian cotton mill.

X-Y Pad

A major feature of Threnody is the X-Y Pad, a four-way virtual joystick which allows you to mix and crossfade up to four presets in real time. Having selected a preset for each of its four corners, you can use your mouse to move the central pointer around the quadrant: position it centrally and you'll hear an equal mix of the four presets, drag it into the bottom left corner and you'll hear only preset number one, and so on.

A simple usage would be to crossfade between two artics such as sustain and tremolo. The X (horizontal) and Y (vertical) controllers are pre-assigned to MIDI CCs 16 and 17, so with sustains loaded into the bottom left slot and tremolos in the slot above, you can automate crossfades between the two styles by recording MIDI CC17 data into your sequencer. Alternatively, you can use the X-Y pad's internal transport controls to record tempo-sync'ed dynamic movements over a specified number of bars.

The X-Y pad can also modulate different parameters such as pan position, filter cutoff and pitch, the latter a good way of creating psychedelic siren effects. I'm not sure what Ligeti would have made of it, but modern-day avant-gardists should have a field day with this powerful feature.

Score Draw

If you ask a group of musicians to improvise freely, the results are often less than scintillating. After an initial outburst of random squiggling, honking, scraping, squawking and bleating, the players tend to settle down and repeat themselves on subsequent takes, leading ironically to the hoped-for glorious pandemonium becoming a predictable set of repetitive aimless noises.

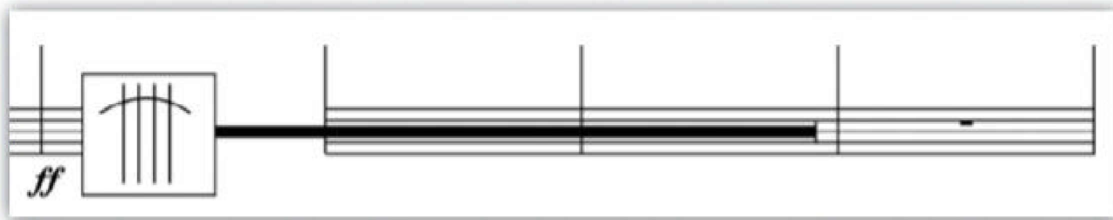
In order to lend purpose to improvisation, composers sometimes use graphic scores which combine standard notation with symbols, images and text. Needless to say, some avant-gardists went over the top and turned in scores consisting of a series of pictograms or abstract coloured brush strokes, techniques unlikely to generate a catchy tune. Meanwhile, those with a more precise musical focus have devised simple and effective notation techniques to convey the sound they're looking for.

Some good examples of this occur in the player scores used on the Threnody sampling sessions: the horror-film 'behind the bridge' bowing is indicated by a simple graphic symbol, while the range indicator and 'x' note heads used in the improvisational 'Spider' artic help shape the players' random performances.

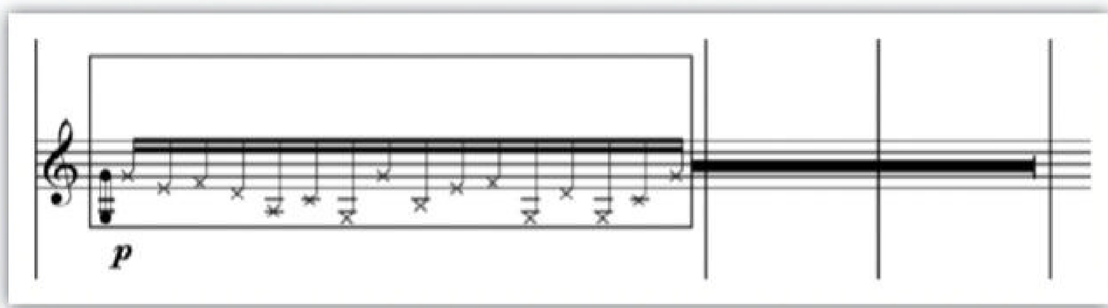
Soniccouture's Dan Powell explains the inspiration behind the articulations: "I began by going through a lot of my favourite 20th-century string pieces and collecting techniques I could perhaps imitate and then score for the entire orchestra. Composers like Penderecki (of course), Ligeti, George Crumb, etc. I took my rough score to Felipe (Téllez), who tidied it up and corrected my copious errors. Felipe also liaised with the conductor so that everyone had a good understanding of the kind of sounds we were aiming at recording."

Asked about the scores, Powell says: "We didn't always stay true to the original score notation. This is another area where Felipe's experience helped, finding the best way to convey the musical effect we wanted to the players. For example, in Penderecki's original Threnody, clusters are written as big black lines that change thickness. It looks cool, but is a bit vague. We scored more precisely, providing the musicians with the highest and lowest pitch for each cluster, strict tempo markings, etc."

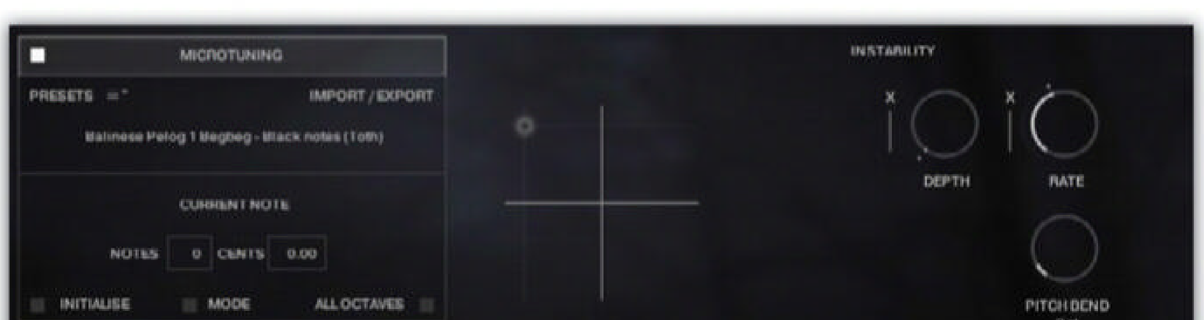
The scores also take into account the all-important click track: 'Four clicks free' indicates a four-beat count-in before the part starts, while a bar's rest separates one performance from the next, an essential precaution when editing thousands of samples. In the wider world of sampling, click-based recording ensures that crescendos and octave runs can synchronise and finish at the same time, not always the case in the days before 'the click' ruled the Earth!



This unconventional notation was used to tell the players to bow behind the bridge, which creates an uncontrolled, screechy fingernails-on-the-blackboard sound.



The so-called 'Spiders' articulation features each musician in the 60-piece ensemble playing freely within a two-octave range in a random and unsynchronised manner, creating a collective dense cloud of activity.



The tuning page includes microtuning presets and pitch instability controls which make each played note drift slightly in pitch — not what you want on a straight pop session, but good for horror and sci-fi scores!

Sound Design

Threnody's sound design presets give a good idea of its extensive sonic capabilities. 'Cluster Sequence', an automated slow, circular crossfade between sustain, tremolo and trill clusters

All 60 players of the Budapest Art Orchestra's string section.

and a micropolyphony artic, creates an accelerating build-up of tension which brings to mind an Arctic wolf pack circling a group of terrified explorers. 'Shiver' and 'Shutter' are great sci-fi effects, with the despairing falling sirens of 'Gravitational Arc of Minus Ten' recalling Vangelis' *Bladerunner* score. If you need shorter sounds for groove programming, 'Col Legno Percussion' shrinks the players' woody bow hits down to a set of small glitchy perc hits, while 'Weaver Shorts' is an excellent starting point for creating polyrhythmic synth sequences.

For those that enjoy tweaking factory presets, there are five parameter editors. The 'Mixer' tab contains the three-channel mic mixer and a main dynamic fader which defaults to CC1 (mod wheel) control. 'Amplitude' provides a standard ADSR envelope along with velocity, attack curve and expression (CC11) controls, while 'Filter' gives you a choice of 25 filter types and a 'Slim' control which thins the sound by removing some of the fundamental frequency of a played pitch, and is different for each note. The 'LFO' tab houses settings for the LFO's waveform, rate and depth, and LFO modulation settings for pitch, amplitude, filter cutoff and pan. The LFO is tempo sync'ed and can be set to different rhythmic values.

The 'Tuning' tab is where you'll find the cluster generator, which creates custom cluster chords of up to 12 detuned voices. Not to be confused with the real-life clusters performed by the string players, this is great for creating detuned clusters, but can also be adjusted to generate diatonic chord voicings. Also included here are a microtuning table and a terrific 'pitch-bend glissando' effect.

Remote Possibilities

Threnody's samples were recorded in Hungary, birthplace of the renowned composers György Ligeti and Béla Bartók. Since the sampling sessions took place during the January 2021 Covid-19 lockdown, this turned out to be quite a palaver.

Under normal conditions I imagine the Soniccouture team would have headed east from their London HQ, turned right at Poland and fetched up in Budapest where a large string orchestra awaited their arrival, but since international travel was off the menu other plans had to be drawn up.

Soniccouture's James Thompson (who designed the library's GUI and contributed to the sound design) explains: "The session was entirely remote, which worked brilliantly. The orchestra provided high-res audio streams and multiple camera feeds of the players and control room, as well as photos and video of the recording. They then uploaded the full Pro Tools sessions from which we could mix down. This was our first experience of working this way, but it went very smoothly."

Audiomovers software was used for high-res audio streaming, with the ubiquitous Zoom handling video and chat. Adding to the sinister atmosphere of this edgy library, the players wore masks; Messrs. Powell and Téllez looked in from Toronto while Thompson maintained a lonely vigil in London, occasionally taking a warming sip of whisky as the wind whistled through the deserted streets outside (OK, I made that last bit up).

Is this the future of orchestral recording? Long before the pandemic struck, UK studios were offering remote sessions with competitively priced European orchestras. What began as a cost-saving exercise is becoming the norm, but while we miss the buzz of the in-studio experience and the bonhomie of mingling with musicians, we should be grateful that technology now allows such projects to go ahead in the first place.

Conclusion

So, into the future with the past. A harsh critic might say a sample library inspired by late-20th-century avant-garde compositions is a bit passé, but from where I'm standing this looks like a new and welcome development in orchestral sampling.

Rather than a historical rehash, Threnody is a bold step forward: it extends the exploratory string writing techniques of yesteryear with atmospheric and powerful innovations which should catch the ear of anyone looking to create sci-fi and horror scores or experimental tracks. Uniting classic avant-garde techniques with fresh musical ideas and 21st-century creative tools, Soniccouture's first orchestral strings library is an unmitigated success.

$ £220
W www.soniccouture.com

Minimal Audio Rift

Distortion & Filter Plug-in for Mac & Windows

Play View provides a streamlined GUI with macro controls that conceal the complexity of Rift's powerful bipolar distortion processing.

JOHN WALDEN

If your sounds need a lift, perhaps you need Rift?

Whether you're working in mainstream pop, experimental electronica or contemporary film/TV scoring, some carefully chosen sonic elements can add an extra dimension to your production that helps you keep the listener's attention. One of the latest effects plug-ins designed with precisely this sort of task in mind is Rift, which developers Minimal Audio describe as a bipolar hybrid distortion effect. While the full price of Rift, Minimal's flagship product, means it's not in casual-purchase territory, the launch offer [*which is still current when going to press — Ed*] discounts it by 40 percent, and Rift Filter Lite makes its filter section available as a separate, more affordable plug-in that is free during the launch period.

As its tagline implies, Rift is built around distortion-based processing. The 'bipolar' descriptor is there because you get to apply different distortion processing to the positive and negative elements of your source audio's waveform. Some 30 different distortion algorithms are organised into five distinct categories, and these start with gentle soft clipping and move through to bit-crushing and sample-rate decimation, with multiple stops along the route. On paper, then, there's the potential here to create almost any type of distortion-based effect.

Rift For Dummies

Rift offers two modes of operation, courtesy of Play View and Advanced View, which are accessed using two buttons at the bottom-centre of the GUI. While many elements are common to both, Play View provides a simpler, macro-based control set, and is ideal when you just want to apply some Rift magic quickly, without having to dig too deeply into how it all works.

At the top of the UI, presets are browsable by category or style tags. Beneath this, Drive controls the input gain, which is used to 'push' the distortion processing and includes a 'x2' switch, while Output adjusts the signal level leaving the plug-in and includes an optional limiter, with two modes of operation (gentle and punchy). The Dry/Wet slider offers instant parallel processing, while Stages can be used to

add more stages to the distortion engine, for a stronger effect.

The Blend knob and Mode switch start to get us into the bipolar side of things. Hard or Soft Mode adjusts how the engine blends between the Positive and Negative distortion engines, while the Blend knob adjusts the point of transition between the positive and negative portions of the waveform, with the red line in the central Oscilloscope panel providing visual feedback. The Oscilloscope display also shows both the shape of the current distortion engines and, when audio is passing through the plug-in, the output waveform in real time. This display zone can be toggled to a Filter view, in which you can see what Rift's filtering is doing. To the left and right of this display, you can select the specific algorithm for the Positive and Negative distortion engines and modify the shape of each distortion response using a macro knob.

Finally, in the lower strip, alongside the input and output meters, there are two (per preset) pre-configured macro controls that target multiple parameters in the engine. These Macro knobs, like other controls in the UI, can be automated using your DAW. Each preset also includes pre-configured modulation; Rift's full modulation system is extensive, but if modest tweaking of presets is all you require, Play View's Macro knobs give you enough scope to do some fine-tuning with a minimum of fuss.

Rift For PhDs

The degree of both control and complexity are taken up a notch or three with the Advanced View's additional controls. At

Minimal Audio Rift

$129

PROS

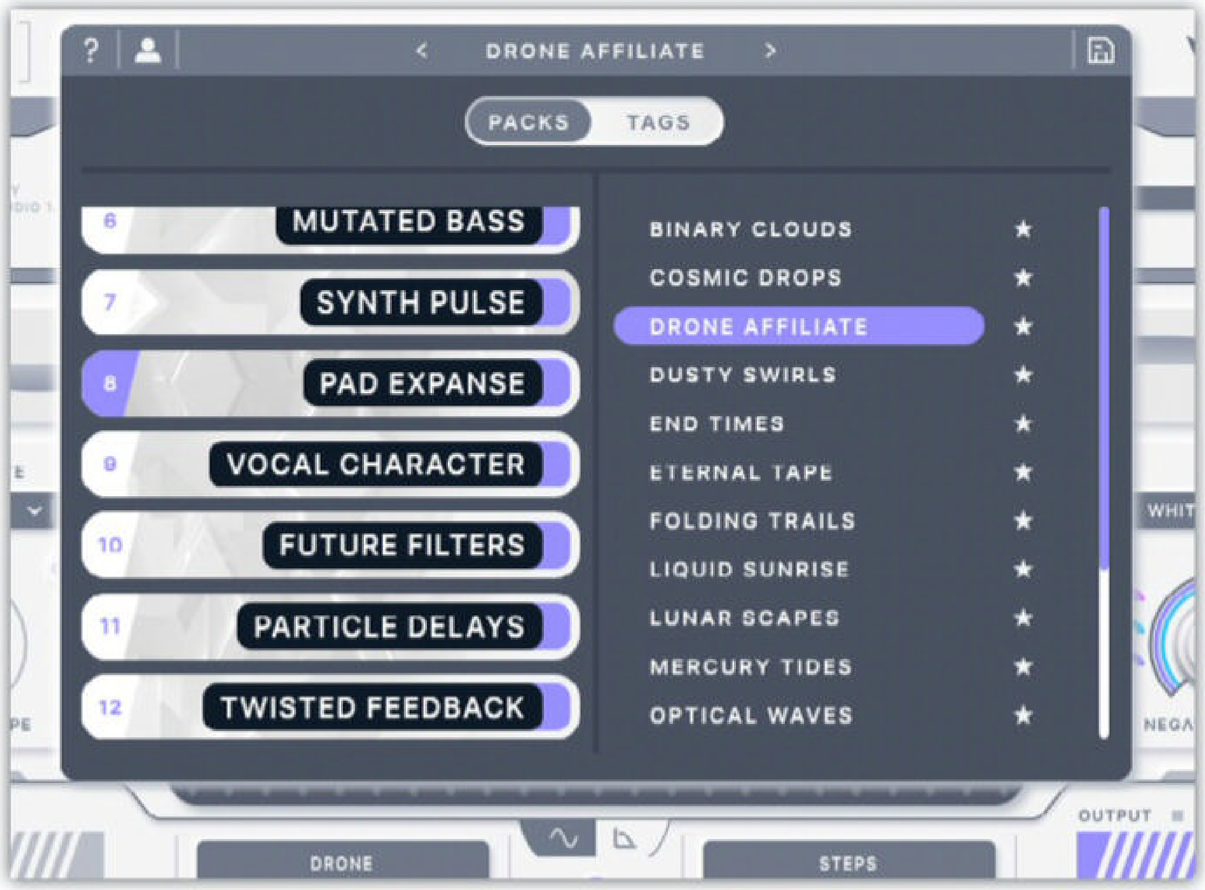
- Powerful processing options that can totally transform sounds — much more than just another distortion.
- GUI options mean it will suit different types of user.
- Flexible modulation system.

CONS

- Advanced View has a steeper learning curve.
- Not light on CPU use.

SUMMARY

Rift can transform the dullest of sounds into the star of the sonic show, and its clever dual-format GUI makes the powerful processing accessible to novice and experienced sound designers alike.



It's not all about distortion; Rift boasts an impressive collection of presets that deliver a diverse range of creative treatments.

the centre-top of the Distortion panel, a Stereo Mode drop-down selector offers choices between (L-R) Stereo, which is the default, Wide, Mid/Side, Mid (only) and Sides (only) processing. If you select Mid or Sides, Rift's processing will only be applied to that component of the stereo signal, with the other passing through unprocessed. In conjunction with the Spread controls on the Feedback and Filter panels, these modes open up some really interesting and useful choices.

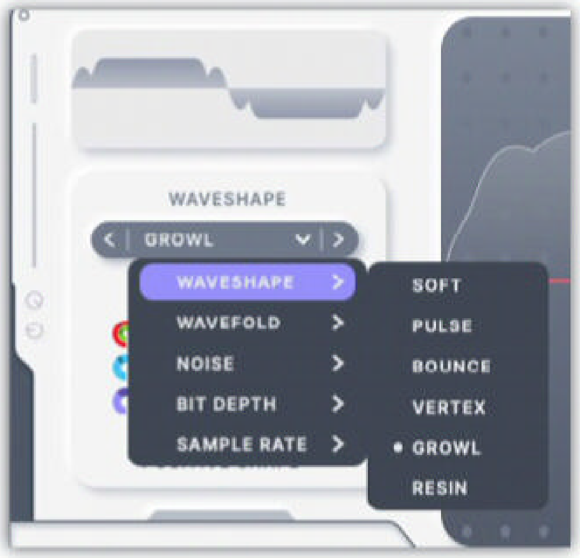
Advanced View also includes the Filter panel in which, alongside some conventional-looking filter controls, you'll find an impressive array of filter types and options to place the filter pre- or post- the distortion engine, and to blend the processed and unprocessed signals. You can also toggle the filter on/off. Cutoff can be set in Hz, to track MIDI notes, or to a specific note. In the latter two modes, the Filter panel's small keyboard icon can be popped open, and it's here that you can select which notes the Cutoff is allowed to snap to. It's a very cool idea and works well, though the manual could perhaps do a better job of explaining it. Depending on which filter type is selected, the Morph control can create real-time shifts in the filter form, while the Spread control adjusts the relative Cutoff values for the left/right channels, allowing you to further manipulate the stereo image.

Incidentally, this Filter panel is essentially what you get in Rift Filter Lite. That plug-in obviously lacks the pre-/post- option, and the internal modulation system is gone too — but you can, of course, modulate the controls using your DAW's automation system.

Bottom-left, the Feedback panel can also be toggled on/off. This panel lets you return a proportion of the output signal to the input, and can be used to create delays and chorus-style modulation effects, as well as to generate even more complex distortion effects; engaging the Distort button passes the signal back through the Distortion engine. You can set the feedback amount, switch between stereo and ping-pong modes, apply high- and low-pass filters and adjust the wet/dry balance.

The Feedback panel's Frequency settings can prompt some head scratching. In Tune mode, they can be used with the Frequency Pitch Map to create pitch-based treatments, while Time mode accesses more conventional delay-style effects. With both automation and modulation available, this powerful facility can initially seem daunting but it's worth persevering; the presets leave you in no doubt about the sonic fun to be had.

Between the Feedback and Filter panels, Advanced View provides you with access to Rift's four internal modulation sources. These are: Follow, LFO, Curve 1 and Curve 2. Follow can apply modulation based on the level of the audio input alongside the more »



Rift's distortion engine offers 30 different distortion modes, ranging from soft overdrive to complete decimation.

familiar LFO and Curve options. For each modulation source there's a very capable set of options with which you can get creative. For example, in the two Curve modulators, you get over 50 preset curves that span simple to complex, as well as the option to modify those curves and generate random ones as starting points. Once you've tweaked your modulation source parameters to taste, assigning a specific modulator to a target is a simple matter of dragging and dropping from the modulator's colour-coded double arrowhead icons onto the target parameter elsewhere in the UI. Individual parameters can receive modulation data from more than one source and the consistent colour-coding lets you keep track of what's happening. You can drag up/down on the small circular modulation icons beside a control to adjust the depth of the modulation.

These practical elements of the internal modulation process are implemented well and easy to grasp. However, how best to use them in the overall 'design' of your Rift-based processing is another matter: Advanced View is fun to explore but I have a feeling that it will take many users some time to master.

Riffing With Rift

Whether you nerd out on the bipolar approach, which is something I've not encountered before, or just think of Rift as a conventional twin distortion effect applied in parallel, the distortion possibilities and the fine degree of control are hugely impressive. But a quick dance through the beautifully organised presets demonstrates there's also far more to Rift than distortion. And while the presets ably demonstrate that Rift can deliver sonic mayhem, if you want to bring an element more subtly to your listener's attention that's also possible. Indeed, while the engines themselves sound very different, Rift can conjure up some of the same sorts of sonic magic as, say, Output's Movement.

If you're stuck with an underwhelming static synth pad or bass, then Rift is quite a tool to throw at the problem! Whether you want the sound to rage, or to give it a strong rhythmic element (for which the Chop & Stutter preset category provides a great starting point), Rift is more than capable of making that happen. It did a great job on a real electric bass too, letting me dial in a perfect combination of clean(ish) bottom end and upper-mid bite. There's also an impressive collection of presets aimed at drums: those in the Drum Heat category can easily be dialled in to add an extra spice to an acoustic kit, while

Things get a little busier when you switch to Advanced View, where you have much more control.

The impressive Filter panel boasts a huge number of filter types, and it forms the basis of the separate, more affordable plug-in, Rift Filter Lite.

the more extreme settings used in many of the Beat Damage presets will inject plenty of character into almost any electronic drum track.

It's not just synths and drums though; Rift has ear-candy for almost any occasion. Many of the presets work just as well with guitars and there are some categories of presets are aimed at vocals. Of course, there are also more 'vanilla' preset types offering conventional overdrive/distortion effects, straightforward filter effects, delays, and sweep/riser effects.

The Gift Of Rift?

Rift undoubtedly ticks the boxes as a sound design tool and, in presenting it with the two-tier Play/Advanced View GUI, Minimal have pulled off a neat trick: they've made it simultaneously accessible and easy to use, and powerful and complex, and that means it should appeal to a wide range of users. If distortion and filtering needs are more conventional, there are certainly cheaper and simpler options out there, including those provided in most modern DAWs, and Rift might be overkill for those sorts of users.

As with many other high-end creative effects plug-ins, Rift's sonic fun does demand a good chunk of your CPU resource, and on less-well-endowed computers or in busier projects, this might require some management.

But whether in the context of music creation or sound design, those that are prepared to dig a little deeper will undoubtedly find plenty of sonic treasure here. If you find plug-ins such as SugarBytes' Turnado, Output's Movement, NI's Molekular or iZotope's Stutter Edit appealing, Rift's Advanced View should be right up your street. It's deep, powerful, and capable of taking even the blandest of sound sources and transforming it into something your listener simply won't be able to ignore.

The bottom line, then, is that Rift is powerful, creative, deep and full of sonic possibilities. It's well worth a try. And if you're not persuaded yet, note that Minimal Audio have a 30-day full refund policy, which reduces the financial risk for the undecided!

System Requirements

Windows 10 or Mac OS 10.9 or higher. Host supporting 64-bit AU, VST 2.4 or VST 3. Internet connection required for activation.

$ $79 when going to press (full price $129).
W www.minimal.audio

JTC Tones Cyber Driver & Daemon Driver

Cabinet Impulse Response Libraries

It takes a hell of a lot of prep and expertise to create the perfect cabinet IR library...

BOB THOMAS

I don't normally get excited about cabinet impulse responses in the way I might the latest piece of audio hardware, but when a highly-respected, UK-based guitar tuition website launches two IR series, with a combined total of 930 impulse responses, it's going to get my attention. The website in question is Jam Track Central (JTC), whose lessons boast content from an international roster of technically gifted guitarists, and these two distinctly different IR series, Daemon Driver and Cyber Driver, were created, developed and curated by guitarist and music technologist Colin Cartmell using his proprietary Modern Vintage Hybrid Technology (MVHT) approach.

The Daemon Drivers are grouped into 17 genre-specific IR packs, each with two sets of 10 single-cabinet IRs (Cab 1 and Cab 2), plus a third Mix set, consisting of an additional 10 'mix-ready' IRs, EQ'd to be dropped into a mix. The Cyber Drivers comprise five IR collections, named after mythical creatures and grouped by sonic character rather than genre. Each collection is split into three IR packs, each containing two sets of 14 IRs for both Cab 1 and its Phase Cab, of which more later. All captures are presented as 24-bit WAV files at 44.1, 48 and 96 kHz.

The cabinet, amp and mic chain used to capture individual IRs is common across both series, and forms the basis of the way in which each cabinet is catalogued within all 32 packs. For the Daemon Drivers, captures were made using five different mics and both tube and solid-state amplification, to create 10 IRs per cabinet. In the case of the Cyber Driver series, the same amplification options were used with seven different microphones, to produce its 14 IRs per cabinet.

Ignoring the multiple sample rates, a quick totting up gives a series total of 510 individual IRs for the Daemon Drivers and 420 for the Cyber Drivers, so a total of 930. To assist potential purchasers in navigating this vast panorama of tone, JTC have created a separate webpage for each pack, with a preview video of a guitarist running through a selection of that pack's cabs with different mics, plus a more detailed description and a selection of recorded tone examples.

When I first loaded up a selection of Daemon Driver and Cyber Driver IRs, I was pleasantly surprised that I didn't need the low-pass EQ I often find I have to apply to third-party IRs, to adjust their sound to my taste. Another pleasant surprise was a realisation that these IRs had been created to deliver specific tonal

targets. Being able to choose between different tonalities — rather than having to search for the most accurate recreation of a specific speaker — felt liberating, though I haven't yet given up entirely on the searching bit! Whilst auditioning more JTC Tones IRs and exploring all their mic and power amp variations, I realised there was much more going on behind the scenes, and that I needed a better understanding of Colin Cartmell's MVHT approach. So I called him...

Meet The Maker

As soon as I started talking to Colin about the JTC Tones IRs, it became blindingly obvious that the man should be honoured as a certified tone hound. He has a very clear idea of what he wants to achieve sound-wise from guitar loudspeakers, and has the technical chops to turn his vision into reality.

I grew up with tube guitar amps and am accustomed to making modifications; I have a stash of used and NOS tubes that gets added to whenever funds allow. But Colin takes my meagre efforts to a whole other dimension: he has amassed, over the years, a huge collection of vintage paper-in-oil capacitors and other components, which he uses to tailor the response of tube power amps to his requirements. Similarly, the microphones that Colin uses to capture IRs aren't always necessarily typical examples of their type, but can be hybrids that have been created to enhance the sonic signatures of a selection of classic microphones. Created by Colin as part of the development of his MVHT approach, these hybrids were designed to extract the required tonal character from a specific combination of amp and speaker cabinets.

Microphone types common to both the Daemon and Cyber Driver series are a 57, based on the ubiquitous Shure SM57; a 57 Off, an off-axis 57; a C1, a U87-based large-diaphragm capacitor; a C2, a more modern C414-style large-diaphragm capacitor; and a RBN, a Royer R121-inspired ribbon. The Cyber Driver range utilises two additional mics: DYN 2, a large-diaphragm PL20-type dynamic mic and REAR DYN, a Sennheiser 421-style dynamic behind the cabinet.

But all this is a relatively small part of the MVHT story, the vast bulk of which must remain untold on the grounds of commercial confidentiality. But our conversation left me in no doubt that the Daemon and Cyber Drivers series »

JTC Tones Impulse Responses

PROS

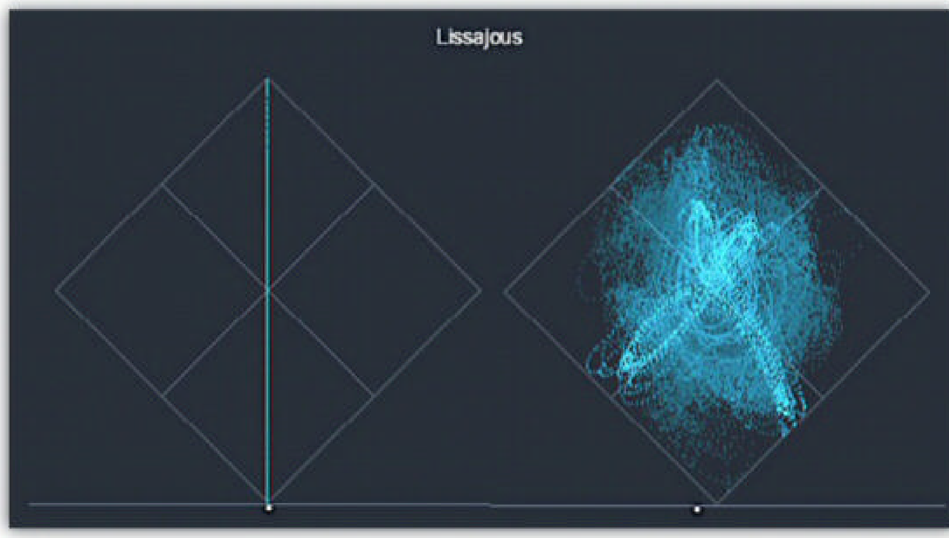
- Both series produce great results.
- Eminently usable without any editing or EQ.
- Tones to suit a wide range of genres and styles.
- Cyber Driver IRs in particular are masterpieces of their kind.

CONS

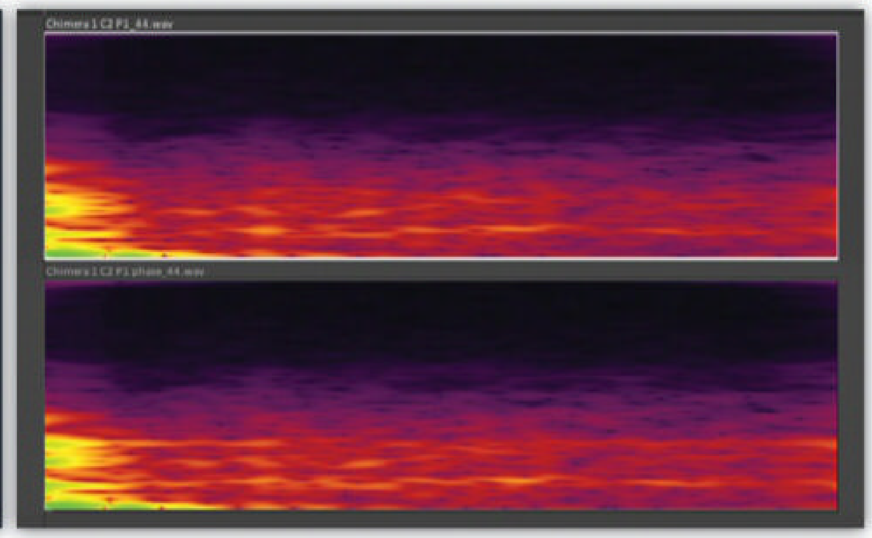
- None.

SUMMARY

These two superb collections of IRs offer a combined total of 930 tonally unique amp, cab and mic combinations. They cover a wide range of musical genres and applications and are great value for money.



The Cab 1 IR on the left has no stereo width, but mixing in the 'phase cab' widens the sound stage due to the frequency-dependent phase differences.



This spectrogram shows the subtly different responses of an IR and its corresponding 'phase cab'.

are products of Colin Cartmell's vision of how cabinets should sound and feel, and his desire to give other guitarists the opportunity to experience them.

Back At The Coalface

Alongside their tonal balance, another very noticeable attribute of both the Daemon and Cyber Drivers series is their clarity of sound. The IRs in both series are clean and dry, with no audible room reflections or reverberation, which not only points to short sample lengths, but also gives you the opportunity to determine the exact type and size of reverberant environment in which your guitar sound will sit.

To fully explore the attributes of the 17 musical genres covered by the Daemon Drivers series would take a book, and the same goes for the Cyber Drivers. But I have prepared a few personal 'tasting notes', to give you a decent idea of what's available: see the separate box. Meanwhile, to hear them in action, I recommend listening to the samples on the JTC website.

Although not every Daemon Driver IR that I auditioned managed to meet my personal tonal tastes, all of them sounded superb, felt good to play on and none of them (provided that I took care to match pickup type to genre), required any EQ to be immediately playable. The five mic styles, differing combinations of the two power amps and three cabinet types in all the Daemon packs that I auditioned produced extremely useful tonal variations that were a pleasure to use. The Daemon Driver's pre-EQ'd Mix cab IRs are not only good enough to drop straight into a mix, but they also offer plenty of options when searching for a cab sound to fit into a track.

Every single one of the IRs in the Cyber Drivers series is a masterpiece of its kind. All are dense, complex, full-sounding and superbly voiced IRs, and I know that I will be using my favourites amongst them well into the future. Whereas the Daemon Drivers are relatively simple captures of single loudspeakers, the density that I hear in the Cyber Drivers' tonality seems to point to a far more complex, extensive and exhaustive creation process. The basic sound of each of the Cyber Drivers collections may be based in the past, but Colin Cartmell's MVHT creation methodology appears to endow them with a modern clarity and definition that I found highly attractive, extremely addictive and eminently playable.

Adding in the phase cabs in order to widen the perceived soundstage only adds to the Cyber Drivers' attractiveness. Interestingly, this cab isn't 180 degrees out of phase at all frequencies, but rather is out of phase by differing amounts at different frequencies, with a seeming bias towards the frequencies in the upper reaches of the fretboard. Although you could pan the pair hard left and hard right, I found that '10 to 2' panning seemed to me to give a better overall result.

Last Words

Having talked to Colin at length about his creations, I can appreciate the study and years of work that laid the foundations of JTC Tones' Drivers series and, while the full details of his MVHT process remain secret, the high quality of all these IRs is clearly the result of Colin's very personal tonal vision. As I've said before, the man is a true tone hound and I, for one, am very glad he is. Whether you're thinking about getting into using loudspeaker IRs, or you are already there, you really should check out both the Daemon Drivers and Cyber Drivers in detail. I am sure you won't regret it.

Tasting Notes

There's a heck of a lot of IRs to choose from in these libraries, so I've tried to offer an indication of the sort of sounds you can expect to find in each pack, and for the Daemon Drivers, I've suggested which guitarists' sounds they put me in mind of.

Daemon Driver
80's rock: Eddie Van Halen, George Lynch
90's rock: Blues Saraceno, John Petrucci
Blues: Gary Moore, Stevie Ray Vaughan
Brown: Eddie Van Halen
Classic Rock: Jimmy Page, Bernie Marsden
Country: Johnny Hiland, Albert Lee
Funk: Nile Rodgers, Cory Wong
Fusion: Alan Holdsworth, Frank Gambale
Fusion Lead: Paul Gilbert, Guthrie Govan
Hard Rock: Tony Iommi, Dave Murray
Jazz: Bill Frisell, Pat Metheny
Modern Instrumental: Joe Satriani, Steve Vai
Neo Classical: George Bellas, Yngwie Malmsteen
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MODULAR

2hp Picnic Basket

Eurorack System

A troublesome 2020 presented American developer 2hp with a host of Covid-related delays when it came to getting their flagship 42hp Lunchbox systems out and into the world. At long last they managed to start shipping back in November, in a total of four fully housed and powered variations. The Synth Voice hosts 2hp's envelope, oscillator, filter and VCA modules; the Drum Machine has their kick, snare, hat and mixer modules; the Effects Rack offers 2hp's 'Grain' granular processor, delay, Freez and stereo reverb modules, while the larger and more mysteriously named Picnic Basket — which happens to be the one in my custody — contains a sequencer, Pluck physical modelling unit, a VCO, multimode filter, kick, reverb, ADSR envelope and random voltage generator. Each module is separated by a blank panel: partly to space the modules out a little (a fully stocked 2hp rack can get a little dense), but more crucially to offer space for more modules, the significance of which I'll come to later.

The Lunchbox case itself is literally an aluminium lunchbox, and even has a little handle on the top so you can take it to school on the bus. It's cute. Not altogether practical, but cute. The lid opens toward you and isn't removable, meaning that it does get in the way a little unless you stand it up vertically, where you may well find yourself pushing the whole thing across the table while patching, considering how light it is. On

its own the case retails at around £140, which feels rather pricey considering its size, not to mention the fact that it is literally a tin. It feels slightly paradoxical that while the Lunchbox series' portability is marketed as one of its main assets, it's a long way from a flight case and more likely to be treated as fragile than a trusty road-worn companion. In fact it feels, dare I say it, somewhat rickety. However: it's light, it's fun, it is portable and, as I said, it's cute. It has character. It plays well either by itself or in partnership with a larger system, so doubtless will make an ideal addition to any creative modular environment. You certainly look at it and want to plug it in — an asset not to be underrated in my experience. In this respect, the case and character of the Lunchbox are a great success. Just don't chuck it in the hold of an aeroplane.

Comparisons will doubtless be drawn between the Picnic Basket and Erica Synths' Pico System, but it's more akin to something like Erica's Black System, in that it's a collection of 2hp's flagship modules in a bespoke case and in a curated, albeit customisable, order. I've never felt that 2hp, aside from their name, try to peddle their micro format as any kind of novelty or 'compact' versions of other modules. Their algorithms are robust and sonically punch well above their (literal) weight, their workflow is inventive and their interfaces are efficient. They also pull no punches when it comes to pushing the envelope with ideas, as demonstrated by things like their recently announced time-domain pitch-shifter module. In creating a range of modules that rely a little more on mults, stacking cables and most of all on one another, 2hp are a company who simply make modular a little more... modular.

One frustration with the Picnic Basket's module selection is that while it promises to be a self-sufficient system (or in the company website's words, 'the complete 2hp experience'), this doesn't entirely seem to be the case. At the very least it still relies on an external mixer module to cope with its possible eight separate audio outputs, and the noticeable absence of a VCA means the filter becomes more or less the only dynamic control on offer, somewhat limiting its scope for other more creative applications. Here the reason behind leaving space for further modules seems to become a little clearer. It's easy to imagine unboxing the Picnic Basket and after a matter of minutes hopping straight back onto the 2hp website to order a mixer and a VCA, so

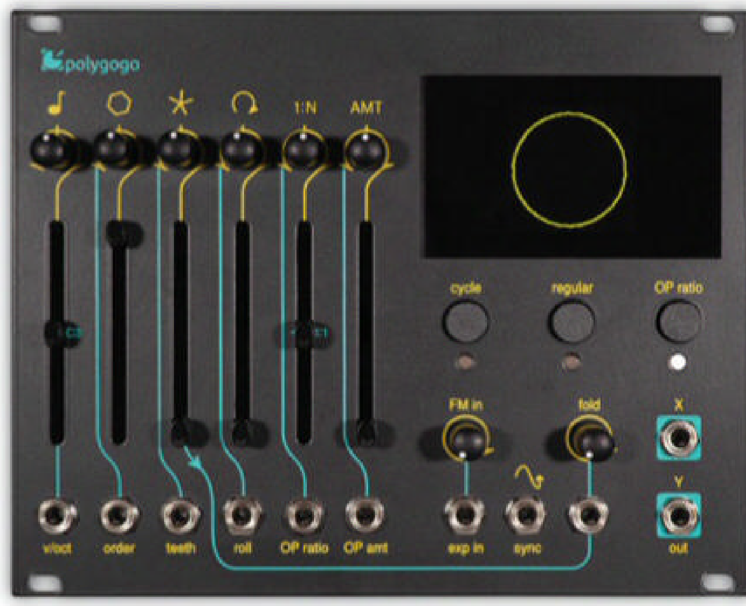
obvious is the need for them. It's equally easy to imagine feeling ever so slightly cheated at this prospect, as the Picnic Basket is not exactly cheap. But, in fairness, it still works out as better value for money than buying its modules and case separately, and since 2hp's modules are by no means the only units available in their size, the Lunchbox system paves the way for a powerful custom mini-system of slimline modules from all kinds of developers. Just remember that if you see a 500-series API preamp and get excited, you're looking at the wrong kind of lunchbox.

For all my reservations with the Picnic Basket's case and module selection, the modules themselves are quite simply excellent. The filter is deep and smooth, the Curtis CEM3340 IC-based VCO is rich and full, and the reverb is lush. Highly impressive too is the Karplus-Strong driven Pluck physical modelling module, which capably sits among category leaders like Mutable Instruments' Rings. The sequencer represents possibly the most inventive and impressive use of the 2hp real estate, managing to pack a full 16-step sequencer with selectable scale, glide and length into its minuscule footprint. The system ships with a set of patch leads and a passive mult, something of a statement of intent from 2hp: to get the most from this you'll need to be inventive and resourceful with your patching, a challenge I'd imagine to be gratefully accepted by many modular synthesists.

All things considered, the Picnic Basket does more or less exactly what it says on the tin (apologies for that). It looks good, it sounds great and its compact format opens up a huge range of possible applications; from field recording expeditions to porta-studios in unusual locations. For newcomers it's a great entry point into the world of modular, and for experienced synthesists it's a worthy addition to the pallet. Sandwich, anybody? *William Stokes*

$ $999

W www.twohp.com



E-RM Polygogo

Eurorack Module

Interesting and unique synthesis techniques seem to be plentiful of late, and the E-RM Polygogo is no exception. It's also one of a growing number of modules that are stereo, plus it may just be the best looking module I've ever seen.

The Polygogo uses a digital synthesis method invented by E-RM called Polygonal Synthesis. This works by plotting the X and Y position of drawn polygons. If, for example, you draw a circle, and you plot the amplitude of one axis over time, you get a sine wave. The speed of the drawing dictates the frequency and by changing the shape and rotation, a huge variety of stereo waveforms can be generated.

A beautiful OLED yellow-on-black screen shows you the drawing plot and updates in real time as you change parameters. The graphics are the exact polygonal plot that is output from the X and Y (left and right) outputs. It's hard to convey in words just how nice this screen looks, and the polygon plotting is rather mesmeric.

Most of the slider controls are focussed on manipulating the polygon shapes and every parameter can be CV controlled. The knobs above each slider change from fine-tuning controls to bipolar CV attenuators when a cable is plugged into the CV input. Order sets the number of corners the polygon will have, from two (a straight line) to 28 (a near perfect circle). You can switch 'regular' mode off to allow for a non-integer number of corners, which will create extra harmonics when the plotting scans a 'broken' edge. With the Cycle button, you can also toggle whether the polygon is always drawn from the same spot, or whether it begins drawing from the last position. This has the effect of 'spinning' the polygon and creating additional stereo movement.

Teeth will take the sides of the polygon and tilt them inwards creating disruptions and overtones in the waveforms. Roll rotates the polygon which causes the waveforms and relative phase relationship between the X and Y outputs to constantly shift, creating spacial movement. The final two sliders change the ratio and FM amount of an internal operator (there is an additional exponential FM input too, with attenuator). The frequency of the internal operator can be smooth, or quantised depending on the status of the 'OP ratio' button. Frequency modulation is represented nicely on the display.

Fold will push the waveform out of the edges of the screen and into the opposite side, adding harmonics at low levels and extreme crushing and distortion effects at higher levels.

The Polygogo is not difficult to master. You quickly become accustomed to connecting the visual to the audible. Smooth circular shapes make sine waves. Jagged shapes add more harmonics. Spinning shapes will have more stereo movement. And if the screen looks chaotic, the sound will be too.

As well as the wide range of stereo synth-tones that the Polygogo can generate, I found a much pleasurable distraction using it as a modulation source. The X-Y outputs make an excellent modulation pair. Sort of like having your own automated joystick controller continually drawing complex repetitive patterns. Although I don't have the hardware to test it out, I imagine the Polygogo would make an excellent controller for lasers or oscilloscope art.

I'm not sure the Polygogo really generates anything that a flexible oscillator can't already. The stereo aspect is definitely a big part, but the raw tones sound much like sawtooths and sine-waves being sync'ed, FM'd, bit-crushed, wave-folded, etc. It does however offer a very unique approach to making these sounds and taking a different path to a familiar destination can offer as much joy as going somewhere new. In the end, the Polygogo is well built, innovative and fun to use and I can't think of much higher praise than that. *Rory Dow*

$ $599

W www.e-rm.de

E-RM Polygogo
32HP
+12V 110mA
-12V 15mA

Clavia Nord Piano 5 Stage Piano

GORDON REID

The celebrated Clavia stage piano continues to evolve.

Despite its name, the Piano 5 is the sixth Nord Piano, coming after the Piano 4 and the more recent Nord Grand. I must admit that I love the Grand, not because it's the most fully featured stage piano I've reviewed (it isn't) but because my fingers love its Kawai keybed, and I couldn't stop myself from playing the damn thing while I had it here. Nevertheless, the Piano 5 reverts to the overall design and the Fatar keybed of the Piano 4, although it now offers two widths — 73 note and 88 note (the smaller of which will appeal to gigging musicians for whom size and weight is a significant consideration) — and much inside has changed.

First Impressions

At first sight, the Piano 5 doesn't look much different from the Piano 4 but, when you inspect it more closely, you'll find that its control panel has been remodelled, and not just to accommodate some important new features. Nevertheless, it took me no time at all to become comfortable with it, and I'm sure that even seasoned players of earlier models will be whizzing around it within minutes.

Since the release of the Piano 2, Nord Pianos have been based upon two of Clavia's underlying synthesis engines, the Piano and the Sample Synth, combining them with a selection of Clavia's customary effects. The Piano is based on the Nord Piano sample library and organised into six Types, each of which can contain a selection of Models. The Piano 5 is delivered with a selection of Models pre-installed but you can replace these by downloading new ones from Clavia's website and swapping them in and out using the Nord Sound Manager. There are four Model sizes, ranging from Small to XL, and you're free to choose from these as you please to make the best use of the available memory.

Depending upon the Models chosen, the acoustic pianos offer string resonance and pedal noise, with three velocity curves, an optional 'softer' release, and three timbres — soft, mid and bright — at least one of which (soft) is obtained not by applying filters but by adjusting which samples are played at any given velocity. The electro-mechanical pianos add Dyno 1 and Dyno 2 settings to create the more aggressive, cutting sounds of the preamps that inspired them, and the same button also provides access to a selection of different Clavinet voicing combinations which, when applied to the pickup combinations available as Models, provide a wide (but not all-encompassing) range of Clavinet sounds. In addition to these, the Piano provides access to harpsichords, digital pianos and a small selection of non-keyboard sounds.

The Sample Synth is based upon Clavia's ever-growing library of sampled instruments, which now includes some single-triggered monophonic lead and bass synths. Again, the Piano 5 is delivered with a factory selection and you can swap others in and out up to the limit of the memory. But beware... the sample format has been updated in the Piano 5 to take account of various new synthesis facilities. This means that the previous library is no longer compatible.

In addition to playing the sampled

instruments 'as is' you have a limited but valuable range of control over them, much of which is new to the Piano 5. There's a new bright/soft filter setting, you can now apply three velocity settings to make sounds dynamic, and you can also now apply three degrees of unison with increasing detune and spread. In addition, a new vibrato generator offers rate, depth and three delay times. This means that, when these are combined with the existing Attack and Decay/Sustain/Release controls (which provide AD, AS and ASR contours) the Sample Synth now offers simple filtering, shaping, modulation, dynamics and unison, so there's a lot more that you can do with it than you might think. The only thing that has been lost from previous revisions is the ability to direct velocity to the filter cutoff frequency; velocity will only affect brightness if the underlying sampler layers provide this, and then only if the amplitude dynamics button is activated. While the additions are welcome, losing the velocity-sensitive filter is a significant step backward.

The latest version of the Nord Sample Editor was also released while I was writing this review so I thought that it would be an opportune moment to revisit it. I set up a simple registration on a nearby organ and recorded a single Wav file comprising every semitone across one manual, playing each note for about two seconds and allowing a second or two of silence between. I then dropped the file into the GUI of the editor, whereupon every note within it was detected, its pitch was determined, and it was allocated to the correct key on the scale. I then selected all of the keys, hit the Loop button, and pressed 'Generate & Transfer To Nord'. In moments, I was able to play my new instrument on the Piano 5. I found just a handful of notes with obvious loops, so I tweaked these in the editor, regenerated the file and downloaded it again. The whole operation had taken just seconds!

I wondered how the editor would handle sounds with fewer samples per octave, so I found a folder than contained a set of RMI Electrapiano samples that I had created years ago by playing a diminished scale rather than every semitone. I selected all 20-something files simultaneously and dragged them into the Sample Editor. Again, the pitch of every sample was identified and this time allocated to an appropriate range of keys. Again, I looped the samples and, again, I transferred the results to the Piano 5. And again, the result was fabulous. Next, I tried a more complex sound to see whether this might fool the software. I took a dual-oscillator-per-voice polybrass patch and followed the same procedure. Manual looping was required for best results but, once I had done so, »

Nord Piano Monitors

The Piano 5 is compatible with the Nord Piano Monitors, which can be mounted via brackets onto the keyboard itself (as they can to the Piano 4, Nord Grand and the heavier Stage 3s). The speakers comprise 4.5-inch woofers and three-quarter-inch tweeters, so they're not going to move much air or offer you the depth or experience of a grand piano, but they could be useful for a self-contained rehearsal setup.

» the results again sounded superb. Finally, I used the Sample Editor to load a handful of discrete samples into the Piano 5, so that I could trigger them from individual notes. The only thing that I missed in all of this was 'alternate' looping, so I hope that Clavia will one day add this.

So now we come to perhaps the biggest upgrade in the new model. Previous Nord Pianos from the Piano 2 through to the Grand have each provided one Piano and one Sample Synth, allowing you to play either, layer the two, or play them either side of one of seven pre-defined split points. In contrast, the Piano 5 offers two Pianos and two Sample Synths. You can't distribute these freely across the keyboard, but you can decide whether each lies across the whole keyboard or above or below the split point. You can also transpose any of the layers up or down in octaves, which means that you can program sounds to lie in meaningful ranges in both regions. In addition, you can change the nature of the split point from an instant transition to a crossfade of 12 or 24 semitones width. Unfortunately, transposition (as opposed to octave shifting) is on a per-program basis rather than a per-layer basis, so you can't set up classic sounds such as I/IV/V solo patches or polyphonic sevenths. However, the Piano section allows you to choose one of three degrees of detune between the two piano layers so you can create some amazing honky-tonks. I accept that that may not be everyone's target in life, but the results can be great fun.

At the end of the signal path, the Piano 5 seems to offer the same effects as its predecessors, but it doesn't. To start with, two of the effects sections have been renamed (a trivial change), and their order has been changed (which may or may not be trivial because nowhere does Clavia reveal whether this represents a change in the underlying order in which the effects are applied). Either way, it seems to me that the effects in the Nord Piano have evolved from one wrong order to another wrong order. Dynamics after modulation effects? Really? The effects themselves are largely unchanged although the delay unit now boasts a ping-pong mode while the reverb now offers a dark mode, models of two additional reverberant spaces, and a Chorale mode that modulates the reverberated signal.

Of course, the ways in which you can use the effects have also been updated to take account of the two new layers. You can apply the Mod 1 effects to Piano A or Piano B or both, or to Synth A or Synth B or both, but not to a mixture of Piano and Synth layers. The same is true of the Mod 2 effects and the amp/compressor effects. This means that you can (for example) apply wah and compression to your Clavinets while at the same time applying phasing to your strings, but you can't apply wah, compression and phasing to both. In contrast, the EQ and Delay have a global mode that allows you to apply them to all four layers but not to just one of the Pianos plus one of the Synths or to any combinations of three layers. Finally, the reverb, as it has always been, remains global. If you think that this all sounds a bit arcane, you're right. But don't worry, the effects routing is far easier to use than it is to explain.

Tying all of these elements together, a Piano 5 Program contains all of the Piano, Sample Synth and Effects settings

The 73-note version offers the same features and keyboard action as the 88, just with fewer keys.

Clavia Nord Piano 5

From $3299

PROS

- It's a stylish, solid and manageable instrument that will grace any stage or studio.
- Four layers are much better than two.
- Its sample memory has doubled when compared with the Piano 4.
- The sample libraries on which it's based continue to grow.
- The Sample Synth is more flexible than before.
- The Sample Editor has evolved into a superbly usable and effective tool.
- It's simple to use and satisfying to play.
- The Triple Pedal is now included as standard.
- It can sound superb.

CONS

- It lacks balanced analogue outputs and digital audio I/O.
- It lacks separate outputs for the Piano and Sample Synth sections.
- It has no performance controllers and does not respond to pitch-bend or modulation over MIDI.
- The Sample Synth has lost its velocity-sensitive filter.
- Although its sample memory has been doubled, it's still tiny by modern standards.
- It responds to a global MIDI channel rather than a channel per layer.
- It has reverted to the Fatar keyboard of the Piano 4.
- It's not cheap.

SUMMARY

The Piano 5 remains an excellent solution for performers who require a simple source of excellent acoustic and electromechanical pianos, Clavinets, and other piano-like keyboard sounds. In addition, its Sample Synth is now more flexible and has evolved into an excellent player for one's own sampled instruments. Whether Nords continue to justify their premium prices is something that you'll have to decide for yourself but, given their visibility on the stages of the world, it's clear that many players feel that they do.

Meet The Family

Clavia sells five families of keyboards based upon four types of sound generator: digital synthesis of various types, sample-based pianos, the Sample Synth, and organs. The simplest is the Nord A1 virtual analogue synthesizer. More complex is the Nord Wave 2, which combines virtual analogue, sample-based, FM and wavetable synthesis. Next come the Nord Pianos, which combine the two sample-based engines, the Piano and Sample Synth. Moving up a level we reach the Nord Electros, which add Clavia's organ emulations to the piano and sample synth engines. Finally, we come to the Nord Stages, which combine four engines — virtual analogue and sample-based synthesis, plus the piano and organ engines — in a single instrument. Unfortunately, the dedicated Nord C-series organs are no longer manufactured and the Nord Lead name has been retired. Given the esteem in which these instruments were recently held, let's hope that they reappear sometime in the future.

that comprise your sound. In addition to the five that are immediately accessible via the 'Live' Program memories, there are 16 banks of these, each containing 25 Programs arranged as five pages of five. You can swap these around using the Organise function on the Piano 5 itself, but it's much quicker and easier using the Sound Manager software.

Second Impressions

Before I got stuck into this review, I had updated the Piano 5 from the factory-installed firmware to the latest version (v1.16), which promised a handful of optimisations, minor improvements, and a bug fix or two. I also backed it up and then downloaded the Bösendorfer Grand Imperial XL piano and installed this. In the past, that might have been the limit of my ambition but I was intrigued to discover what might be possible using the dual piano layers. So I confined the rich, rolling Bösendorfer to the low registers and loaded a livelier piano — in this case, a Steinway — into the upper registers to create a hybrid that combined the best of both worlds. It sounded great and, when I crossfaded the two, there were no obvious artefacts in the transition zone. I also swapped out a few of the factory Models in the Sample Synth, because one can never have too many Mellotrons, Chamberlins or string synths at one's disposal. Having dual layers meant that I was now free to emulate my dearly departed Mk1 Mellotron, using my own composite brass samples below the split point and the unmistakeable violins above it.

Although I occasionally found myself pining for the Nord Grand's Kawai keybed, playing the Piano 5 was still a pleasure, and the seamless transitions between Programs was a boon when compared with keyboards that cut one sound abruptly when you select the next. So everything was (nigh-on) perfect, then? Well... no, it never is. I've been saying for years that Nord Pianos deserve dual, balanced stereo outputs — one pair for the Piano and a second for the Sample Synth. On previous versions you could at least direct the pianos to the left output and the synths to the right, but even this has now been discarded. It's time that Nord Pianos grew a full set of analogue outputs as well as digital audio inputs and outputs.

The lack of pitch-bend and modulation wheels, a joystick or some other method of performance control is another limitation that has always existed on the Piano series. This was irrelevant at first because the original Piano 88 was what the name implies — a piano. But the lack became apparent when the Sample »

The updated Nord Sample Editor features improved sample import and pitch-detection function.

» Synth was added and now, with dual Sample Synths and increased synthesis capabilities, it's reached the point where it matters. Sure, there are nearly 70 MIDI CCs controlling almost all aspects of the sound generation and effects, but the fact that the Piano 5 has no physical controllers and still doesn't respond to pitch-bend or modulation messages over MIDI seems crazy to me.

Another shortcoming is the lack of independent MIDI channels for the four layers in a Program. Again, this wasn't really an issue before, but the Piano 5 would be far more flexible — indeed, four-part multitimbral — if they could be accessed independently.

Third Impressions

The Piano 5 is solid, robust and designed to a level of quality rather than to meet a price. But despite remaining instantly recognisable as the successor to the Piano 4, its sound generation has taken a significant leap forward. It's not cheap, but Nords tend to hold their secondhand values very well and, if I were the owner of an earlier model, I would think seriously about upgrading. Having said that, I might also be tempted to wait to see whether Clavia released a Nord Grand 2 with the same capabilities. Sure, it would be less appropriate for stage use but, for me, its Kawai keyboard would overshadow any minor inconveniences. Either way, I wouldn't buy any Nord that wasn't compatible with Sample Editor 4; once in a while one encounters something a bit special, and this is one of those times.

But having extolled the virtues of the Nord Pianos and their software tools, it seems to me that they're no longer as far ahead of the competition as they once were. What the Piano 5 does, it does extremely well, but the likes of Yamaha, Roland, Korg and Kawai have released some excellent stage keyboards in recent years and, while some would argue that none of these are as immediate or as stylish as the Nords, one can't deny that there's a breadth of options today that didn't exist a few years ago. Furthermore, the dividing line between dedicated instruments and fully featured workstations is narrowing, with the latter generating piano sounds that are leagues ahead of those we once expected to hear (and hoped to avoid) in previous generations. At some point, Clavia's designers will have to decide whether future Nord Pianos remain primarily pianos or continue to evolve into something more powerful but potentially less focussed. But for the moment, the Piano 5 still looks, feels and sounds like a Nord Piano, retaining the simplicity and immediacy of previous versions despite its extra capabilities, and very nice it is too.

$ Nord Piano 5 73 $3299,
Nord Piano 5 88 $3499.

W www.nordkeyboards.com

Big Sounds In Small Memories

While the Piano engine's polyphony has remained the same at 120 voices, its sample memory has doubled from 1GB to 2GB (as it did on the Grand), which means that there's room to install more of the XL Models that sound so good. Similarly, while the Sample Synth polyphony appears to be about the same (it's now quoted as 40+ rather than 46 voices) its sample memory has increased from 512MB to 1GB. Having acknowledged the improvements, I have to point out that these remain tiny numbers in an era of massively sampled pianos and 1TB memory sticks that cost little more than a large pizza and a couple of beers. But does one require gigabytes to emulate grand pianos? This question was brought home to me during the course of this review when, for quite unrelated reasons, I had occasion to liberate my Kurzweil K250 from storage. Is it as realistic as the Nord? No, of course not, although it's still admired by many players. But here's the point... the revered Kurzweil Grand Piano is stored in around a quarter of the 2MB of ROMs that contain the underlying samples for all of the K250's sounds, which is around 1/4000th of the memory available in the Nord's piano section! But far from suggesting that the Piano 5 doesn't need the 3GB it currently offers, I'm going to argue that an even larger memory would be a significant improvement because it would make it possible to load all of Clavia's Piano and Sample Synth libraries simultaneously. There must be a technical reason why we're still talking about single-digit gigabytes rather than single-digit terabytes when discussing the Piano 5, but don't ask me what it is. I don't know.

The Rear Panel

The Piano 5's rear panel is sparse, offering just left and right audio outputs and a headphones socket that echoes the main outputs. Alongside these, there's a 3.5mm audio input that's routed to the L/R outputs as well as to the headphones so you can use it for backing tracks and similar purposes.

Analogue control is provided by quarter-inch inputs for sustain and volume/control pedals, the first of which accepts the Nord Triple pedal. This offers half pedalling, Una Corda and sostenuto and, I'm pleased to report, is now supplied with the Piano. (This wasn't the case with early versions.) The second input can be used to control the volumes of the Piano and Synth sections, the amount of the tremolo and pan effects in the Mod 1 section, plus the wah's filter frequency and the rate of the ring modulator. Digital control is provided by 5-pin MIDI In and Out sockets, and you can use USB for MIDI, software updates, instrument downloads and backups.

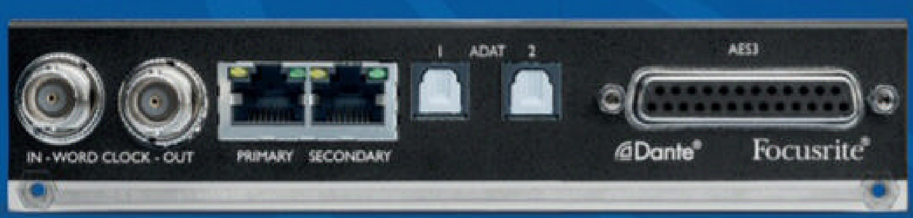
The final socket is an IEC power connector. While I'm delighted that the Piano 5 has an internal power supply, I was bemused that the review unit accepted (nominally) 230V mains only. In these days of universal power supplies, this seems an odd deficiency.

ISA828 MkII

Dante SPOKEN HERE

ISA ADN2

ISA ADN8



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ISA ADN8 is an optional A-D card for the ISA 428 MkII and ISA 828 MkII. It provides ADAT Optical, AES3 and Dante connectivity at rates up to 24-bit/192kHz. The card offers primary and secondary Dante output ports, along with an AES59-configured DB25 connector for AES3 output, and a pair of ADAT optical outputs.

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Alto Professional Bluetooth Total & Bluetooth Ultimate

Bluetooth-to-analogue Converters

There are plenty of cheap Bluetooth adapters on the market but most are aimed at the consumer sector, whereas these two models from Alto focus on the pro audio market. Accordingly, as well as offering a decent sound quality, both have male XLR balanced outputs: the Bluetooth Total is a mono device with a single output protruding from the left-hand side, whereas the Bluetooth Ultimate is a mono/stereo device with two outputs, with the additional connector recessed into the opposite side of the casing. Naturally, the latter costs a touch more but while the former is mono, note that two units (you can mix and match models) can be linked and sync'ed for stereo use. A benefit of this over simply using one Bluetooth Ultimate is that you don't need cables running between the speakers, which makes it viable to use these devices in locations where the speakers are placed far apart, or trailing cables might be a trip hazard.

The devices come in disposable plastic blister packaging, which I don't like. As well as the ecological considerations, I always seem to cut my fingers opening them — but I can also see people wishing there were some form of reusable protective case for storage and carriage when not in use.

On the top panel the two units are almost identical, the key difference being that the Ultimate adds a two-position slide switch to select mono or stereo operation. You press and hold a Bluetooth button for three seconds and the device becomes discoverable for pairing (for example with a phone or computer). It emits a recurring beep which can be helpful feedback when setting up, but can also be annoying if you have the speaker/amp levels cranked at the time!

The Stereo Link facility works like the power line Ethernet adapters I've used: you press and hold a button on one unit then do the same on the second, and they'll then pair automatically. The first device is the 'primary' unit, which to you and me means the left channel, and if an Ultimate model is used in this role it will output the left signal on both XLRs, just as it does in mono mode.

The final 'control' is an on-off slide switch. Power comes from an internal rechargeable battery that's capable of six hours' operation, and is charged using a USB cable which plugs into a mini USB socket on the side panel. The supplied cable is very short, and if you want it permanently plugged in you'll need a longer one. You'll need your own mains-to-USB adapter; you can charge it from a device's USB bus but if you want this thing to live permanently on the back of a speaker that's probably not the best approach. Different LED colour combinations helpfully tell you the present state of charging and battery power levels.

That's pretty much it. When charged and paired (there was no passcode or anything like that to contend with) it works as expected. There's no volume control; the paired playback device and your speakers/amp dictate the output level in the room. The conversion sounds as good as I'd hope for from any Bluetooth device, and with a maximum output of +4dBu there's plenty of level going into the speakers for normal playback of commercial material.

The Bluetooth range seems decent. The spec suggests that this is just over 30m, and while the range will inevitably be compromised to some extent where walls or other physical barriers are involved, it passed my go-outside and wander-around-the-garden tests with flying colours.

As with any Bluetooth system, there is some latency involved. Thus, it's not brilliant for playing virtual instruments or, say, through guitar amp sims that you have installed on your phone. For regular music playback that's not a problem, although, as with any Bluetooth system, you'll encounter sync issues if watching YouTube videos and the like. That's something I overcame easily enough using a free browser extension (YouTube Audio/Video Sync for Chrome).

Some of the settings will be difficult to see in the dark (eg. behind a speaker in a venue; while the LED indicators are great, the position of the on/off switch and the Ultimate's mono/stereo one are harder to see. But once set up, you barely need to look anyway.

On the whole, both devices perform well. They sound fine, they're easy to use, they're affordable, and I have no real complaints. So I'm happy to recommend them. For the small amount extra, I think I'd opt for a pair of Bluetooth Totals, though no doubt some will prefer the simplicity of the Ultimate (only having to link a single device) if feeding a stereo amp. *Matt Houghton*

$ Bluetooth Total $59.
Bluetooth Ultimate $99.
W www.altoprofessional.com

JMG Audio Expanse 3D

Spatial Audio Plug-in

Adding to the growing portfolio of unitedplugins.com is Expanse 3D, JMG Sound's spatial enhancement plug-in for Mac and Windows DAWs (VST, AU and AAX), which the designers tell us employs psychoacoustic principles to "make your sounds deeper, wider and bigger". Authorisation is via a personal key code that can be used on multiple machines, and sample rates up to and beyond 192kHz are supported.

The resizeable user GUI is about as simple as it could possibly be, with three main 'amount' controls linked to three sides of a cube graphic, joined only by In and Out level controls. Each section offers a small menu of processing modes. We're informed that the processing includes a combination of analogue-style saturation, spectral phase offsetting, re-synthesis, intricate delay networks and other techniques, but a key point is that all the processing is mono compatible. The plug-in can also generate a pseudo-stereo output from a mono input.

Width is adjusted using a spectral phase processor, and this section simultaneously reduces the stereo width of very low frequencies to keep the low end sounding tight. An auto gain switch allows the user to audition settings without the loudness changing. The Width modes are Normal or Alternate, both of which are similar but have their offsets at different frequency points. Each has a normal and inverted version, and to separate instruments you can choose different modes, so that they will not share the same frequency offsets.

The impression of front-to-back depth is created using delay networks, transient manipulation and phase processing, to »

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add a subtle ambience that helps glue sounds together by placing them in the same space. The three modes essentially offer different delay times.

The Height section enhances the highs and lows using established 'exciter' principles to generate new harmonics. In subjective terms, this produces a brighter, more airy sound while also adding harmonics to the low end, to create the impression of a more powerful bass sound when heard over smaller speakers. Where filtering is applied, it is designed not to introduce phase shifts, and the saturation algorithms include anti-aliasing measures and up to 8x oversampling to keep them sounding natural. Ext Lo helps fatten up audio that has little bass, while for audio with little treble the Ext Hi mode may be appropriate. Ext Lo+Hi suits sounds with little in the way of highs or lows, though it's worth noting that the Normal mode is generally the best of the bunch for using on full mixes.

I tend to be wary of 'make it sound better' boxes, but Expanse 3D has proved a useful tool for making tracks or mixes sound a little bit wider and a touch larger than life — just as promised. As with all such tools, it has to be used sensibly, but deployed tastefully it can help either increase separation between instruments or glue a group of sounds into a more cohesive whole, while expanding the stereo width to a useful degree. It won't turn a bad mix into a good one, but it's quite capable of adding an additional polish to mixes that already sound good. If you're still not convinced, there's a 15-day free trial, which should allow you plenty of time to decide. *Paul White*

$ $120.
W https://unitedplugins.com/Expanse3D

United Plugins Urban Puncher Saturation Plug-in For Mac & Windows

The name of this plug-in gives you a hefty clue as to what it's about, but as with most of its apparently simple United Plugins stablemates, there's more going on under the hood than it might at first appear.

Urban Puncher's primary role is in beefing up drum and percussion sounds, though it can also work well on bass parts or to add a sheen of analogue warmth to other instruments. Supporting all the usual plug-in formats, and with a generous authorisation system that allows you to run it on multiple machines, Urban Puncher was designed by United Plugins boffin Boris Carloff, and combines saturation based on transformer emulation, with dynamic processing that seems to combine elements of compression, transient shaping and spectral manipulation.

The presence of a Wet/Dry control indicates that Urban Puncher is quite happy working as a parallel processor as well as a serial one. Other than a variable output level control, the whole thing is controlled by just two knobs and one button, with a further slide switch acting as a bypass. When no signal is passing, the plug-in relinquishes its hold on CPU resources, and operating bypass causes no clicks or changes in latency. Helpfully, the GUI can be freely resized.

Punch is where the dynamic processing happens and, to my ears, turning it clockwise introduces what sounds like a blend of progressively heavier compression/limiting and transient attack enhancement. At its maximum setting, the sound is quite aggressive and 'smacky' but that can be just what you need when mixing in some of the dry signal. Dialling in a little saturation can beef up the sound even more, but if you're after a more overt effect, then pressing the Destroy button essentially triples the amount of saturation, making for a sound that adds crunchy to punchy. Again, this can work for you in lots of different ways, from treating a dull snare track to making the whole drum kit sound pleasingly trashy.

As suggested earlier, the 'beefing up' capability of Urban Puncher often lends itself well to processing bass synths or bass guitar too, but it is well worth experimenting to see what it can bring to other instruments — just try it where you might normally opt for compression and see what you get. Instruments with well-defined attack sounds, such as guitars, are definitely good candidates for treatment, but it can be useful just about anywhere that you need to add a bit of analogue-style warmth, and where you want to lift up those lower-level details that add interest and attitude.

There's enough control range to go from super subtle to quite hard hitting, so Urban Puncher turns out to be a very flexible tool with more applications than its name might suggest. I also love the immediacy that its simple controls offer — the learning curve is so flat you could climb it in carpet slippers! *Paul White*

$ Introductory price of €19 (about $23) when going to press. Full price €59 (about $69).
W https://unitedplugins.com


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SAMPLE SLICING
Beatmaking With Hardware

SIMON SHERBOURNE

Chopped-up loops are at the heart of many genres of music. There's never been a better time to get hands-on with sliced samples.

Sample chopping remains a core compositional tool in modern music production, especially hip-hop and the various flavours of breakbeat music that have evolved from jungle. By chopping I specifically mean sampling an extended musical or rhythmic phrase and slicing it into sections, then re-sequencing and reworking the parts to make something new. In hip-hop this technique is routinely used to generate the key hook or seed for a beat. Drum & bass typically uses acoustic drum breaks, cut up (and sped up) and reprogrammed into a new drum track. But any genre of music can benefit from the idea, whether it's used to generate ideas and interesting loops or layers, or to extract a kit of individual hits.

Two Turntables & A Microphone

The Musique Concrète movement was the first to use pre-recorded sounds as compositional sources, and John Cage even experimented with a pair of turntables as far back as the 1930s. But the technique of repurposing existing recordings as we recognise it today comes down from New York's hip-hop pioneers and Jamaican dub DJs, who looped and scratched drum breaks manually on turntables while MCs rapped over the top. As hip-hop developed, sampling became more than just a means to an end, but part of the sound and identity of the form.

Affordable sampler hardware like the Akai S series made more complex production possible, letting you grab and cut up raw material even without turntable skills or tape editing. Akai and Roger Linn developed the MPC60, adding a built-in performance interface to the sampler, and making it easy to juggle multiple samples. Computers brought sequencers and trackers, and later Propellerhead's ReCycle accelerated the edit process for samplers, making sample slicing simple. Chopping samples and using loops became a common part of electronic music and pop production across the board.

DAWs and even simple audio editors now offer almost unlimited scope for assembling music from recorded audio, essentially providing an efficient and non-destructive upgrade to tape workflows. But triggering samples dynamically remains a faster and more performance-oriented approach than assembling clips on a timeline. And so performance sampling workstations, which all share a lineage with the MPC60, are more popular than ever.

In this article I'm going to focus on sample-chopping workflows for the 'big three': MPC, Maschine and Push. There are of course many other devices which excel at chopping and playing samples, including the Elektron Octatrack, Roland MC-707, Akai's Force and now the Novation Circuit Rhythm. Then there are plug-ins like FXpansion Geist or Reason. Many other software and hardware samplers can let you get to the same place. But there's a specific convergence of features and techniques found on MPC, Maschine and Push.

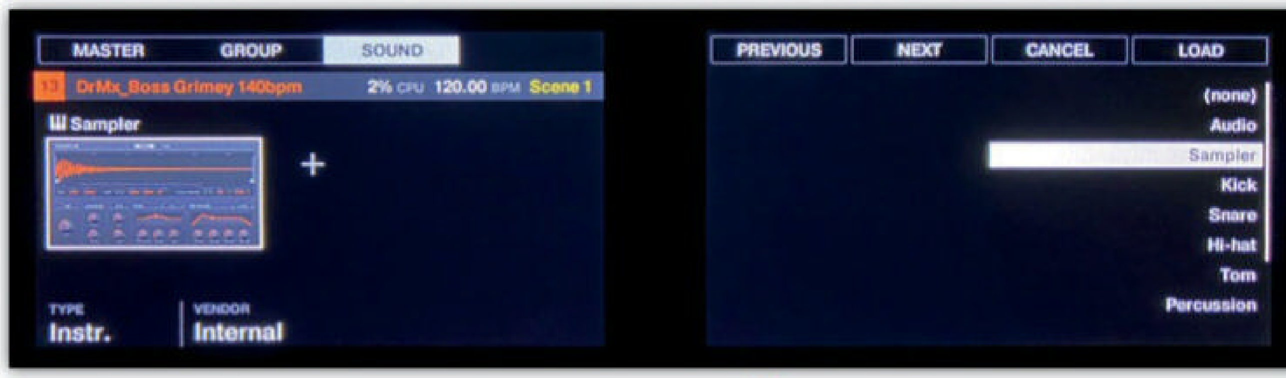
Outline

The process of slicing and playing a sample is similar across the three workstations. The steps may be slightly different for each, or occur in a different order, but here's what we need to do:

1. Load in a sample, either by browsing files or recording directly.
2. Chop it up using one of several options, depending on the material and how you want to use it.
3. Tempo-match with your project if necessary.
4. Convert the sliced sample into a playable kit or program.
5. Play and sequence the sections to create something new.

There's a ton of possibilities and creative tricks to be explored, but we'll start with the basic steps to getting started with the three workstations, then try to draw out the common themes.

There are three ways samples are going to get on your chopping board: direct recording from an external source, importing from a file, or internal re-rendering (bouncing). For this section we'll look at how you get started with each workstation individually.



If you import a loop in Maschine, make sure you switch the pad from the Audio to the Sampler plug-in to enable Slicing.

Sampling In Push

Direct sampling on Push means recording into an Audio Track in Live — there's no separate sampling mode. (You might want to keep a dedicated track in your Session for sampling.) Simply record from your chosen external or internal source into a Clip. With the Clip selected on Push, press the Convert button and choose the first option, which is Simpler. A new track will be created with your sample loaded into the Simpler device, ready for live triggering and chopping. You can do the same thing with any Clip in the Arranger timeline, but you'll need to select it from the computer instead of Push.

To load a sample file instead of recording, you can Browse with a MIDI track selected and your sample will load straight into a Simpler: no Convert needed. Alternatively, you can load samples into Clip slots on an audio track and Convert from there. Push's Browser can access anything in the Live core library, Packs or User Files areas. If you select Current Project you can also load in any sample that's already used in the Session, so this is another way to grab clips that are in the Arranger. Once again, when your sample is loaded, press Convert to load it into a Simpler.

Sampling In MPC

MPC has a dedicated Sampler mode for capturing audio. Choose from the available inputs, or the Resample option for an internal bounce. Sampling can be armed and triggered manually from the onscreen record button, or can be set to trigger at an input threshold. In the standard Sample mode, after recording you are given a choice to name the sample and choose where it will go. Choose <none> in the Program selector, so that the recording will be placed into the Project's general sample pool.

The Slice mode in MPC's Sampler view lets you grab slices to pads live as you record.

The MPC is unique in letting you chop a sample while it's being recorded. For this, use the Slice sampling mode. Tapping the drum pads as you sample will now create slice markers in the sample on the fly. After recording you're given the option to save to the pool, or create a new program with your sample slices assigned across the pads. You can trim the slices later.

If you're loading a sample from your library or a thumb drive, use the Browser. There's no need at this point to use the Sample Assign function, as we'll be creating a new Program after we've chopped.

Sampling In Maschine

Like the MPC, Maschine has a Sampling mode for capturing and re-rendering, with manual or threshold-triggered recording. There are three record modes: Detect, Sync and Loop. Manual recording is available by choosing Detect and setting the Threshold all the way to the Off position at the far left. What you record will appear in a Sampler instrument on the selected pad and Group. Subsequent recordings will appear in a pool for that pad on the left-hand screen, unless you select another pad first. The Sync recording mode is useful for re-rendering, or grabbing a loop from a sync'ed device. You can set a number of bars and recording will start and stop in time with playback. We'll leave the Loop mode for another time.

It's easy to get tripped up if you import a file from the Browser instead of recording, because Maschine will load it into an Audio device on the selected pad. This is designed for playing back loops and linear recordings rather than triggering samples, and does not support slicing. After import you'll need to press

Unless cutting to individual hits, you'll need to tempo-match samples. On Maschine, process before chopping; MPC and Push can warp.

» Shift-Browser to go to the channel plug-in selector. Switch the Audio device to a Sampler and the Slice option will become available when you're in the Sampling view.

Editing

If you've captured something from vinyl or YouTube or wherever, you might want to 'top and tail' it to remove excess material around the part you're interested in. It's not strictly necessary, but cleaning up now may save some time later, and tempo-based operations are simpler when you have clean in and out points on the beat.

Push makes life easy at this point. We already got as far as converting the sample into a Simpler patch, so the overall Start and End times will be available on screen already, and that's it. Sample editing on Maschine happens in the Sampling page. If you've just captured something you'll already be there, so you can switch from Record to Edit mode and trim up the sample. Unfortunately this is then ignored if you switch to Slice mode, so you'll want to Truncate the sample first from the process menu on the right-hand screen.

To start editing on MPC, go to the Sample Editor and use the data wheel to select your new sample, then adjust the Start and End times. When you switch to Chop mode, this Start and End time will be respected unless you're in Manual Chop mode, in which case it might be worth using the Extract process to create a truncated sample.

Time-stretching

I promise we'll actually chop this sample up eventually, but we need to make a quick pitstop on behalf of the Maschine crew. If you need to match the tempo of the sample to your project, this has to happen before you chop on Maschine. Maschine's Sampler instrument can't warp audio in real time (even though the Audio loop player can), but you can do an offline stretch in the sampler editor. Tab through the processing options above the second screen until you get to Stretch. Set the sample's current tempo (Maschine will try to auto-detect). The new tempo will default to the project's tempo. You can change Pitch independently.

On Push and MPC, you can switch pitch-independent tempo-matching (called Warping on both platforms) on or off after you've chopped. On MPC you can set this per-slice; to apply to all, set the Slice select field to All.

If you'd prefer to adjust the tempo by a traditional speed/tuning adjustment you can do this too. On MPC's sample editor, make sure you're in the default Trim view and tap the From BPM button next to Tuning. In the dialogue that pops up, make sure the correct number of bars/beats are selected so that the correct original tempo is detected. Tap on Match and the sample will be tuned to match the tempo of your current Sequence. On Push, simply turn Warp on and switch the mode to Repitch.

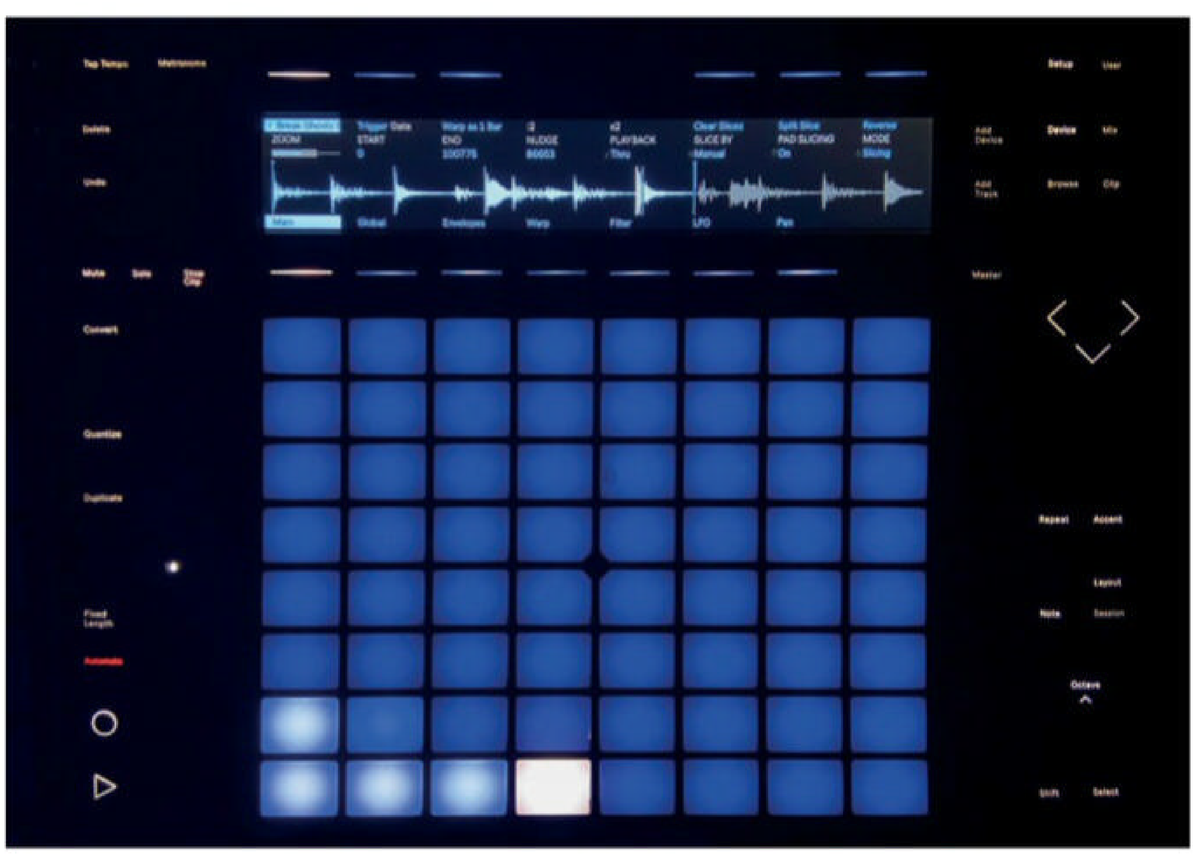
Cut Here

Finally it's time to chop! The Slice/Chop tools on our workstations offer several ways of working. Manual mode is for dropping slice points on the fly while listening to the sample. Simply hit the first pad and the sample will play back, then every time you hit the next free pad it will make a split point and assign it to that pad. Manual mode is fast, and great if you already have a rough idea how you want to divide up the sample, or want to divide up unevenly. You don't have to get it exactly right: you can just tap a pad roughly where you want a cut, then go in and nudge it afterwards.

You can immediately start playing the pads you've assigned at any point. If you let playback continue past the last slice you can then resume adding more chops from the next free pad onwards. MPC and Live have an extra trick not available on Maschine: you can hit the next free pad at any time during playback, and if you're in a section that already has chops, a new one will be inserted, causing all the slices to ripple to the right on the pads. On Maschine you'd need to select a pad, choose Split, then manually edit.

Transient Detection analyses the waveform and drops slice points at clear hit points. MPC calls this Threshold mode, Maschine calls it Detect and Push calls it Transient. All three workstations have a setting for adjusting the sensitivity of the detection. Transient Detection mode is good for cutting up drums into individual hits.

Regions mode (called Split on Maschine) simply cuts the sample up into an arbitrary number of sections. This is particularly powerful on Push, where you can set the



Manually chopping a break into phrases on Push. Real-time warp tempo matches to the Session.

Transient Detection is useful for cutting a sample into individual hits.

overall Start and End time on the same screen, within which the grid will be constrained. This makes it easy to focus on and cut up a section in a longer recording that's not cut to bars. MPC can do similar but with some view flipping.

Beats mode (BPM on MPC, Grid on Maschine) is similar to Regions but uses musical timing values, assuming that your sample is an evenly cut loop. This one is enhanced on Maschine by the option to set a BPM value. None of the workstations support triplet divisions though... use Regions mode, 6/8 fans.

Chopping Notes By Workstation

Maschine

Maschine has three pages of controls under the left screen when chopping, accessed with the left/right arrow buttons. Page 1 sets the mode and its options. Page 2 provides edit controls for trimming the start and end points of slices. Page 3 contains export settings that will take effect when you hit Apply. You can set whether the finished kit Group will play in mono or poly mode, and whether a replay pattern is created. A replay pattern is a MIDI sequence that plays your slices back to reconstruct your original loop.

When you're done chopping and editing, hit Apply. The next step determines where and how your chopped sample patch will be created. The pads for the current Group will be displayed, and the current pad will flash. If you hit OK, your slices will be key-zone mapped across MIDI notes from C-2, within a single Sampler instrument and channel. To play them you'll need to switch to Keyboard mode and shift the octave range down. In most cases I don't do this (although see the arpeggiator tip later), because I want my slices laid out like a standard Maschine kit across the Group, with each section on a different channel. To do this, after you hit Apply, tap the Group's master select button (or a different Group if you prefer); then, when you hit OK, it will be split out for you. The Group/Kit format is especially powerful on Maschine, as it gives you full pitch-sequencing control over each sound as well as multi-lane gate sequencing.

Push

The main idiosyncrasy to note for Push is that instead of editing Start and End points of slices, you have a single Nudge control for adjusting the start. So if a slice glitches at the end because it catches a bit of the next transient, you need to select the next slice and move the Nudge value a bit earlier.

On Push, your chops are performed in the ready-to-go Simpler instrument, so there's no need for an Export step. However, as with Maschine, you might prefer to work inside a multichannel kit instead of a single-channel instrument to get the benefit of individual controls per sound, and multi-lane step sequencing. This is achieved by pressing the Convert button and choosing Drum Rack. The new Drum Rack will be created based on the settings you had in the Simpler. For example, if you were in Mono playback mode, the Drum Rack channels will all be choked.

MPC

The essential thing to know on the MPC is that you need to hold the Shift key to get to some extra commands, such as Delete Slice and Convert. Convert is the Export operation that will turn your chopped sample into a playable instrument, in this case a Drum Program. When prompted choose New Drum Program, with Non-Destructive Slices. You'll then need to select this program in a track to play it.

More Settings

Velocity

When you're dropping and editing your chops, playback will be at full velocity on MPC and Maschine. Once you've exported as kits/instruments they will be fully velocity-sensitive, which is good for individual drum hits, and not so good for longer samples and breaks. Both workstations will let you edit the programs to change this... just remember to select all pads first. On Push, Simpler defaults to a slight velocity-to-amp mapping (35 »

Crate Digging

Although this article focuses on the production technique, half the skill and creativity is in finding and choosing samples to start with. There's the romantic image of the crate digger, hunting through old records for sublime and undiscovered gems ripe for repurposing. I've gone for a more sedentary approach of late, grabbing random weird stuff from YouTube (staying mindful of usage rights of course). But you don't have to find something unique: most hip-hop and breakbeat producers have probably taken a swing at the 'Amen Brother' break at least once. I'd highly recommend the YouTube channel Eightminutesupside which produces compilation videos showing the samples used by a particular artist, or where an artist themselves has been sampled. It's fascinating to discover how much some of my favourite records are built from samples, and how different artists use the same sample.

There's also no reason to be snobbish about using library samples. Our three workstations all come with access to tons of loops and expansion packs. The beauty of chopping a loop instead of just using it straight is that you're putting your own twist on it that's unlikely to overlap with someone else's. If you still feel squeamish about using a sample library, why not make your own? I have a collection of loops, short 'needle drops' and sound design textures I've been using and reusing for years. Most were created and captured with a friend in just a few sessions, and cut in ReCycle. We've all got countless ideas and loops languishing on our hard drives and drum machines, so why not bounce them out as samples to use as inspiration in other tracks?

Taking A Break

The temptation when cutting up a drum loop is to let your system's transient detection slice the sample into individual drum hits. This is fine if you just want to take the character and vibe of the original sounds and turn them into a drum kit. It's also the classic ReCycle/REX file method for turning a loop into a sequence that will follow your project's tempo without time-stretching. But often there's more creative fun to be had if you make fewer chops, leaving some pads playing sections of multiple hits. This borrows more of the groove of the original sample, and lets it breathe.
You'll know exactly what I mean if you've ever tried to program a drum and bass pattern from individual hit samples. It nearly always sounds disjointed and lacks that smoothly rolling cadence. Instead, pull a drum break into a high-tempo project and speed it up to match. (As well as getting the classic pitched-up sound, the tempo match is essential when individual slices contain more than one hit.) Now chop it into regular slices — four per bar is a good place to start. With the click track playing, start by tapping your pads sequentially on the beat to recreate the original loop. Now experiment with playing back different patterns. You're guaranteed authentic-sounding drum & bass and hours of fun! The same idea works across other genres too. The screen here shows a one-bar acoustic drum break from the MPC factory library that I cut up into just three slices. There's a couple of kicks in the first slice and a side-stick on the second, with a ride cymbal on the eighth notes through both, then a kick and open hat stop on the last slice. This time I stuck at the original tempo of 92bpm, and when I played with it, a whole new groove with many variations and fills fell out of it naturally.

A nice Push/Live tip to mention here is the Thru playback mode in Simpler, which is also a feature in Reason's new Mimic sampler. In this mode, when you trigger a slice it doesn't stop at the end of that region, but continues to play the rest of the whole sample. This makes it even easier to generate natural-sounding breakbeats and drum performances from a drum loop, with less coordination required to avoid gaps in playback.

Even just three slices can be enough to create a completely new beat from a sample.

percent) which will also carry through to a Drum Rack if you convert. As with all global settings, you definitely want to do this before you Convert because there's no way to edit multiple pads simultaneously on Live's Drum Rack!

Mono/Poly

All the samplers default to Mono playback when slicing, which is normally the way to go when re-sequencing something you've sampled. However, there are times you'll want to layer the slices, especially with single drum hits. Again, this can be set after you've created your patch, but it's quicker to do it from the outset.

Envelopes

Chops default to One Shot/Trigger mode, so each slice plays back in full when you hit its pad. Sometimes you might want to swap to a gated play mode with amp envelope; individual hits are a prime candidate for this.

Getting Creative

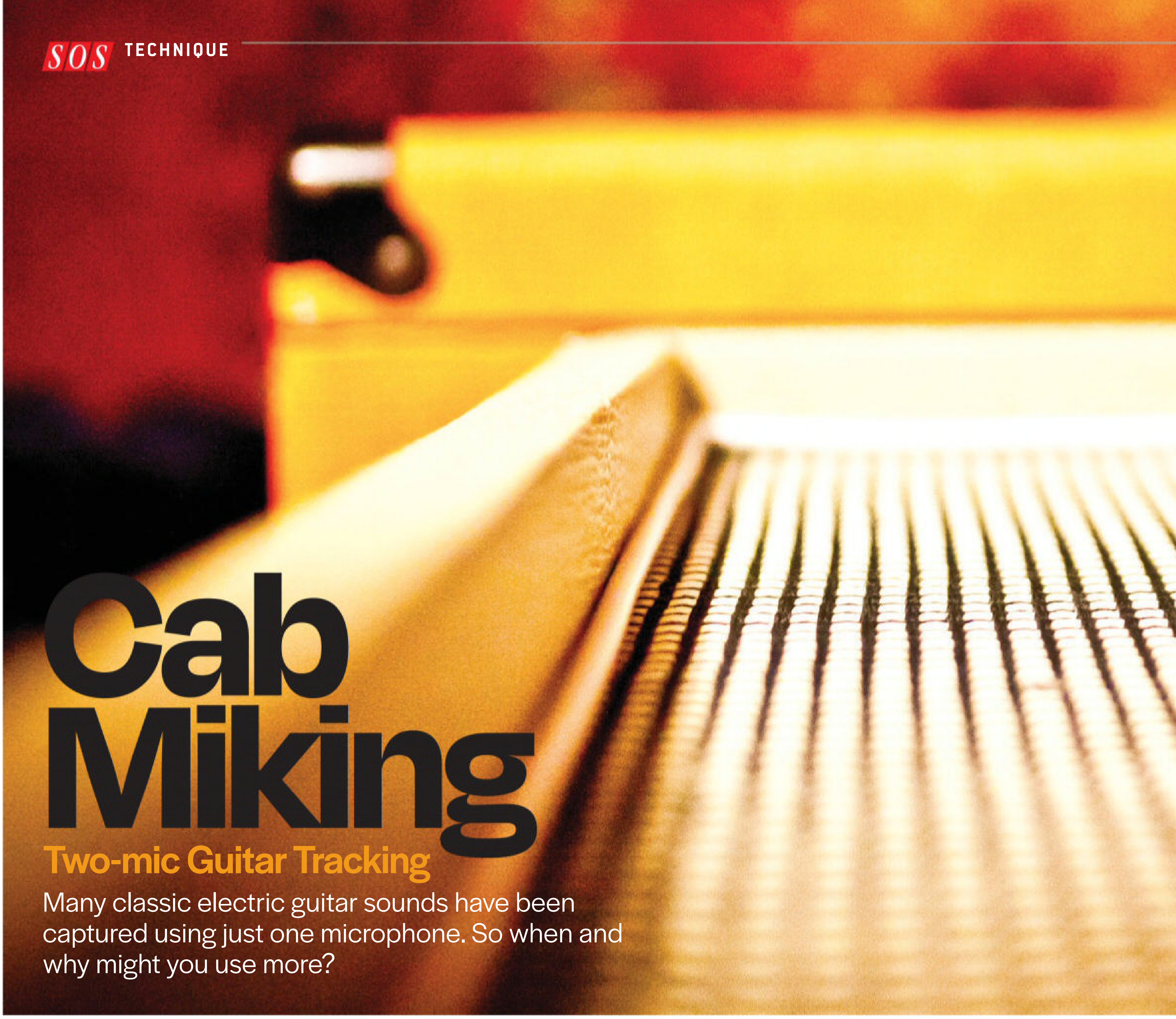
With all the technical processes out of the way it's time to have some fun and experiment. The beauty of this method is that you can jam and try things out in real time, and record what you're doing as MIDI patterns. You can also try different ways to generate sequences for your slices. You can create patterns through step sequencing, or try the randomisation or probability features if your workstation has them. Try using the arpeggiator in Maschine or Push to play back slices in changing sequences on a single-channel instrument.

When you're playing your slices you'll probably notice that you're only using a handful of what's there. Don't forget that it's easy to delete, duplicate and move around pads on all the workstations. I like to remove pads I'm definitely not going to use, and organise a bit. Often there are slices that are similar to each other in a repeating musical or rhythmic loop. Alternate between the different slight variations of the same slice to get a more natural feel. If you're playing a slower, more open groove than the original sample it might sound 'gappy'. Try chopping a pad of 'dead air' to fill the spaces, or give the slices some shape with an envelope and add a reverb to the group.

You can also treat each slice differently with effects, modulation or pitching. Duplicate your snare and tune it up a couple of times for that classic drum & bass ascending pitch fill. If you have vocal or string samples, try switching to poly mode and layering things up, duplicating and pitching as required. If you're sequencing sliced hits from a drum loop, change the original feel with groove quantise.

Whatever kind of music you make, sample chopping can be an inspiring and freeing way to spark new ideas, borrow a feel or mood, or add life and interest to a track. Your taste in samples can become part of your sonic signature, show off your genre knowledge or sense of humour, or give a nod of respect to a favourite artist. And bouncing and chopping your own loops means everything you record when you're tinkering in the studio can be stocking up your own personal crate for later digging.

From the Chop screen on MPC, hold Shift to access the Convert button to create a program from your slices.



Cab Miking

Two-mic Guitar Tracking

Many classic electric guitar sounds have been captured using just one microphone. So when and why might you use more?

PAUL WHITE

My article in *SOS* July 2021 (www.soundonsound.com/techniques/how-record-guitar-cabs-one-mic) focused on the basics of recording electric guitars using a single mic. In many cases that can deliver perfectly decent results, and I'd always recommend that inexperienced recordists start there. But it's long been common in professional circles to blend more than one signal when tracking electric guitar. There are various established multi-mic techniques to try and probably plenty more awaiting discovery, so in this article, I'll explain when and why two mics might be better than one.

Take Two

The most obvious reason to use two is when, despite having tried all the mics at your disposal, you can't quite get the sound you want using only one. Perhaps the best mic choice and position deliver the tonal 'bite' you're looking for, but lack a little punch in the low end. When you add a second mic, the relative position of the two mics can make a big difference to the sound, but it's best to start by choosing mics for their tonal characteristics and placing them right next to each other. That said, if you have the time to experiment, it's worth trying all of your mics in different pairings, as it's not always immediately obvious why two mics happen to work well when combined.

A popular, well-established tactic is to pair a moving-coil dynamic mic, such as the ubiquitous Shure SM57, with a ribbon mic. The Royer R121 is a popular choice in the latter role but they all have broadly the right characteristics: typically, the full bottom end and smooth, rolled-off high frequencies of a ribbon mic combine very well with the more mid- and presence-focused SM57, without either needing much EQ. You can balance the two to taste and either record that blend to a single track, or capture them separately on two tracks. The latter approach can leave you with greater tonal control when mixing, without needing EQ — though I'm of the opinion that having lots of options when mixing isn't always a good thing!

Combining a dynamic mic with a capacitor model is also common practice. The Neumann U87 and various AKG C414s, usually in cardioid mode, seem particularly popular but, while the results seem to vary more from one capacitor mic to another than with ribbons, you can try any of them — that's how I came to know that one of my personal favourites

in this role is the Audio-Technica AT4050. Don't worry too much about whether the moving-coil or capacitor mics you try have a cardioid or hypercardioid polar pattern, as it's the subjective end result that counts. But you should be aware, when it comes to spill from unwanted sources, that hypercardioid mics are more sensitive at the rear than cardioids.

Close Mics & Phase

When two close mics are placed on the same source, their phase relationship is naturally a consideration. However, because the electric guitar is inherently an 'unnatural' sound, the comb-filtering effect you hear when moving one mic a little further from the speaker isn't necessarily problematic. In fact, it can be a useful means of fine-tuning the overall tonality without resorting to EQ.

If you treat the two-mic blend as a single signal (ie. pan the two mics identically, maintain the same blend of their signals throughout the song, and apply any processing on a group bus) then phase cancellation needn't really be a concern. But if the mics aren't in phase and you choose to pan them to create a sense of width, the tonality will sound different for mono playback than for stereo — it might or might not be a problem, but you should definitely consider the impact on the mix balance. Also, if you change the balance between the two mics during a song, the nature of that tonal change can be less predictable if the two aren't broadly in phase.

A coincident array, which places the two mic capsules as close as possible, ensures that you avoid phase problems, but a simple rule of thumb is that if you make sure that both mics are the same distance from the speaker cone, phase cancellation is unlikely to be a significant problem. For more precise alignment, you can position one mic, then flip the polarity of the other and move it around while feeding a noise source through the amp. Cancellation between the two mic signals will be greatest in the most phase-coherent positions, and once you've found one, filpping the polarity back to its original setting should give you a satisfyingly full sound. With the amp in a live room, you'll need to listen on headphones for the cancellation, but if your amp is near your DAW you can feed both signals to the same level meter and watch for the lowest reading as you move the second mic.

A further degree of tonal flexibility is available by moving the second mic relative to the axis of the speaker: the tone can change quite noticeably as you move the mic from aiming at the centre of the speaker to aiming at its edge. As I explained in the previous article, one of my favourite mic positions is around 30cm from the cabinet, aiming at the cabinet's top edge, and it's well worth trying this alongside your favourite close mic sound.

Another versatile option, the Fredman technique, which has become popular in metal production, combines on- and off-axis placement. You put one SM57 close up, on-axis, aiming at the middle of the speaker, and place another right next to it, the same distance from the speaker but angled inwards, about 55 degrees off-axis. The on-axis mic delivers a very bright sound, while the off-axis one is dominated by mids and lows, so you can adjust the balance of these mics' signals to control the tone. The result tends to sound richer and thicker than a single SM57, retaining some of the highs while also smoothing them somewhat. Again, it's a good technique if you have a tendency to procrastinate: if you record the mics to separate tracks, you can put off deciding on the final tonal balance until the mix.

Rearing Up

The most obvious options involve two mics in front of the speaker, but the cabinet resonance picked up by a mic at the rear of an open-backed cabinet can often help beef up a top-heavy front-miked guitar sound. As with miking the front of the amp, you can try different distances and positions relative to the axis of the speaker, and it's worth experimenting with

A classic combination is to pair a ribbon mic with an SM57: the frequency responses of the two seem to combine particularly well.

» both the cab and mic position, as factors such as the distance from a wall or from the floor will also affect the sound.

There's no right or wrong — just go for what sounds right for the track — but it's worth a few notes of caution. First, as when using top and bottom mics on a snare drum, you'll usually want to flip the polarity of the rear mic; it's pointing in the opposite direction from the front one, so if you forget to do that the combined result will sound thin, weedy and phasey. Second, although the trouser-flapping low end you experience when standing close to a big amp can seem impressive, don't obsess too much about it when recording, as very little of it ends up on most records. If you do capture lots of low end, you'll often find yourself filtering it out when mixing to avoid clashes with the bass guitar and kick drum, and more so when the guitar is one of many instruments in an arrangement.

Room For Improvement

A different reason to add a second mic is to capture room ambience. Typically, you'll use one or maybe two close mics for your basic guitar tone, as above, and then position another much further from the speaker, usually a couple of metres or more, to capture more in the way of reflections. It's only really worth attempting this in a room that has a desirable ambience; if you're working in a bedroom, the result will usually end up sounding dull and boxy, and you'll almost certainly find that faking it with a reverb processor delivers better results. But don't write off a room straight away, as you can work to make an unflattering space sound better. For example, try improvising screens behind and to the sides of the more distant mic to dry up the room's natural ambience; rugs or duvets draped over chairs work well. Also, applying artificial reverb to a room mic can sound very different from adding it to a close mic.

The final sound will be the result of a number of complex interactions between the close and ambient mics, as well as between the direct sounds and the floor/wall reflections. You can experiment with the polar pattern of the room mic and, assuming it's a directional one, with where you aim it. For instance, you'll often get different results from pointing a cardioid mic at the amp, past the amp, and directly at the wall to capture first reflections, and aiming a figure-8 mic's side null at the amp, so it captures reflections to front and rear, but no direct sound from the amp.

The comb-filtering effect when using one close and one distant mic might be stronger than with two close mics, and this depends on various factors, including the floor material and how far the guitar cab and mics are from the floor. The higher the amp from the ground, the greater the path length of the first floor reflection and thus the greater the time difference between the direct and reflected sounds reaching the room mic. Again, any comb-filtering could be seen either as a problem or as a variable for adjusting the timbre. For

A Sound To DI For?

Capturing a DI along with a miked amp sound opens up more options, not least 'faking' a second amp and recording chain using modelling hardware or software, for the sort of stereo setup I describe towards the end of the main article. You might think that if a modelled tone is good enough to form part of your sound, it would be good enough full stop. Sometimes it is, but in my experience, I play better when I can hear the sound coming from my amp, and there's something about the more nuanced aspects of the sound that seem to come across better than when relying entirely on modelling. Nevertheless, mixing real and modelled amps can add an extra dimension to your sound.

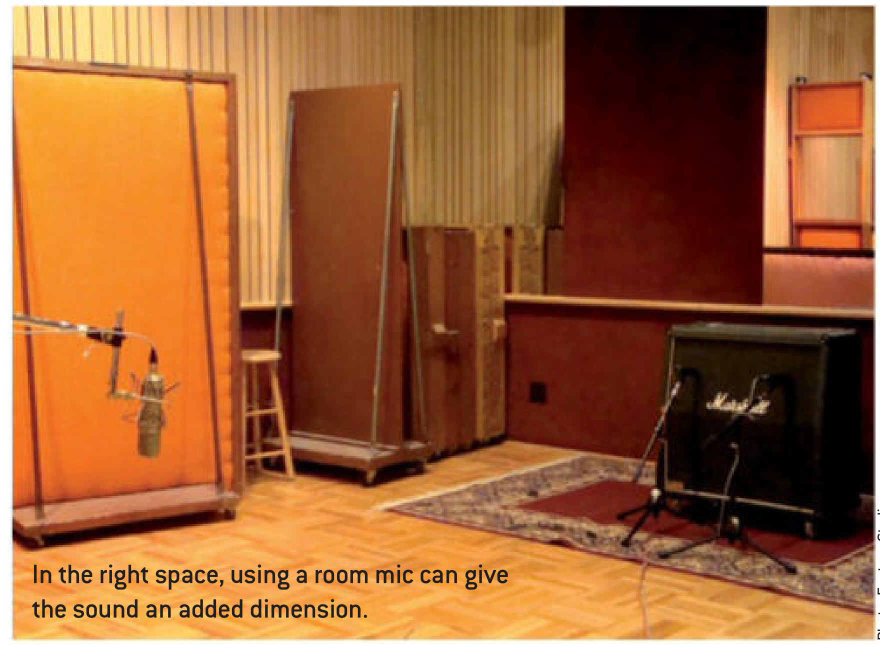
You can take your DI feed from a conventional DI box placed before the amp or, if it has one, a line output on the amp itself. A major advantage of splitting the guitar's output is that you can choose a completely different second amp type from the one you're miking. Using different amps will give a more dramatic stereo spread but also opens up opportunities for combining cleaner and dirtier sounds, to give a sense of power and sustain without losing clarity.

Should you choose to work from your amp's line output, you'll have only the cabinet and mic/room modelling to explore, but a good speaker-emulation plug-in might still give you enough tonal variety to create a worthwhile stereo image. Note that some amps have only a 'speaker emulated' output built-in, and while these rarely give you the same tonal options as dedicated hardware or plug-ins it's still worth feeding that signal through a better speaker emulation, just to see if it sounds good in combination with your miked amp. Often it does.

Most good modelling software also allows room ambience to be introduced, and this can be used to simulate a room mic for your close-miked amp. Even where it doesn't, though, you can run the DI signal through your modelling software followed by an ambience patch in a separate reverb plug-in, to emulate a good room mic.

One important thing to check, whenever working with both a mic and a DI, is that the polarity of the DI'd signal matches that of the mic. Your mono button should tell you that fairly swiftly, but so too will inspecting the waveform in your DAW, or polarity-inverting the DI track. Hardware processors will add a small amount of delay, so if you record their output on a separate track you can also adjust the negative track delay parameter to ensure that the waveforms from the mic and processor line up correctly.

The Fredman technique employs two SM57s, with capsules adjacent but at different angles relative to the speaker, giving you a lot of tonal control simply by balancing the two mics.



In the right space, using a room mic can give the sound an added dimension.

Photo: Fantasy Studios

instance, hard floors produce the strongest reflections, so they'll emphasise any comb filtering — but they will also make the timbre of the reflected sound reasonably bright, which could be useful. In short, experimentation with the mic choice, and the speaker and mic placement/orientation is important if you're to get the best sound.

Nice Spread

Another reason for using two mics is where a single mic produces a perfectly good sound but you want to create a sense of stereo spread beyond simply adding stereo ambience. One way to do this is to mic one speaker in two different ways, then pan the contribution of one mic left and the other right. This could be achieved using two mics, two different mic positions, or both.

Alternatively, you could mic up two different speakers. If working with two speakers in the same cab, note that the direction in which you move the mic when working off-axis will also affect the composite sound. Moving the mic further away from the nearest adjacent speaker will reduce the amount of spill from that speaker; moving it towards another speaker will increase it. Again, there's no inherent right or wrong: make the effort to listen for the differences and make a decision. Of course, there's nothing to say you can't pan a close mic one way and a distant mic the other, and in a nice room you could also use a stereo pair for the room, for some natural stereo spread.

Whichever option you choose, check the result in mono just to ensure there are no serious compatibility issues, since when you listen in stereo while panning sounds hard left and right, phase problems will tend to be hidden. As when using two mics for tonal reasons, if you keep the close-mic distances similar you'll avoid most of the phase issues that produce unwanted tonal changes.

If you're lucky enough to have two amps, and a big enough space in which to record them, you can split the guitar signal to feed both amps, and mic each amp's speaker using any of the techniques I've discussed above. By setting up the amps to produce different sounds, you can fashion hybrid tonalities and convincing stereo sounds without resorting to plug-in trickery. You can also create a usefully subtle stereo effect by using a chorus pedal or rotary speaker effect on only one of the amps — this can work particularly well on clean guitar sounds. Note that you might need to use a commercial splitter or a DI box with a Thru connection to avoid ground-loop hum when feeding two amps in this way and, for the same reason, should plug both amps into the same power outlet.

Still More Options?

You can experiment with using even more mics, of course, and you might find some useful results that way. For example, a three-mic setup known as the 'phase EQ' technique can provide you with oodles of tonal control at mixdown. But generally I prefer to keep things uncomplicated, since using multiple mics can easily introduce phase variations that weaken your sound rather than make it stronger. Which options work for you will depend on the tools and spaces you have at your disposal, but I urge you to try any mic and/or DI arrangement you can dream up, just to see what happens. If putting your amp in the bathroom and aiming one of the mics down the toilet gives you the result you're after, then why not? The only real rule is that it should sound good in the context of the track — curiosity costs nothing but time and the rewards can be very worthwhile.

Pairing a capacitor mic with a moving-coil dynamic is also common. Here, you can see a Neumann U67 and SM57 combined by engineer Vance Powell on a Fender Princeton amp.

MONITOR CONTROLLERS

An Introductory Guide

Why might you need a dedicated monitor controller — and what separates the good models from the bad?

HUGH ROBJOHNS

In days gone by, analogue consoles included a monitor section that, at the very least, provided a volume control (usually the largest knob on the desk) and a source selector, typically to switch between monitoring the mix output and one or two external two-track recorder playbacks. Today, most of the studio action centres around a computer, DAW software and an audio interface. Without a console, you'll either need your DAW and audio interface to perform those duties — or a monitor controller.

Can I Just Use My Interface?

In its most basic form, a monitor controller controls the listening volume of a single set of speakers. Most also include some means of switching between a few sources, and of sending the selected source to different loudspeaker and headphone systems. More elaborate units enable you to check various aspects of the monitored signal and possibly control artists' cue headphones.

The combination of DAW and audio interface is capable of performing the same tasks, and can be a very cost-effective solution: the signal path from interface to speakers is direct and simple, with nothing in between to degrade the signal; a multi-channel interface can cater for multiple sources and destinations; and your DAW can be used for various signal checks.

The fact that most interfaces adjust the monitoring volume digitally seems to concern some people: for each 6dB of digital attenuation you use one less bit of the source's word length through the D-A converter, so that must mean less 'resolution', right? No! The signal-to-noise ratio does decrease as the signal is attenuated, but the quietest elements disappear gracefully into the (dithered) noise floor, just as in the analogue world: there's absolutely no 'loss of resolution' relative to an analogue volume control.

What is important, though, is to work with a properly optimised gain structure throughout the monitoring signal path. This ensures that the interface's noise floor is kept as low as possible. Most budget interfaces' D-A converters are capable of 115dB or more dynamic range, and most nearfield monitors can produce peaks at 1m of around 112dB SPL or more. So, if the peak digital output level is aligned to match the peak SPL capability of the speakers, the interface's noise floor will lie comfortably below the

threshold of hearing: -3dB SPL, with the above figures. I covered the optimisation of the monitoring path gain structure in detail in a 2014 article, which you'll find on the *SOS* website: http://sosm.ag/reference-monitoring.

What most interfaces don't offer is any signal-conditioning functions so, in such a setup, these must typically be performed in the DAW software, and some DAWs make that easier than others. For example, Cubase Pro and Nuendo include a Control Room facility for switching between different speaker and headphone sets, setting and recalling reference levels and hosting plug-ins in the monitoring path.

That all sounds very promising, doesn't it? So why bother with a dedicated monitor controller? Actually, there are various reasons. First, although computer crashes are rarer today than in the early days of DAWs they still happen, and when they do there's a chance that the audio interface crashes too, generating full-level noises that aren't good for speakers or the listener's ears. Having a reliable volume knob or mute switch a hand-stretch away can be an 'ear-saver'.

Another consideration is that a standalone controller makes it possible to select and audition external sources when the computer is powered down. For example, you can connect a CD player, or take a minijack feed from a phone. Or perhaps you want to play a synth or through a hardware amp modeller without booting up your PC.

Furthermore, as a hardware controller is a self-contained entity, there's no need to dedicate space on your screen for monitoring functions, or to switch screen views to get at them — something which I find can become particularly tiresome if working on a laptop.

Finally, using a monitor controller means that you can employ multiple speaker sets without using up dedicated audio interface I/O, and that can mean that you have a wider range of audio interface options within your budget.

Active Or Passive?

The simplest monitor controllers are passive volume controls comprising a potentiometer or adjustable attenuator in a desktop box. However, passive controllers are not always cheap and basic. Many include alternative input selection and multiple loudspeaker output routing, and some very expensive passive designs employ elaborate switched attenuators as precision volume controls. Some even use bespoke multi-tap transformers. However, it's difficult to include complex signal-conditioning options in a passive controller.

A simple passive monitor controller can work well, but can be compromised if using long cables. Some models, like this Mackie Big Knob Passive, feature a Dim control, that allows you to monitor temporarily at lower levels without disturbing the volume control.

Many people believe that passive controllers are inherently better than active ones simply because there are no electronics in the signal path to add noise or distortion, but the reality is more nuanced. Provided the connecting cables are kept reasonably short (a couple of metres or less), they usually work perfectly well, so at the budget end of the market, they can offer good value. A potential downside, though, is that as the volume control is adjusted the source device 'sees' a varying input impedance, while the loudspeakers receive a signal from a varying source impedance. With good design and short cables these variations aren't usually problematic, but if working with vintage impedance-matched equipment or long cables, they can cause a loss of high frequencies and introduce distortion. Radio-frequency interference (RFI) is often a concern too, particularly with budget passive controllers, because many don't maintain an accurate impedance balance between the hot and cold sides of the balanced signal paths, which substantially degrades the common-mode rejection ratio (CMRR).

Active designs employ electronics to buffer the inputs and outputs, to ensure stable and properly optimised input and output impedances and good CMRRs, so long cables or unusual source and destination impedances can be accommodated. With good design, the noise and distortion introduced by the active circuitry is far below audibility, often bordering on being unmeasurable. Active designs also make it easier to incorporate additional functionality like headphone amps, talkback and metering, as well as making it easier to implement signal-check facilities. Low-level, unbalanced consumer signal sources can also be accommodated because the necessary gain is easy to provide.

Naturally, active monitor controllers cost more to make than simple passive types, and most upmarket monitor controllers are active. But not all good ones are expensive: the 'sweet spot' for home studios is probably in the $300-600 range, where you'll find technically well-specified and well-equipped products from the likes of Drawmer, Audient and SPL. The bottom line, whether a model is active or passive, is that if the volume is set to unity and you can hear a quality difference when the monitor controller is plugged in or bypassed, it's not good enough: a monitor controller's role is to allow you »

Desktop Or Rackmount?

Most basic monitor controllers are desktop devices that put all the controls within easy reach, while more elaborate models often host the electronics and most I/O in a rackmount chassis. Some have front-panel controls, but more expensive versions usually involve a desktop remote control. While the all-in-one desktop units are convenient in many ways, they inevitably entail several cables snaking across the desk and keeping this neat and your controller securely in place can be challenging. In a rackmounting system that cabling is kept well out of the way and, if using a 'producer' style desk with built-in racking, a model with front-panel controls can be convenient. Most remote controls communicate using digital command signals, so there's usually no practical restriction to the cable length. However, a few rackmount models route audio signals from the chassis to the control and back, and in those cases the use of long remote cables can potentially degrade the signal quality.

Some models, such Heritage Audio's RAM System 2000, include a Bluetooth receiver, which could prove handy given that most new smartphones lack mini-jacks!

to hear only what is presented by the source signal, so it is essential that it adds nothing and takes nothing away.

Inputs & Outputs

A basic requirement of a monitor controller is to allow comparison between a current work-in-progress and a reference: assuming you want to work with an external reference source like a CD player, that means at least two stereo inputs. Most studio equipment has balanced analogue line-level outputs, but the ability to accept signals from unbalanced consumer equipment is often important too. A mini-jack socket, for example, allows many smartphones to be hooked up. Many more advanced monitor controllers also include digital and/or USB inputs and a reference-grade D-A converter, and some models now cater for Bluetooth streaming. When comparing sources, it's usually important that they have the same perceived volume, so the better-equipped monitor controllers include facilities to adjust or 'trim' the level of selected inputs.

It's often helpful to be able to audition signals on a couple of different monitoring systems. The primary reference would typically be a set of full-range, high-quality speakers, perhaps with headphones as a secondary option. In addition, checking a mix on a set of compact and limited-bandwidth speakers gives a useful impression of what someone listening on a TV or laptop computer might hear. Where multiple speaker outputs are provided, the monitor controller usually incorporates facilities to adjust the level of the additional outputs relative to the primary set, so that the SPL in the room doesn't change significantly when switching between systems.

In installations involving a subwoofer, it can be helpful if the sub can be switched on and off from the monitoring controller (that may require bass-management facilities in the controller rather than the sub), and in multi-channel rooms it's generally necessary to be able to switch between the surround speaker set and a stereo set, to check downmixes. It's preferable to check mono compatibility on a single dedicated mono speaker too, so an output configurable for that purpose can be useful.

Drawmer are one of a handful of companies offering very capable active monitor controllers at an accessible price for the average home and project studio.

On The Level

It's important not to disturb the main volume control, because we automatically acquire an acoustic frame of reference

» when working at a consistent listening volume; if that volume changes, our frame of reference changes too, and mixing decisions made with the different monitor levels come out differently. More sophisticated monitor controllers usually allow a 'reference' listening level to be recalled at the press of a button. With less elaborate systems, we rely on using a calibration mark around the volume control — I often use a wax pencil mark. Ideally, this would be around the 1 or 2 o'clock positions, where ganged rotary pot controls offer the most precise gain adjustment and accurate tracking between channels, leaving an extra 10-12 dB of gain for checking low-level sounds.

After the volume control and source/speaker selection, I find that the most useful controls are the Dim and Mute buttons. Mute kills sound whereas Dim reduces the level, typically by 20dB, and they're useful if you need to quickly quash loud sounds (such as someone moving or unplugging a mic) or have a discussion in the control room, without shouting over the music or changing the volume setting. Importantly, a Dim button allows you to check a mix at low levels — which can be very helpful in identifying balance problems — without disturbing the volume control. But it also neatly side-steps the problem that most ganged potentiometers suffer from very poor channel tracking at low volume settings, causing the image balance to be offset and/or move around wildly with small changes in volume.

Signal-check Facilities

Most monitor controllers cover the basic signal-switching and conditioning facilities described above, but when auditioning music or other recorded signals, other tools can help you identify common problems. The most important is undoubtedly the ability to check **mono** compatibility, which requires listening to the sum of the left and right channels. This summed signal is likely to be louder than either channel individually, so most (not all) mono-summation circuits introduce an overall attenuation of 3-6 dB, to maintain a consistent listening level. This summed-mono signal is typically routed to both speakers, which is useful for checking the location and focus of the phantom image. In a correctly setup system that should manifest as a narrow, sharply-focused sound source precisely mid-way between the two speakers. If not, one speaker may be defective or set

The control panel of the author's Crookwood mastering console, which features all the signal-check facilities discussed here.

up incorrectly, or there could be problems with local reflective surfaces. Some monitor controllers also include a **balance** control to shift the image centre left or right, to compensate for the individual listener's hearing; different people have different sensitivity in each ear, and so perceive 'the middle' differently.

Listening to summed mono on two speakers is the default condition, but it creates a very different impression, particularly of the low end, from listening to mono on a single speaker. More capable monitor controllers allow the summed-mono signal to be routed to a single speaker in a stereo pair — traditionally the left one — and some can be configured to route the summed-mono signal to a separate dedicated 'mono-check' loudspeaker. If this facility isn't provided, an acceptable alternative is to send summed mono to both speakers, then mute the right speaker, which requires individual **channel mute** controls in addition to the overall mute function. Usefully, though, this facility doubles as a **channel solo**: auditioning each channel in isolation can help you identify problems on a single channel, which can be difficult to detect with both channels running.

The next most useful feature is a **polarity reverse** (often mislabelled 'phase'). It's usually applied to the right channel but occasionally each channel can be inverted individually. Flipping the polarity of one channel provides a helpful 'sanity check' when auditioning very wide or 'phasey' mixes: if the mix becomes more stable and focused with the polarity reversed, there's probably an unwanted polarity inversion in the recorded signal.

Most acoustic signals have a distinct natural polarity, typically with more positive energy (compression) than negative (rarefaction). Ideally, the monitor system should reproduce sound in the same polarity, often called absolute phase, and flipping the polarity of both channels simultaneously allows that condition to be checked. Many monitor systems will sound quite different if, for example, kick drum transients cause the woofers to move inwards instead of out!

Polarity reverse functions should be implemented before the mono sum circuitry, so that when the right-channel polarity reverse and summed-mono facilities are both employed the output is L-R instead of L+R. This is the '**stereo difference**' signal, or Sides channel. Soloing the Sides channel makes it quick and easy to align channel gains on stereo sources or stereo mic arrays, as well as to assess what's being lost in the summed-mono signal. The stereo difference signal is very revealing of the damage imposed by lossy codec formats such as MP3: lower bit-rates

What About Surround?

Multi-channel monitor controllers normally include options to work with stereo as well as various multi-channel surround speaker formats, and often include facilities to insert signal processors into the monitoring path, as some legacy formats require monitoring through the complete encode-decode signal chain. They also typically provide an option to audition a stereo down-mix in addition to summed-mono, to check compatibility between different mix formats and ensure that nothing essential is omitted for stereo or mono listeners. Ideally, the contribution of each channel into the stereo down-mix is adjustable to suit different platform requirements, but many offer only a fixed generic downmix matrix. With so many channels, individual channel mutes can usually be switched to operate as individual channel solos instead, but all of the signal conditioning and auditioning facilities mentioned here are typically available (or can be created) in the virtual monitor controller section of many DAWs, too.

affect stereo imaging and reverberation character significantly.

Other useful facilities are **left-right swap** and **mono left/mono right**: the mono left/right modes route the selected channel to both monitor speakers, while the left-right swap exchanges the two channels. The former is helpful when working with single-channel material or assessing the quality of each channel independently. The latter is useful for identifying where different elements are in the stereo image, and confirming whether a channel swap has occurred somewhere in the plugging. Taken together with the polarity inversion functions, these facilities allow individual auditioning of the Mid, Sides, Left and Right elements of a stereo programme.

Many monitor controllers are intended to serve as the central control hub of a project studio, with facilities for artist headphone monitoring (with independent source selection and talkback). Metering is sometimes included internally too, but more often, a dedicated output is provided for an external metering system. Sometimes the DAW input is made available at buffered outputs to feed external hardware recorders.

Decisions...

Different virtual or hardware monitor controllers include different subsets of the features and facilities described above. Selection of an appropriate model depends on the physical installation requirements, and your normal workflow patterns and personal preferences. I would prioritise the ability to set a reliable reference listening level above all else, followed by having dim and mute buttons easily to hand, being able to check summed mono on a single speaker, and access to the stereo difference signal — these are the functions I use all the time. If your budget is tight and most of what you want to check emanates from the DAW, you could consider a hybrid approach whereby your DAW provides sophisticated signal-check facilities, and feeds a simple monitor controller offering external inputs, a level control and mute/dim.

MIX RESCUE

Kirsty Cooke

There are times when a mix engineer needs to do more than just mixing — even if that takes you out of your comfort zone!

The free Arturia Analog Lab Lite plug-in provided the perfect piano sound for the song.

SAM INGLIS

Mission creep is a fact of life in mixing. Our job is to generate a finished track that sounds good. If that can only be achieved by editing, pitch and timing correction, replacing sounds or even adding new parts, so be it. The challenge is to know when this additional work is necessary, and to retain enough objectivity to judge whether it's really helping.

This month's Mix Rescue was a case in point. Kirsty Cooke had posted several iterations of her song 'Heydays' on the *SOS* Forum, but despite lots of helpful feedback from Forum members, the mix wasn't quite fulfilling her vision for the track. Realising that potential required a lot of conventional mixing work — but it also meant stepping out of that box.

Imagination

'Heydays' is a heartfelt ballad, with Kirsty's voice backed by piano, strings, synth bass, numerous backing vocal parts and a handful of other elements. She wanted the vocal to be quite forward, "like a solo vocalist in front of a choir in front of an orchestra", but in her own mix, the backing instruments were too low in level and very reverberant; and although the prominence of the lead vocal highlighted the excellence of the performance, it also brought out loud breaths, sibilants and harshness.

The sophistication of Kirsty's vocal arrangement was obvious on opening the beautifully organised multitrack. She had exported the backing vocals in named groups, suggesting that they were not intended to form a single choir-like mass, but distinct parts with their own musical identities. All of them had been generated using her own voice, with artificial harmony generation extending her impressive range even further. The vocals were supplied dry, but the piano recording had rather more reverb baked in than I'd have liked, so my first step was to recreate a new version from the MIDI part.

Kirsty had mentioned John Lennon's 'Imagine' as a point of reference for the piano sound and, slightly to my surprise, I discovered the perfect patch in the free Arturia Analog Lab Lite plug-in supplied with Pro Tools. This let me vary the hardness of the virtual hammers to dial in an appropriately muted tone. I then put together different impulse responses in HOFA's IQ-Reverb V2 until I had something that sounded both intimate and expansive, and turned my attention to the vocals.

Marginal Gains

Hearing the vocals in isolation made clear that the issues I was noticing in Kirsty's would need dealing with at source. Given that these parts had been recorded on an AKG D5 — an inexpensive handheld mic — the overall tone was surprisingly good, and there were no issues with background noise or spill. However, some breaths were distractingly loud, and sibilants and 't' sounds likewise stood out jarringly, especially when they were duplicated in stacked parts. As is often the case when the singer is moving the mic around, the vocal levels also varied rather unpredictably, with sudden level changes in the middle of notes, or quietly sung notes coming through louder than belted ones.

When you have more than 30 vocal tracks to contend with, it's obviously a massive time-saver if you can handle these issues using automated tools such as compressors, de-essers and so on. But in a case like this, where the issues are clearly audible even in the raw tracks, and the level variations aren't necessarily related to the content of the parts, there is only so much that these tools can do. In other words, you need to roll up your sleeves and get busy with the mouse.

There are a few ways to go here, depending on the tools available in your DAW. Since Avid introduced Clip Gain into Pro Tools, my preference has been to start a mix with the faders at zero, and use this feature to adjust the levels of any clips that are inaudible or much too loud. I took this approach further here by making the Clip Gain line visible and using it to write clip-level automation, aiming to do most of the heavy lifting at the lowest possible level. One benefit of doing it this way is that the height of the on-screen waveform reflects the adjustments you make, giving you a handy visual guide as to what sort of level to aim for. Another is that it leaves track fader automation available as a second line of defence for fine-tuning.

It will be apparent from the screen captures just how much Clip Gain automation I ended up writing. The total number of automation nodes ran into four figures, and there were places where I redid it several times. It's vital to audition the results in context, especially with backing vocals. When there's a prominent consonant or sibilance in six parts simultaneously, it's often necessary to make much more drastic moves than you might do in an exposed lead part.

I also bussed each thematic group of backing vocals to its own stereo Aux. As most of them included obviously artificial harmonies anyway, there seemed little point in aiming to have them sound natural. Instead, I decided to try to give each group its own 'signature sound'. This involved using a different saturation or band-limiting process on each Aux»

Clip Gain was used extensively to control levels before the signal hit the Pro Tools mixer. This is just one small group of backing vocal clips!

Rescued This Month

Kirsty Cooke is a solo songwriter, vocalist and producer based in Somerset, UK. Her writing style varies depending on the message she needs to convey in her music, fluctuating from electronic pop and trip-hop styles through to more cinematic emotional songs such as 'Heydays', hence taking on an eclectic mix of references during writing. Kirsty spent three years studying Digital Music at Brighton University back in 2004 where she discovered her love of music for film and soundtrack, and tries to draw this type of emotion into her music. She releases independently and you can listen on all streaming platforms. 'Heydays' will be available to stream and download on all platforms from 20th August.

Leapwing Audio's StageOne stereo width plug-in was used to spread the backing vocals.

The Perils Of Doubling

One thing that puzzled me about Kirsty's own mix of 'Heydays' was a sudden huge jump in the level of the lead vocal, about a minute in. Opening the multitrack revealed a not uncommon problem in mixing. To try to give the vocal a stereo spread, she had duplicated it and panned the copies hard left and right; but unless these duplicates are shifted in time or pitch, they'll simply recombine with the original part to give a louder mono vocal. To achieve any sense of width, the duplicates need to be processed to make them different from the main part, and each other.

» but also the creation of separate, deliberately 'characterful' reverb treatments. In one case, for example, I fed a rotary speaker emulation from IK's Mixbox into SoundToys' Little Plate; elsewhere, I had a lot of fun with the more weird and wonderful IRs from IQ-Reverb. Finally, reasoning that the centre of the stereo field belonged to the lead vocal, I bussed all the backing vocal Auxes to a global Aux where I used Leapwing Audio's StageOne to spread them as wide as I dared.

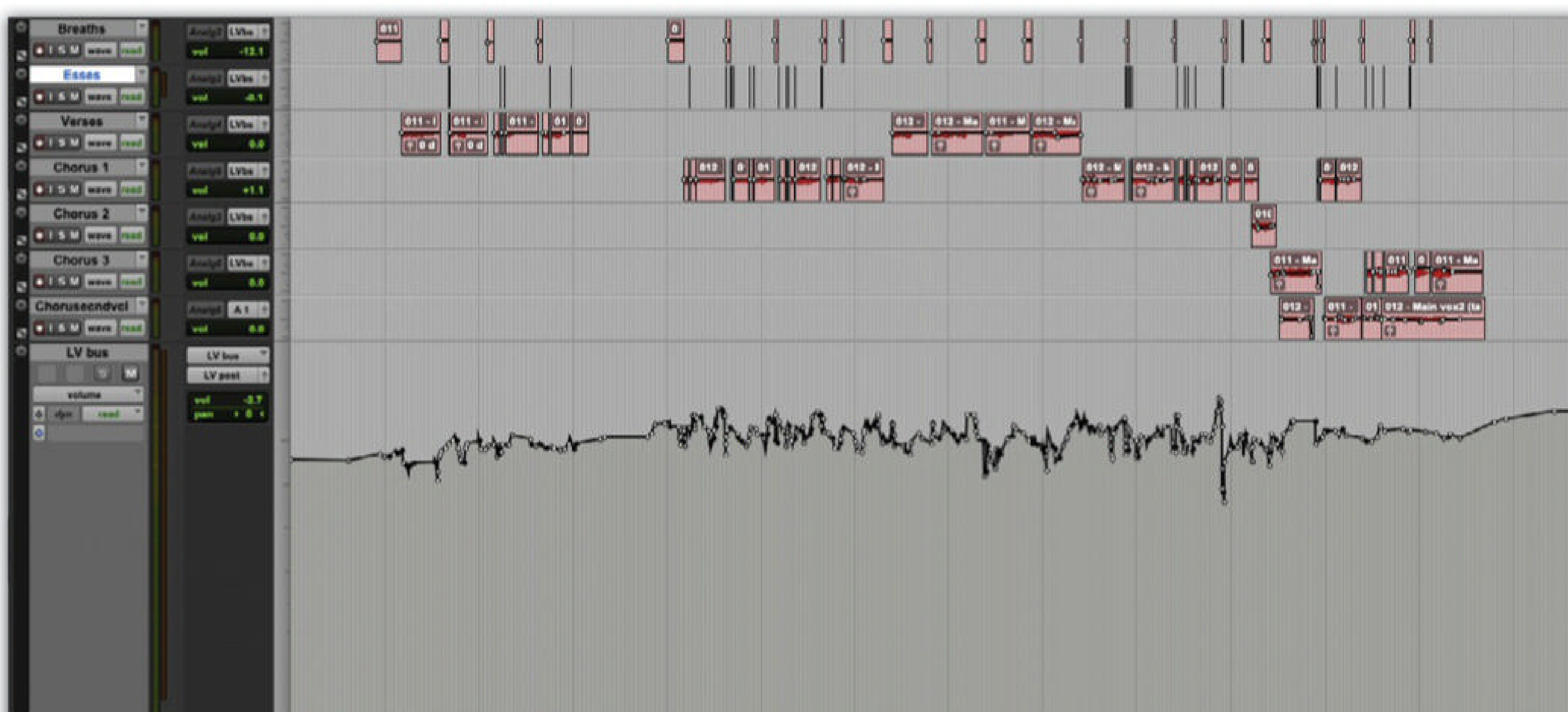
Leading Role

In contrast with the backing vocals, the lead vocal needed to sound as natural as possible — and, perversely, this meant doing quite a lot to it! Reducing breaths and 'ess' sounds in level by exactly the right amount to create the illusion of naturalness is always a delicate balancing act, and I decided this would be achieved most easily by splitting them out to a separate track. Doing this also allows them to be processed differently, which can be handy if for instance you want to use a super-bright vocal reverb; in this case, though, I bussed all the lead elements to a single Aux track to recombine them and apply processing. This included Oeksound's Spiff plug-in, which I find to be very effective at attenuating unwanted spittiness and other transient elements.

Conventional EQ is often of limited value on this sort of vocal, because the same treatment that enhances a softly sung verse makes a powerful chorus harsh or brittle. Again, there are several ways you can get around this. You can split the verses and choruses out to separate tracks and apply different treatments. You can automate an EQ plug-in to apply different settings at different points in the song. Or you can do what I did, which is to use multiband dynamics, in this case FabFilter's Pro-MB. I divided the frequency spectrum up into three bands, set so that quiet signals would receive a small gain boost in the midrange and highs, but loud ones would provoke compression in the same regions.

Reverb is crucial on an exposed ballad vocal, but again, it can be difficult to get right. The treatment needs to give the vocal that all-important sense of spacious, lavish opulence, but at the same time, it needs to be intimate, as though the singer is standing next to you. The key here is to use plenty of pre-delay to separate the vocal from the reverberation, and to keep the reverb itself quite dark. This led me to a vintage Lexicon-style hall algorithm in Relab's LX480 Essentials, augmented in small doses by a much larger patch from EastWest's QL Spaces II.

Another thing to bear in mind is that you don't always want the vocal reverb and other effects to remain static »



Achieving a natural lead vocal sound meant splitting problem sounds such as esses and breaths to their own separate tracks.

throughout the song. It's worth looking for opportunities to use delay throws to highlight single words, moments where you might want to ride up or duck the reverb levels, and so on. In the mix of 'Heydays' I tried to add a subtle extra dimension to the choruses by bringing in a very short, pitch-shifted stereo delay.

Keeping the vocal at the right level against the backing required not only the use of a conventional compressor and the aforementioned Clip Gain adjustments, but also plenty of fader automation. The last chorus featured a second lead vocal; this received similar treatment, but with the addition of some stereo spreading from SoundToys' Microshift to try to keep it out of the way of the first.

Pulling The Strings

At this stage, I had what seemed to me a pretty great-sounding piano ballad. But whenever I tried to reintegrate the other elements of the arrangement, it just didn't work. No amount of EQ, reverb or compression seemed to help, and I became convinced that the problem was down to the arrangement. So, with Kirsty's permission, I hopped across the line that separates mixing and production.

I liked Kirsty's idea of using a deep, sustained, almost organ-like sound to fill out the very bottom end of the mix,

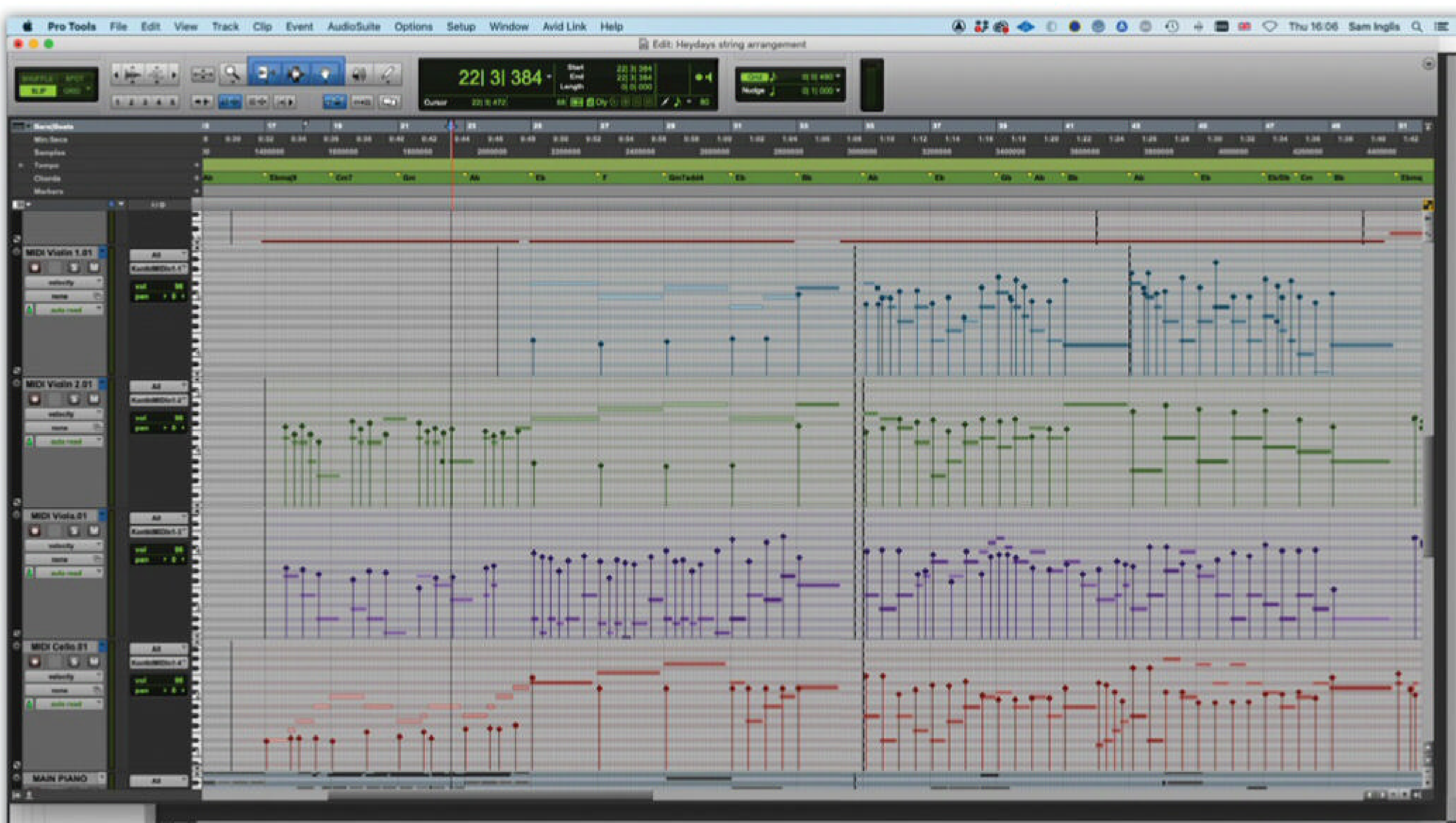
A suitably dark-sounding lead vocal reverb was derived from Relab's LX480 Essentials.

but her part was perhaps a bit too busy, with occasional wide leaps and staccato notes that just manifested themselves as a momentary loss of low end. This was easily sorted by simply removing any note shorter than a minim, and making sure that the remaining long notes didn't stray on to the fifth of the chord. A suitably drone-like bass patch in Sonic Academy's well-equipped ANA 2 completed the job.

The string arrangement was rather more challenging! Kirsty's original arrangement featured just viola and cello, with no violins. The cello part contained two lines, with a lot of thirds and other closely spaced intervals that wouldn't usually be found in the nether regions of a string arrangement. These congested lower registers, combined with a general lack of motion, meant that it had a tendency to disappear into and muddy the mix.

Rather than use Kirsty's original as a template, I decided to see if I could write a completely new arrangement. What was needed, I decided, was not a full symphonic string section, but a more intimate chamber ensemble. Spitfire Audio advised that their Chamber Strings should be perfect, and so it proved.

My first step was to bounce out the existing vocal-and-piano mix and load it into a new Pro Tools session, allowing me to lower the buffer size in order to play string parts from a keyboard without noticeable latency. Then, I worked out the chords from the piano part and wrote them to the Chord track in Pro Tools. This revealed a slightly unusual song structure, with a bridge section that appeared only after the first verse, and a restless chord sequence in the chorus. Each chorus



The string arrangement was drafted in a separate Pro Tools session, with the Chord track (top) providing a helpful reference.

» was preceded by the dominant chord of Bb; however, instead of resolving immediately to Eb, the chorus began on the subdominant, and only visited the tonic chord in passing before working its way back to the dominant again.

A Bluffer's Guide To String Arranging

With the vocal melody more or less internalised through repeated playing, and the chords laid out in front of me, I worked through the song in sections, sketching out and refining ideas. I am no Igor Stravinsky, so any attempt on my part to compose anything of this sort takes a long time, involves a lot of trial and error, and generally results in quite a bit of cursing. What follows should be taken as the lessons of bitter experience rather than formal training or theoretical correctness! With that in mind, here are a few thoughts that might be helpful to anyone in a similar position:

- Less is more. Don't feel that every bar must have string action, or that all the instruments need to be playing all the time. Solo lines or two-part harmonies can be very effective, especially if they're juxtaposed with denser arrangement elsewhere.
- Rather than just playing block chords to reinforce the harmony, try to think of melodic or motivic ideas that mirror or answer what's going on in other parts. These can always be simplified if they get too busy.
- Don't be lazy. Once you've come up with an arrangement that works for the first verse or chorus, don't automatically copy and paste it to the others. Think about how it might be varied, extended or built on to take the listener on a journey through the song.
- If you're struggling to get started, begin with the outer parts — usually

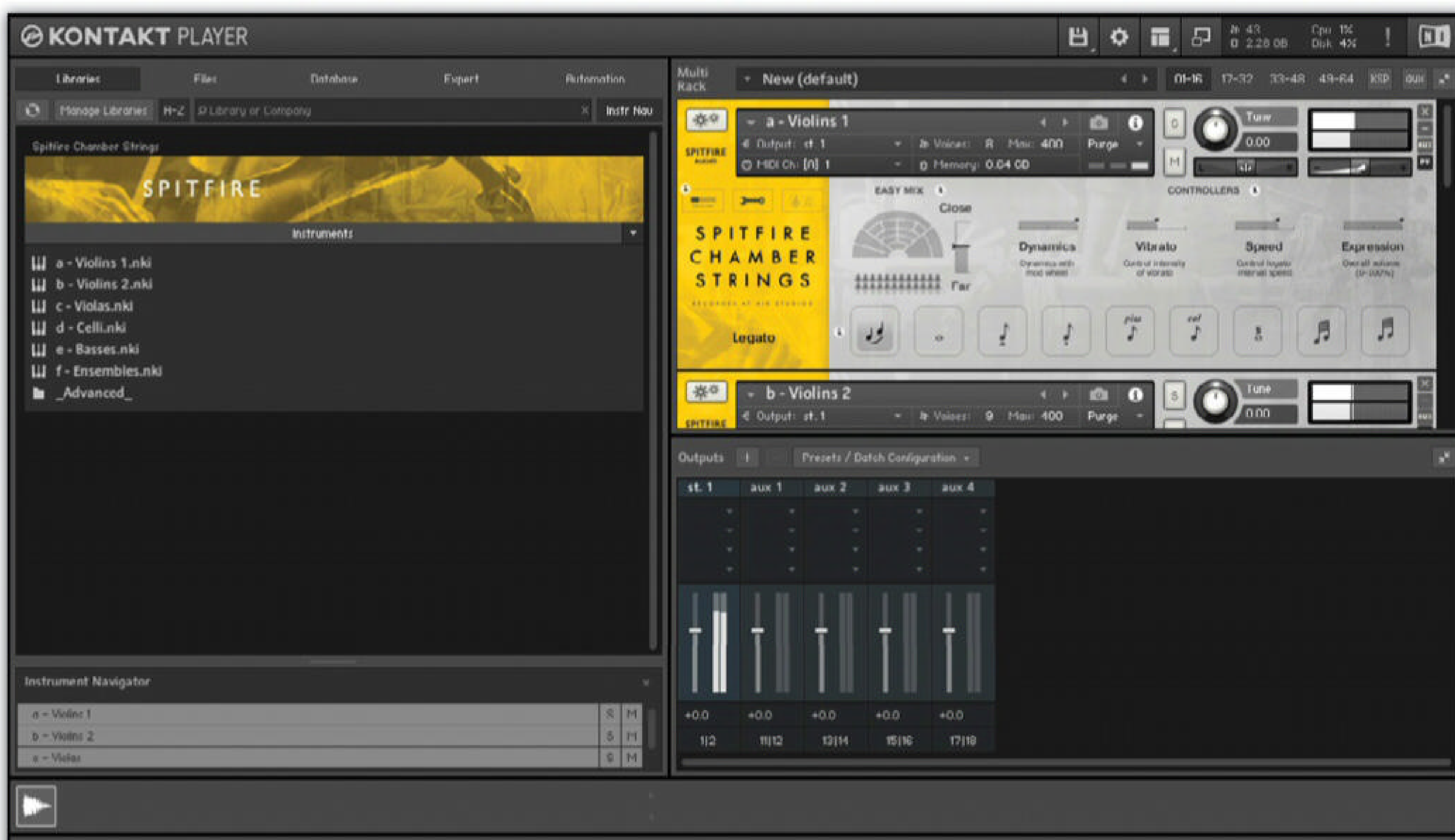
Remix Reaction

"I could hear in my head exactly how I wanted the song to sound: big, emotive but well balanced. I sought advice on the *SOS* Forum as the mix was sounding muddy and unbalanced. I wanted my vocal to stand out, but it was overpowering the instrumentation as well as the backing vocals drowning each other out. I got a to a point where I managed to mix the vocals to a better standard, however the main vocal was always far too loud during the chorus after mixdown when I attempted to double the track and give it some stereo width. On top of this the instrumentation was dwarfed by the many vocal backing tracks during chorus sections and I struggled getting the levels to sit right without peaking it despite spending many hours playing with automation, EQ and plug-ins.

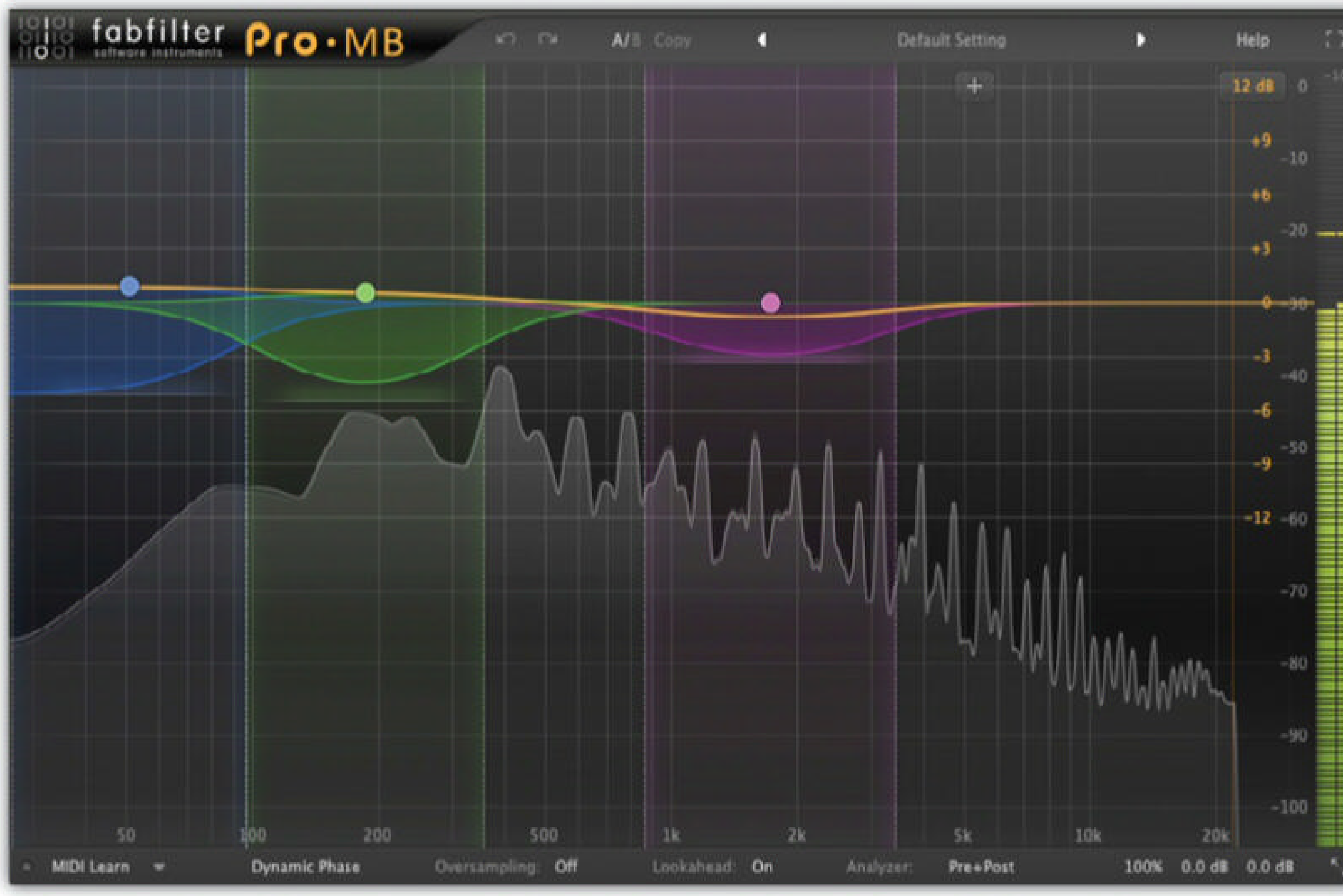
"When Sam offered to assist with the mixing and string arrangement, I was excited to see what he could come up with. I can't lie, I was a blubbering wreck when I first listened to his new arrangement/mix as it just sounded so emotive and beautiful, with big cinematic movements that are utterly captivating. When I originally wrote the song it was going to be a piano and vocal piece only, but it just didn't seem up to par with the lyrical content, hence I added cello, bass and viola to fatten it up, but there was a certain edge missing.

"My original mix didn't accommodate the lower frequencies enough to give a very rich sound, my piano was especially lacking in the low end. I mentioned to Sam I loved the piano timbre in John Lennon's 'Imagine' and I think he did a fantastic job of emulating this, which is really very special for me as 'Imagine' was a song that my father loved and this song was written for him.

"The addition of the string arrangement Sam added has really helped all those vocal parts to slot into place without overpowering the instruments. The track still feels vocally led but now has that 'big fat' sound with the nicely balanced 'big notes' that I heard originally in my head. I had mentioned to Sam that I like the build-ups and big orchestral sounds of Florence Welch and he has delivered beautifully on all these references. Thank you, Sam for bringing this song up to its potential, you have done a wonderful job!"



Spitfire Audio's Chamber Strings offered an intimate string sound which was just right for the song.



Ducking in action: the upper-mid band is keyed from the external side-chain input, named 'LV Limit Out', with a heavily limited version of the lead vocal.

the first violins and the cellos — and fill in the inner parts later.

- Whilst you don't need to follow all the 'rules' of harmony in order to produce an effective arrangement, it's worth knowing some of the most important ones. They are rules because, by and large, the things they prohibit tend to sound bad! So, in general, it's desirable to avoid parallel fifths and octaves; don't use chords in the second inversion except under very specific circumstances; aim for wide spacings between the lowest parts and closer ones in the upper parts; and avoid doubling the fifth note of a chord if you have the option to double the root or third instead.
- Any good string library will include different articulations such as pizzicato, marcato, staccato and so on, which can be selected using keyswitches — low notes on the keyboard that are below the range of the actual instruments. Used judiciously, these can add interest, life and realism to your arrangement. Used injudiciously, they can sound like a complete mess. The more exotic ones should be avoided unless you really know what you're doing!
- String sounds take time to 'speak'. In MIDI terms, this means that hard-quantised notes will often sound late. Go with what sounds right, not what looks right.
- Be open minded to the possibility that the ideas you come up with on a MIDI keyboard might actually work better with other sounds, not just strings.

Putting it All Together

To my mind, the ideal string arrangement is one that adds melodic and harmonic interest to the song whilst supporting, rather than overshadowing, the other elements. Several times I found my own attempts yo-yoing between 'boring' and 'fussy', and it took plenty of MIDI editing to prune them to a reasonably happy medium. I tried to reinforce the dynamics of the song by using soft, staccato sounds in the quieter choruses, and switching to legato for the more dramatic choruses. I even sneaked in a bar of trills at the end of the bridge.

Whatever its musical merits, or otherwise, the new arrangement at least made life easier from a mix point of view. With fewer notes crammed into the congested area below Middle C, it didn't clog up the low midrange; and its more melodic, mobile nature helped it stand out even at a fairly low level. It didn't hurt that Spitfire's Chamber Strings sounds great, too.

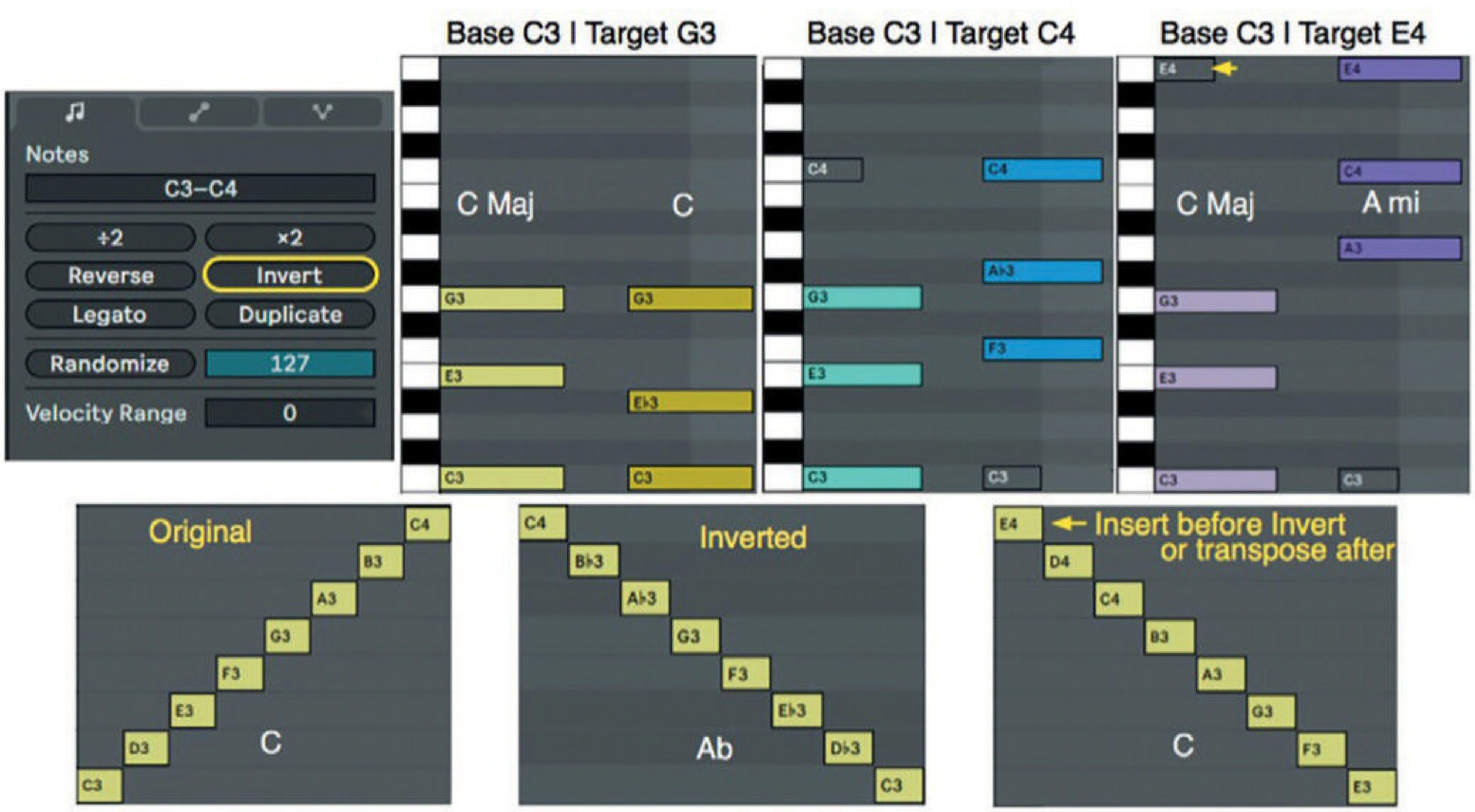
Nevertheless, fitting everything together was still a challenge, and there was a tendency for other elements to fight with the lead vocal for attention during the chorus. To keep the lead vocal in its rightful position — in front of the mix, but not so much so as to make the rest of the track sound small — I routed all the backing vocals and all the instruments to stereo Auxes, so I could use the vocal as a trigger to duck them.

The most basic way to set up ducking is to place a compressor across the backing Aux and send directly from the vocal to its side-chain input. The problem with this is that you get more ducking when the vocal is loud, and less or none when the vocal is quiet — yet it's the loudest sections of the vocal that need less help from ducking. To counteract this tendency, I sent from the vocal to a separate mono Aux, used a limiter plug-in to completely flatline it, and then sent that signal to the compressor side-chain. This makes it easy to achieve a consistent amount of gain reduction. Rather than using a conventional compressor on the instrument Aux, I used another instance of Pro-MB to target the upper midrange where things needed to be kept particularly clear for the vocals. There was also a whole lot more compression going on on the mix bus than I'd normally be comfortable with, but that seemed the best way to achieve the sense that everything else was present, yet behind the voice.

Some songs invite you to throw the kitchen sink at them, and I did consider adding drums or electric guitars too. 'Heydays' is certainly strong enough to stand the treatment, but by this point it was working pretty well in its current incarnation, and I decided that I'd stepped far enough beyond the remit of the mix engineer already. The problem with mission creep is that, unless you keep a firm watch on it, it never stops!

Hear For Yourself

Audio examples to accompany this article can be heard by visiting http://sosm.ag/mix-rescue-1021.



1: A C-major triad is inverted with different targets (top), and a C-major scale is inverted and transposed up a major third (bottom).

Introduce some creative weirdness with MIDI inversion.

LEN SASSO

This month we'll have a look at MIDI inversion, a simple but sometimes bewildering technique for breathing new life into your MIDI clips and streams. Inversion changes the relationship between notes in a chord or melody so that up becomes down while distance is preserved. For example, a major triad will invert to a minor triad, and a melody will invert vertically to a mirror image of itself. But the root of the inverted triad or key of the inverted melody might be anything. For the complete answer we need to know the 'axis' around which the inversion is measured. In Live it is possible to invert MIDI in clips or in real time. We'll start with clips, where the process is easier to understand, and then build a MIDI Effect Rack for inverting MIDI on the fly.

When Up Is Down

For MIDI clips, the Invert button in the Notes tab of Live's MIDI clip editor does the job, but in an unusual way. Instead of rotating notes around an axis, it measures each note's distance above the lowest note in the clip (the 'base') and then moves it to that distance below the highest note in the clip (the 'target'). That in itself can produce useful results, but it doesn't give you control over the axis. However, that's easily fixed by adding the desired top and bottom notes when the existing ones don't suit. (Deactivate the added notes so that they don't play in the inverted clip.) For example, if you want to have C as the axis of rotation, ensure that both the top and bottom notes are Cs, invert the clip and then transpose up or down by octaves to move the top C to the axis you want. In the middle example at the top of screen 1, you would transpose the F-minor chord down one octave to get C3 as the axis of inversion.

You'll notice that the key usually changes when you invert. If you invert an ascending C-major scale, you'll wind up with a descending C Phrygian scale (Ab-major scale played from C to C). You can work around the scale change by transposing after the fact, or as just described, by adding a note (an E in this case) above the original scale before

2: A MIDI Inverter Rack for processing incoming MIDI played live or from a clip or generated by a sequencer.

3: A Pass Thru chain is added to the Inverter Rack from screen 2 to allow unprocessed and inverted segments of incoming MIDI.

inverting. One way to think about the scale change is to consider the sequence of intervals that make up the scale. For an ascending major scale this is 2212221 measured in semitones. Inverting the scale gives you the same sequence descending, resulting in the C Phrygian scale. Transposing that up a major third gives you an E Phrygian scale, taking you back to the key of C.

A handy thing to keep in mind is that when some notes in a clip are selected, only those notes are inverted. For example, if you start with a C-major triad followed by an F-major triad and select them one at a time to invert, you'll get a C-minor triad followed by an F-minor triad, whereas if you invert them together, you'll get an F-minor triad followed by a C-minor triad. The MIDI Note editor's Fold button is another useful tool. With Fold engaged, all notes are pitch-corrected to notes in the folded view. For example, a folded ascending C-major scale inverts to a descending C-major scale rather than to a descending C Phrygian scale.

Who Played That?

The first step in building a MIDI Effect Rack to invert incoming MIDI is to set up Live's Scale MIDI effect to do the inversion. The Scale effect presents a 12-wide by 13-high note matrix with incoming pitches indicated by their horizontal position. Blue squares in the matrix indicate the outgoing pitch (vertical axis) corresponding to each incoming pitch. In the default setting they occupy the ascending diagonal from C to B, so that incoming equals outgoing. For inversion, we want to change that to the descending diagonal from the top C down to C#. The only other Scale effect parameter we'll use is Transpose, which I'll get to in a bit.

Place an instrument after the descending Scale effect, play a chromatic scale ascending from C3 (middle C) and you'll hear a chromatic scale descending from C4. When you reach C4, the pitch will jump two octaves to C5. Because we want the scale to continue to descend across the full keyboard, we need multi-octave transposes at each of these octave transitions. This is the tricky part in building the Rack, and it's a bit simpler if we limit the note range to C1 through B7. Create a MIDI effect Rack with nine chains — one for each C-to-B octave as shown at the bottom of screen 2. Set the Rack's Key zones so that each chain only receives the notes in its octave. Insert a Pitch MIDI effect in each chain, set the C1 chain's Pitch effect to +96 st and then subtract 24 for each subsequent chain. For example, the C3 chain will have a 0 st Pitch setting (or no Pitch effect) and the C7 chain will have a -96 st Pitch setting. The chains will then transpose their incoming C-to-B Key zones so that the full C1 to B7 MIDI note range is inverted when passed through the Scale effect.

To finish off the inverter Rack, insert a Pitch effect before the Octave Zones Rack, insert another Pitch effect after the Scale effect and gather everything in a new Rack. Assign the new Rack's Macro knobs 1 and 2 to the first Pitch effect's Lowest and Range settings — they set the range of notes that will be inverted. Assign Macro knob 3 ('Axis') to Scale's Transpose; that transposes the output after inversion. Finally, assign Macro 4 to the last Pitch effect's Pitch knob. That's used for transposing the inverted output (ie. the axis) by octaves. The first Pitch effect blocks notes outside of its range, so if you enclose the Inverter Rack in another Rack and add an empty chain with a Pitch effect or a restricted key range, you can split the incoming note stream between regular and inverted segments playing different instruments. This can be lots of fun with a full-size MIDI keyboard. With a +4 Axis setting the descending keyboard geography mirrors the ascending geography, so that playing downward from E with the left hand mimics playing upward from C with the right hand (Google 'Joe Zawinul inverted keyboard').

The February 2019 Live column, 'Melody Makers', covers a number of step sequencers, all of which make good sources for inversion. Step Sequencer from the free M4L Big Three Pack is a good place to start: insert it before the Inverter Rack, add an empty chain to the Rack, insert a Pitch effect in that chain and map the Rack's Range Lowest knob to the Pitch effect's Range setting. Range Lowest will now split the input between inverted and normal. Insert a polyphonic instrument after the Rack and start the sequencer.

A Different Drummer

Inversion is also a useful tool for repurposing drum clips. Screen 3 starts with the Azimuth kit and five Azimuth MIDI clips in Live's Beat Tools Pack. A chain holding a Pitch MIDI effect is added to the Inverter Rack from screen 2 to allow some incoming MIDI notes (kicks, snares and hi-hats, for example) to be passed directly to the Drum Rack. Macro knobs have been added to separate the ranges of the Pass Thru and Inverter chains and to mute either chain. The simplest approach is to insert the Drum Rack after the Drum Kit Inverter Rack, start a clip playing and twiddle the knobs. The settings shown pass the kick, snares, hi-hat and clap unprocessed and send all other notes through the inverter. The Axis and Xpose knobs control what happens to the inverted notes, letting you shuffle the remaining kit pieces — voice, synth, guitar and sound effects in this case. A more flexible approach is to add a MIDI track to capture each chain's output and to route the output of those tracks to the drum kit. I've done that in screen 3 and added Follow Actions to the captured clips to produce an ever-changing drum track. Try the same strategy with audio clips sliced to Drum Racks (dialogue, scat singing, chord progressions, sound effects and so on).

STEPHEN BENNETT

We explore Logic Pro's bounce and export options.

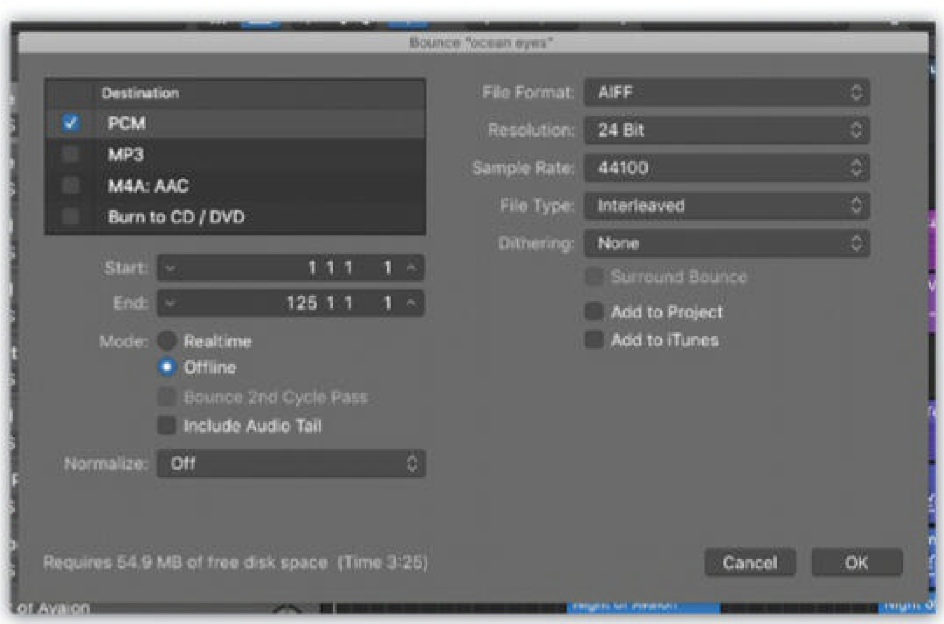
The Level Meter and Loudness Meter plug-ins offer plenty of options for keeping an eye on your track's levels.

Though it may seem that things were simpler in the 'analogue days', delivering content for release on vinyl or cassette was fraught with problems. Variables such as differing tape noise-reduction systems, badly maintained or misaligned master tape machines and equalisation choices in the cutting room could mean disappointment when you finally got to spin your latest opus. It got easier for a while in the early days of digital as all you needed to do was supply a 44.1kHz, 24-bit file (or CD-R) to your manufacturing plant and you'd be reasonably sure it would sound the same on CD playback. But with the explosion in the delivery ecosystems for music, you need to be sure that any files that you send to a streaming site, duplication plant or mastering engineer are of the correct format.

All About The Basics

Most of the time, you'll be bouncing out (or rendering) stereo files, and the format for these depends on where your music will end up. Logic's Bounce and Export features are accessed from the File menu. If you open the Bounce window, you'll see that several options are available. You can bounce the whole project or just a selection, in which case you need to make sure the start and end parameters are correctly set — these values are also the same as those for a Cycle region. If you have no external hardware patched in, choose Offline bounce, otherwise you'll need to wait while the whole track plays as it renders to stereo in real time. The Include audio tail option is there to make sure any reverb or echo at the end of the bounce is captured, but sometimes this can extend the song length — so it's better to make sure that the last bar in the bounce is beyond any residual processing. You can load the bounce back into Logic for further editing if you wish.

The 2nd Cycle Pass option performs a 'practice' bounce once before looping back to complete the actual rendering. This can be useful when you want to have any effects from the end of your project, such as a reverb, incorporated into the start of the actual bounce — such as when making loops. The settings for the bounce will depend on the ultimate destination of your rendered audio.



Logic's various Bounce options.

Your Level Best

The only rule here is that the output level on the master Track should not go beyond 0dB. While the internal processing in Logic means that it's unlikely you'll clip any individual channel, some plug-ins that emulate vintage processors expect to see a level that is closer to that which the real hardware would accept. It's perfectly possible to reduce the stereo master fader to avoid your bounce clipping, but it is much better practice to avoid this by properly setting the gain of individual channels. What loudness level your bounce should be depends on the destination of your audio and, perhaps, the 'market' at which you are aiming your music. Logic has a few metering tools that will help you set the correct levels, so insert a Level Meter and Loudness Meter (both are found in the Metering menu) as the last plug-ins on your Stereo master channel.

Probably the most useful setting on the Level Meter is True Peak & RMS, as this will alert you to any unexpected transients as well as give you an idea of the general loudness of the track. It's instructive to play your reference tracks through your Level Meter too — you'll find that while peaks are often hitting close to 0dBFS, the RMS usually sits at or below -12dBFS.

The Loudness Meter displays the Loudness Units Full Scale (LUFS) of your file — the smaller the value, the greater the apparent loudness. It's useful to know the LUFS of your bounce as some online platforms specify a maximum value. Setting the correct level for a bounce is usually part of the mastering process though, so if you're bouncing out to send off to someone

with magical ears (and a treated studio), you'll want your master track's metering bouncing around at the -12 to -18 dB level to give the mastering process some headroom.

Formats

If you're bouncing your project as a file destined for CD manufacture, you'll choose PCM, AIFF (or WAV), 16-bit interleaved with dithering. This latter process is seen as a bit of a black art but dithering introduces noise into the bounce to 'cover up' the effects of the reduction in bit depth. Some people can spot the quality of the different dither types, so if you are one of these, you can experiment with the different dither types available. For the rest of us, the UV22HR option should be fine.

Once bounced, the CD manufacturing process shouldn't adulterate the audio further. For all other platforms, you'll need to take their advice as to what format and loudness of audio they expect. For example, Spotify recommend the ITU 1770 (International Telecommunication Union) standard of -14dB LUFS. Most platforms will adjust your audio automatically to the levels they require, so if you don't want them to meddle with your carefully created tracks, it's essential that you give them exactly what they require. But as a rule of thumb, for mastering, export your audio at the sample rate you recorded it at, with 24-bit depth and plenty of headroom, while for streaming, export your audio at 24-bit and at a LUFS level requested by the platform. Logic's Adaptive Limiter, placed just before your meters, can help increase the average apparent loudness of your bounce, so it's worth experimenting with this — but be aware that too much gain reduction applied here can have deleterious effects on the audio, and a proper mastering process is a better option. You can also use this plug-in to set the output ceiling to make sure your audio never clips the output — but again, this is no cure for proper gain staging elsewhere in your project.

It's possible to bounce your homemade loops directly into the Apple Loops library, along with metadata for tempo, key, genre and so on.

You can, of course, also bounce MP3 and AAC files directly from Logic if you want a quick and dirty way to audition your mix or, if you're feeling brave and have a suitable burner, create a CD-R from your project. If you've recorded at high sample rates, you might need to downsample your audio for distribution — I usually do this as a separate process with my bounced files. The quality of the various methods of sample rate conversion is quite a hot topic and, while it's true that not all processing is equal, I have found that loading higher sample rate audio into a Logic project set to 44.1 or 48 kHz produces very acceptable results when bounced out.

Stem The Tide

Increasingly, engineers are requesting stems rather than complete mixes. Those working in the film and TV world have been supplying these for a while now and I've produced many stems this year for my online collaborations. Fundamentally, stems are just submixes of tracks. For example you might create stems of vocals, basses, drums, guitars and keyboards. The creation of stems makes any post-mixing balancing, and the creation of vocal-free mixes for advertisements and so on, much easier. As I write, Logic unfortunately doesn't (yet) have a 'stem bounce' feature, and creating them can take a little planning and thought, depending on the way you work. We covered the creation of Stems for those working in the box and in a 'hybrid' fashion when using external hardware, in the May 2021 issue: www.soundonsound.com/techniques/exporting-logic-projects-other-daws.

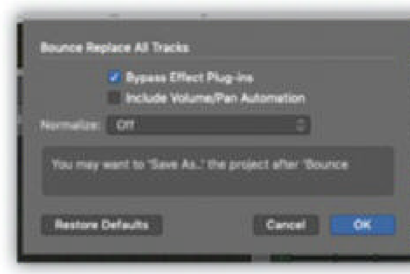
Going Loopy

Many people use Logic to produce beats and loops. These can be created from a set of Regions, or by using the Live Loops window (covered in the October 2020 issue: www.soundonsound.com/techniques/live-loops-logic. The Export Region/Cell window has various options to make sure you have all the metadata required when using these loops in other projects.

Here you can determine if the beat loops and/or conforms to the project tempo or not. Choose the Instrument Descriptors carefully, as these will be searchable from the Loop Browser. You can set the Musical Scale and Key, if required, and a tempo for the beat or loop. Once created, these are accessible from Logic's Loop Browser and can, of course, be exported as audio files by dragging them to the Arrange page, Ctrl-clicking on the Region created and selecting Export as Audio file.

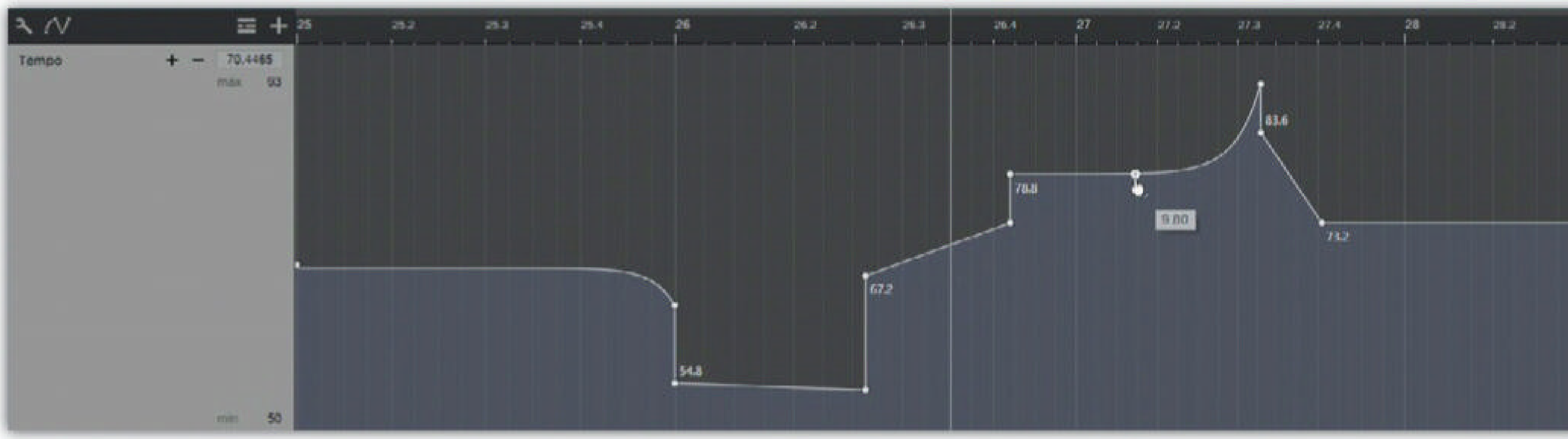
The Collaborators

When I'm working remotely with others, I normally send them stems or audio files even if they are using Logic, as you can never guarantee that the other person will have access to all your instruments or plug-ins. If they are using Logic, I usually use the strangely named 'Replace All Tracks', to 'burn' the plug-ins and instruments as audio files. This is also a quick and easy way to make sure that when you return to a project in the future, you can still access any plug-ins that have been rendered obsolete by the march of technology. You can choose to bypass all plug-ins in your render, and whether to include automation in the bounce. If my collaborator is not a Logic user, I'll also include an exported MIDI file. Not only will this contain any musical notation for the instrument parts, but also any tempo and time-signature changes in the project (I make that kind of music!). My colleague can then be sure that everything lines up when adding further recordings.



The Replace All Tracks option is useful for sharing your audio with people who don't use Logic.

Andrew S Tanenbaum once wrote, "The good thing about standards is that there are so many to choose from," and the digital music age has brought about an explosion of different distribution outlets, all requiring different types of content. It's always worth making sure what format is required by each platform but, with a little thought, Logic should be able to generate all of the files required.



The Tempo Track allows you to change the tempo of your song, either abruptly, in a straight ramp, or as a smooth curve.

Manipulate the timing of MIDI and audio with Studio One's Tempo Track.

ROBIN VINCENT

Trying to keep everything together is a bit of a lifelong mantra for musicians. Whether you prefer to languish in free-form movements or keep things as tight as an atomic clock, you inevitably need to pull disparate parts from disperate tracks towards some kind of common sense of rhythm. Let's take a look at the Tempo Track in Studio One and see if it can help us hit that downbeat with everything all at once.

The Tempo Track resides at the top of the tracks, in the same place you'd find the Chord Track, Marker Track, Arranger Track and so on. It's shown with a clock icon, but if you can't see that you'll find it under the 'Global Track Visibility' icon which looks like a hamburger with a single beady eye staring at you from the meat layer.

Stuck In The MIDI

If your song is made up of MIDI tracks then the Tempo Track can be a useful place for messing about with the timing of your song. When you open it, you'll find it's there as a single line set at the song's tempo, as defined in the transport bar. In the header for the Tempo Track you can enter a tempo and the line will change to reflect that. You can also change the minimum and maximum tempo range to keep your focus where you need it and give a better resolution.

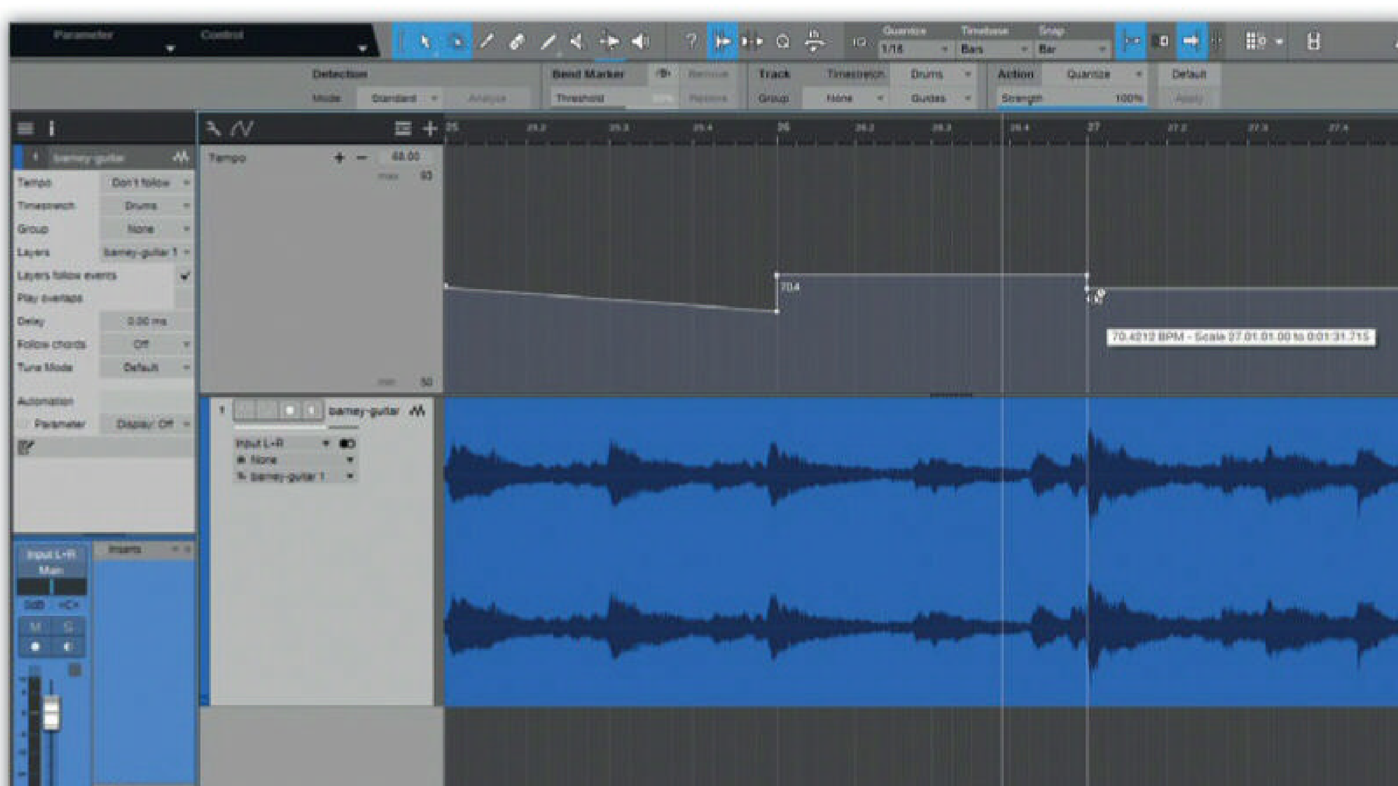
Your mouse pointer has two functions within the Tempo Track. Firstly, in the lower section of the track it becomes a pointy finger, which lets you add a tempo change at that point on the line. Secondly, if you hover towards the top of the track the cursor becomes a trim tool, shown by a short horizontal line with a vertical line on either end. If you click and drag up or down you'll move the entire tempo or selection up and down.

Inserting a new tempo change gives you a node that you can move about and place along the timeline, drawing a straight line from the original tempo point. A central node between the two lets you pull in a curve if you want a less linear movement from one tempo to another. To jump tempos rather than transition, it's easier to enter the tempo value into the box on the track header and hit the '+' to enter the change at the position of the song pointer.

You can create quite complex lines of timing change, and there are a couple of ways of viewing them in the timeline. In the top toolbar under Timebase you have two options: Time-linear and Beat-Linear. With Time-Linear the song pointer is always moving at a constant speed, so when you change the tempo the bar lines on the timeline expand and contract to compensate for the changes. If you select Beat-Linear then the bar lines remain a constant width while the song pointer speeds up or slows down depending on the tempo.

Conforming Audio

Once you introduce audio files to a tempo environment things can get out of hand. Studio One has lots of automated processes that try to make conforming the tempo of audio as painless as possible. If an audio file has tempo information built into it then it will automatically adapt that to the song tempo using time-stretching.



Conforming a project's tempo to an existing audio recording.

It's also possible to warp audio so that it plays in time with the rest of your project.

If an audio file doesn't have an associated tempo then Studio One can attempt to detect it and then do the time-stretching after that. This is particularly effective with loops and other tempo-based audio files, but if you are pulling in a recording of a guitar, drums or other instrument then the tempo might be a bit more up for discussion.

Here's one tip on doing some manual time stretching if you do have a piece of audio with a constant tempo but which Studio One doesn't quite nail automatically. Open the Inspector and make sure Tempo is set to Timestretch. Then put your cursor at the end of the clip and hold Alt+Ctrl (Windows) or Opt+Cmd (Mac), then click and drag the end of the clip and it will time-stretch or time-compress the file. Then all you do is move it until the obvious start of a bar visually lines up with the bar lines.

Things are rarely that easy though, so a bit of manual tempo editing might be required to pull the audio in line with the song tempo or the tempo in line with the audio.

Tempo Maps

If you like the groove of an audio recording, then you can create a tempo map that will follow the variations in tempo, which you can then apply to all your MIDI tracks to pull them in time. This is perfect for trying to build a song around an acoustic guitar track, or if you are wanting to develop some MIDI instruments around a multitrack recording of a band. Choose a track that has a lot of obvious rhythm; drums are perfect but that acoustic guitar track will work too.

For this we don't want Studio One to do any automatic time-stretching, so open the Inspector on the audio track and set the Tempo option to 'Don't Follow'. You'll want to start with a rough tempo for the audio track, which you can do by playing the track and then use the Tap-Tempo feature to give you a bpm. Click your mouse a few times on the word 'Tempo' on the transport bar in time to the track, and you will have tapped the tempo into the transport bar — it's quite a brilliant little feature. Next, visually line up the start of your audio track with the start of a bar. If you turn on the metronome you should be able to hear where it goes in and out of time, and this is where the Tempo Track comes in. It's all done visually because you can see the rhythmic material in the audio track, and we want that to line up with the bar lines in Studio One. So, on the next bar hold Ctrl (Windows) or Cmd (Mac) and your cursor in the Tempo Track will turn into a Range tool. With Snap enabled the tool will snap to the bar line, and you can click and drag it until the line matches up with the appropriate transient on the audio track. On either side of the tool the tempo will increase or decrease reflecting the change. If the Timebase is set to Time-Linear mode then the bars will expand and contract with the change; if you're in Beat-Linear mode then the audio file will expand and contract instead. I find I prefer Time-Linear so I can see the bars change as I edit the tempo.

Once you've gone along adjusting the tempo for every bar where it's necessary, listen to it again with the metronome and you should find that it's all in time. Now your MIDI tracks will follow along as if it's the most natural thing in the world.

You can make this a little bit easier by detecting the transients on the audio file first. This will give you accurate lines on the key segments in the audio file with which to line up the bar lines. To do this, select Audio Bend from the Toolbar and the tools panel will appear. Under Detection hit Analyze and Studio One will work out all the transients in the audio file. It will probably find far too many when really all you need is the start of each bar, so use the Threshold slider to dial things down a bit. Then you can make the manual tempo changes just like before but with a better visual guide. Once you've finished you can remove the transient lines (they can get quite distracting) by right-clicking the audio file and unticking the Bend Marker box.

Bending Audio To Tempo

The other scenario is that you want to break the groove of an audio recording and bend it to the will of the existing tempo. For this you need to use Audio Bend to Analyze the audio track so we can see the transients. Rather than adjusting the tempo to fit the audio we're now going to move the transients to fit the existing bar lines. Select the Bend tool and you'll be able to move the transient Bend Markers, which time-stretches and compresses the audio file either side to bring them in line.

Melodyne Essential

Melodyne Essential, which is built into Studio One, also has some powerful tempo-related functions that are definitely worth exploring. However, most of the documentation and online tutorials refer to the Studio version of Melodyne, and many of the tools and functions mentioned are not available in Melodyne Essential. I think a workshop on exactly what the Essential version can do would make for a useful workshop though, so I'll be covering that in a future article.

JOHN WALDEN

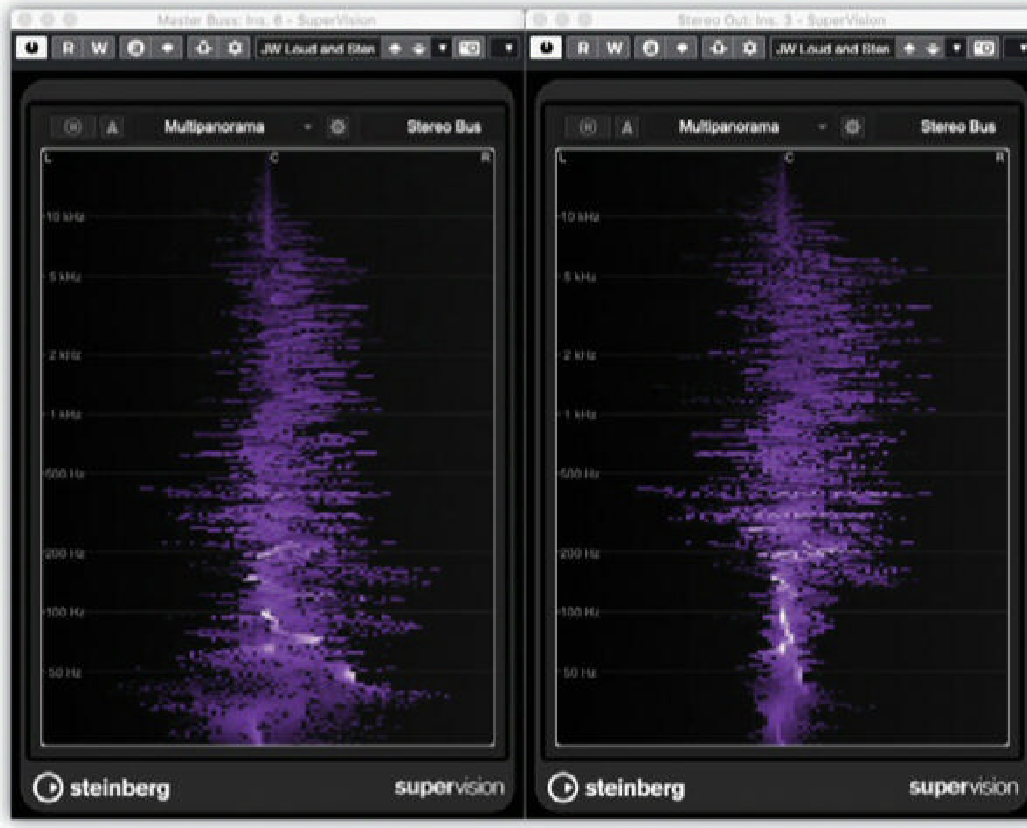
Take full control of your stereo image with Cubase's powerful bundled plug-in suite.

Cubase has long included stock plug-ins for manipulating the stereo image and they all have a role to play, as I'll explain, but the recently added Imager is particularly interesting. It allows you to process up to four different frequency bands and, as is so often the case with multiband processors, it's a powerful tool that's capable of great results in the right context — but it's also fairly easy to screw things up if you don't know what you're doing! In this article, I'll work through a few examples that show you what Imager has to offer, while noting some of the potential pitfalls. You'll find some audio files to accompany each example on the *SOS* website at https://sosm.ag/cubase-1021.

Perhaps Imager's most obvious application is as a mastering-style processor, whether used on the stereo mix bus as you put the final touches to a mix, or to tweak a bounced stereo file. The first screenshot shows Imager as inserted in my main stereo bus processing chain, with a typical configuration for this kind of application that serves as a good starting point for exploring Imager's control set.

I've used all four of the available bands here, but three could easily be enough in this context; you can specify the number of bands at the top-left of the GUI. You can adjust the frequency of the filter crossovers between bands and in this case I've gone for 200Hz, 1kHz and 5kHz, to create low, low-mids, high-mids and high bands. Each active band has three controls. Leaving the Output and Pan controls untouched for now, I've adjusted only the Width in each band. As the control's name suggests, it manipulates the stereo width of its band, with a value of 100 percent leaving the stereo image unchanged and higher/lower values making the image wider or narrower, respectively.

Moving from low to high, I've chosen values of 20, 125, 150 and 170 percent. This keeps kick and bass instruments firmly focussed in the centre (generally a good thing) and gradually adds greater Width through the low-mids, high-mids and highs. There are no hard and fast rules here; you should judge things by ear. But I'm generally cautious about going beyond 150 percent in any band, as stereo enhancement can potentially produce unwanted side-effects when the mix is heard on a mono playback system. Having checked the mono playback compatibility (as I describe below), I felt able to push the width a little harder than usual in this case, for a little extra high-end 'pop'.



SuperVision's Multipanorama module before (left) and after (right) Imager's processing of my worked example.

The next screenshot shows two instances of SuperVision's Multipanorama module, one before and one after Imager, and these make the narrower lows and wider upper-mids and highs easily visible. Check out the accompanying audio example: to my ears, the result is a subtle but rather pleasing widening of the stereo image, while the low end is kept nicely focused.

Imager: up to four bands of stereo width control for Pro and Artist users.

On The Busses

Imager doesn't have to be used on a full mix; it can be just as useful on individual instruments or subgroups. There are plenty of possibilities, but a few simple suggestions should illustrate the potential.

If you need to give your lead vocal a little more space in the centre of the stereo image, try inserting Imager on your backing vocal group channel (bus) and using it to push the backing vocals a little further towards the sides of the stereo image. Unlike with simple panning or single-band wideners, Imager's multiband options give you control over how you spread the main frequencies of the backing vocals.

When you want to help a specific instrument peep out of the mix a little more clearly, Imager can also be a good alternative to a level change. By applying some subtle widening to a specific frequency band or two, you can place the sound more obviously at the edges of the stereo field, which will make it more noticeable to the listener.

For those who are very particular about their reverb treatments, another option is to insert Imager on your reverb's FX Track. A three-band approach that narrows the lows (so you don't get low-end reverb clogging your stereo image), gently widens the mids, and spreads the highs out wide can be fun to experiment with.

Faking stereo from a mono piano recording. Moving from left to right, the three Multipanorama displays show (a) the original mono signal, (b) the result of the MonoToStereo plug-in and (c) combining both MonoToStereo and Imager.

Finally, whether used on an instrument bus or your master bus, automating Imager can be useful in an 'arrangement/production' context. Whether by toggling Imager on/off or using automation to adjust key parameters, you can use it to apply a touch of widening to chorus sections, to give them an extra sonic lift. You could even try applying gradually more widening (while ensuring to check for mono compatibility) as the song progresses, to make each chorus feel slightly 'bigger' than the last.

Stereo Fakery

While Imager provides plenty of options to adjust the stereo image of a stereo source, it's not designed to fake a stereo sound from a mono source. Thankfully, the MonoToStereo plug-in (in Cubase Pro, Artist and Elements) has this task covered, and the audio files include an example based on a mono electric piano recording. As shown in the third screenshot's second MultiPanorama, GUI MonoToStereo appears to achieve this fakery by applying the well-established EQ-based trick of panning narrow EQ ranges of the mono recording to opposite sides to the stereo image.

You can use Imager for a particular type of mono-to-stereo fakery, though, and mono piano recordings are an obvious candidate. Stereo piano recordings usually convey the natural left/right balance of low to high notes. In a mono recording, you could simulate this effect to some extent by using the Pan controls for each Imager frequency band, panning lower frequencies (notes) to one side of the stereo image and higher frequencies to the opposite side, and use the Output controls to adjust the tonal balance. It's difficult to get a truly convincing result on an exposed part, but combining MonoToStereo and Imager does have potential if you want to fake both stereo and the left/right note pattern from a mono piano recording that will be used in a busier mix.

Mono Matters

Checking the mono compatibility of a stereo mix is always informative, but it becomes absolutely essential if you're using stereo image processing, and widening in particular. Cubase's Control Room allows you to switch between stereo and mono playback at the click of a button, but this feature is only available to Pro users. Artist and Elements users, then, need an alternative approach, and Stereo Enhancer allows just that. This plug-in includes a Mono Compatibility Check button (in the centre of the UI). If you insert an instance of this on your master bus, making sure it's placed post any stereo image processing, and leave this button switched on (it lights up blue when engaged) you can use Stereo Enhancer's main bypass button to switch between normal stereo monitoring (plug-in bypassed) or mono compatibility monitoring (plug-in engaged). Just remember to have it bypassed when bouncing your mix down to a stereo file!

With StereoEnhancer inserted on your master bus and the Mono Compatibility Check button engaged, the plug-in's bypass button lets you quickly switch between stereo and mono playback.

It's also worth pointing out that this simple workaround generates a phantom mono signal; you hear the same audio from both speakers, resulting in a 'phantom' signal that appears to sit in the centre of the stereo field. This is not quite the same as listening on a single mono speaker, where all sound emanates from the same place — and Control Room allows you to send the mono output to just one of your monitors. Workarounds are possible without Control Room, but I'll leave that discussion for another time; until then, feel good that phantom mono is better than no mono compatibility check at all!

Finally, a potential drawback of Imager is that it can add some latency. This is not usually such an issue while mixing, but it can easily be distracting in a real-time recording context if monitoring through Cubase. Thankfully, then, Steinberg have included a LIVE button (top-centre of Imager's UI), which engages a zero-latency mode. It's at the cost of some processing quality, but the results are certainly good enough if you suddenly find yourself needing to lay down a further live track or two in the middle of your mixing process.

Look East

ALINA SMITH

How To Make It In K-Pop

Successful songwriters and producers explain how to adapt your skills to the massive Far Eastern market.

I'll never forget the moment my publisher told me they were thinking of dropping me — my heart plummeting into my toes, the cool air of an autumn Nashville afternoon seeping through the office window and wrapping around my throat like a vice. Dropped? But I had only signed my deal the year before. Surely, I could have a little more time? Unfortunately, my publisher told me, my time was just about up. I had been signed for a year and a half, and if in about six months I didn't start getting 'cuts' (industry jargon for having your song recorded by major-label artists) they wouldn't be able to afford to keep me on, no matter how much they liked my songs.

It was 2015, and I had only recently moved my family to Nashville from Las Vegas, a decision largely fuelled by the fact that, after a decade of creating music for free, I now had income from my publishing deal. Needless to say, the prospect of losing it had me panicking, my palms sweating as I imagined the career I'd been working toward since I was 17 collapsing into dust. Every clock suddenly seemed to tick louder, as if reminding me of my six-month deadline.

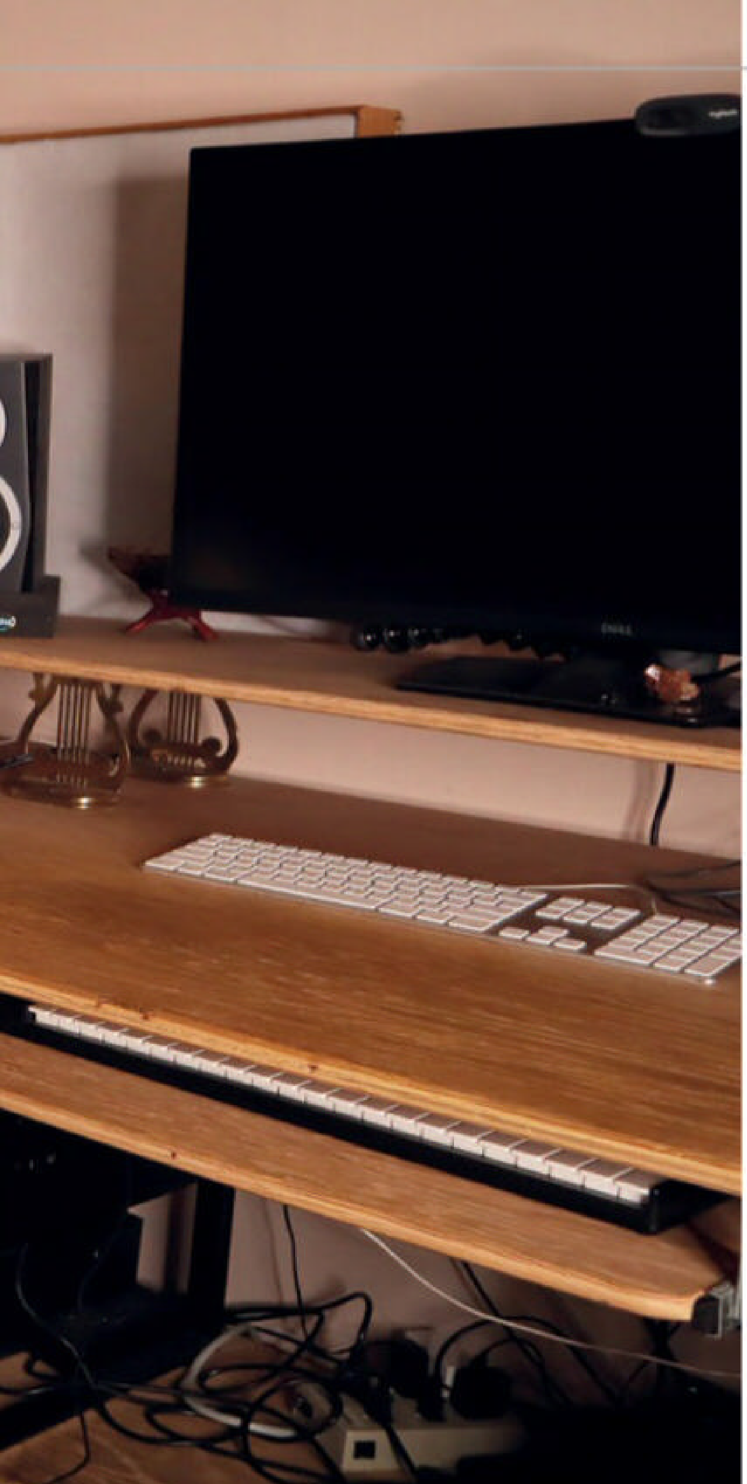
I remember calling my best friend, songwriter and artist Elli Moore, immediately after that meeting. "What am I gonna do?" I asked her in a voice like sandpaper. I didn't have to explain my predicament: every Nashville writer knows it takes years to become established in country music, sometimes even decades. And all I had was six months.

"You need a new angle," Elli told me firmly. "They said you should get cuts. But they didn't say they had to be in country." Her words sparked a distant memory. Back in 2009, I met a well-known Japanese producer, Shin Murayama, on Craigslist of all places (ah, 2009). I was barely out of my teens then and as green as they come, but Shin liked my voice and writing and sent me some tracks to topline — to write lyrics and melody to. The very first song we had worked on had gotten cut by the J-Pop artist Aisha, featuring DMC from the group Run-DMC. And although I hadn't worked on J-Pop since then, the memory felt like a lightbulb sparking in my mind.

"We should write some songs for the Asian market!" I had blurted to Elli.

Elli, 18 at the time and adventurous, agreed easily. We didn't know it then, but that little moment would completely change the trajectory of both of our careers.

Being in Nashville, neither of us had relationships in the Asian song market,



Alina Smith has worked with Far East artists such as Aisha, ITZY, Rocket Girls, and many more.

so we did what any sensible writer would do: we used Facebook to look for collaborators who did. Within that six-month window, Elli and I would get several cuts in the K- and J-pop markets with bands like Red Velvet and RAINZ, as well as several ‘holds’ — labels seriously considering our songs for their artists’ projects. I still chuckle as I recall my publishers telling me they were proud of me... even if they did have to look up what K-pop was. At the end of the six months, I was told the publishing company was picking up my option. My deal was saved.

That night, as Elli and I celebrated, we chatted about possible next steps. Now that my deal was continuing for at least another year, I could return to writing country. The thing is: I didn’t want to. We realised then that what started as a last-ditch plan to rescue me from career death had transformed into something much bigger. We had both fallen in love with the creativity of writing and producing Asian pop music. And we would follow that love to Los Angeles, transforming Elli and Alina into LYRE: the writing/production team we’re known as today.

More Is More

Coming from Nashville, where most production, at least at the demo stage, revolves around guitar and simple percussion, one of the most shocking changes I’ve experienced as I began producing Asian pop was just how much production there was. If a Western song usually has three sections — verse, chorus, and pre-chorus — which often stay within the same groove and use the same chord progression, an Asian-market song will likely have five or six, swinging between chords and grooves like an enthusiastic aerialist. It’s a lot of parts to produce.

Posing an extra challenge, there are often more production elements in these tracks than there would be in a Western-market song. On ‘Rocket Girls’ by the Mando-pop band of the same name, I recall having over 50 stems just in the drop: several layers of Serum synths, growling and sawing at the listener, multiple loops racketeering the top end, while a busy kick pattern pounded underneath it all. In the US, less is often more. In Asian music: more is more.

“There can be no awkward silences in K-pop,” songwriter Melanie Fontana Schulz shares as we chat over Zoom, confirming that this theory applies not only to the track but to the topline as well. “The average Asian music consumer likes to karaoke, to go out and sing these songs.” Melanie, who’s written hit records in the Asian market, from BTS’s ‘Boy With Luv’ to TWICE’s ‘I Can’t Stop Me’ continues: “It gets awkward if there’s a break. It’s the best piece of advice I’ve gotten about writing K-pop,” she adds with a chuckle, explaining she first heard this from an A&R. “Now I make sure to get as many hooks in there as I can. The first verse will 100 percent be different than the second. The second pre-chorus will have a layer of background vocals that wasn’t in the first pre. There might even be two different hooks. It’s all about re-combination and lots of parts.”

Melody Is King

My partner Elli and I celebrated when we first heard that the K-pop band ITZY were cutting our song ‘Mafia In The Morning’ (simply called ‘Mafia’ then). But our joy became tinged with confusion »

Melanie Fontana and her husband and collaborator, producer Lindgren Schulz.

as the band's label, JYP, began asking us for revision after revision on melody, especially on the pre-chorus. "They wanted it to be perfect," Elli laughs now — an easy feat seeing how the song has garnered over a hundred million plays in barely a month. "But it was definitely an adjustment to do so many revisions when we're used to getting only a few from the US and European labels we work with."

It turns out, our experience is quite common in the Asian market. "Melody is king," says Linnea Södahl, better known by her artist name Nea, a songwriter for TWICE and TAEYEON. "It's very important to have strong hooks. But there are also some specificities to writing for this market. For example, several A&Rs told me you shouldn't have long notes. The Korean language has a lot more syllables. So, you need more syllables to be able to say anything at all. And that's why you need to write quicker rhythms in your melody."

Another songwriter, Katrine Klith, who goes by the professional name Neya, agrees that the melody can make or break an Asian-market song. Neya and her husband Daniel Durn, who had both co-written Weekly's newest single 'After School', spoke with me over Zoom from their Copenhagen home, and they unanimously told me: "It's all about the good hooks." They might just need to be a bit more unusual than those you'd write for the US market. "You really have to think out of the box and be creative with your arrangement," Daniel clarifies.

"But it's also so fun," Nea interjects, smiling, "because you can do something that would never be acceptable in Western music. Like chromatic melodies! You can't have a K-pop song without a chromatic melody!"

Linnea Södahl: "The Korean language has a lot more syllables. So, you need more syllables to be able to say anything at all. And that's why you need to write quicker rhythms in your melody."

The three of us laugh in unison because it's so true: chromatic melodies, which are uncommon in the West, are the staple of Asian pop music.

Even very well-known writers are subject to the rules of the K-pop melody. "We had collaborated with Ryan Tedder on TWICE's 'Cry For Me'," Melanie Fontana shares. "The label was looking for a slow, sexy 'Fifty Shades Of Grey'-type song. We ended up hitting up Ryan, asking him to throw some melodies on the track we made. Once he sent us his melodies, I ended up going through them to make sure they were all 'K-poppified' for the girl group. All small tweaks, but it helped the label envision TWICE singing the record."

"As a writer, it's quite fun to write these more experimental songs," Nea confesses, over Zoom from her Stockholm kitchen. "When you write for the Asian market you don't have to overthink the lyric. You can let the melody steer the song. You get very free in your way of creating. It's fun to mix it up!"

Get Conceptual

One of the first things we'd learned about writing for the Asian market is that, although most of the English lyrics we write do get translated, one aspect matters a lot: the concept. "I don't know if 'Mafia' would have gotten cut if not for that concept," my partner Elli confesses, referring to the party game Mafia which had been the inspiration behind the song. "Korean labels want something they can build a whole vibe around, something that can be reflected in not only in the music, but the graphics, and the music video."

Melanie Fontana agrees on this point. "Korean labels have a very strong vision for every project," she tells me. "In fact, they've probably cooked up this whole world these artists will live in for this album. We'll get very detailed briefs from them, and sometimes they even include the title. When we co-wrote 'Boy With Luv', the label had the title already picked out."

"The most important thing is to have a good concept," Neya confirms, her husband Daniel nodding his agreement. "I always imagine the music video in my head when I'm writing a song. Like, could they be in space or jumping on giant strawberries? It needs to be something that can be envisioned easily."

"Although," Daniel chimes in, "sometimes a good concept is only the starting point. Labels might be inspired by it but twist it into something different. We did a song called 'Yum Yum' for the TV show *Produce 101*. The original concept was about candy, all of its different flavours. The label liked the food reference, but chose to go with a broader meaning — which is how we ended up with 'Yum Yum'."

"For us, it's important to have a good concept even if it does end up getting

Linnea Södahl, aka Nea.

changed," Neya agrees. "We care about what we write and want to present the song in the best light, to help the label visualise it as we do."

Take It To The Net

We've all heard of the old days where a piano/vocal recording on a cassette was sufficient to present a song. Those days are long gone, but in recent years I've noticed the quality of demos consistently climbing up and up until many are indistinguishable from final records. This is even more prevalent in K-pop than Western music, especially when it comes to vocal production.

"Do you remember how aggressively we used to stack our BVs?" I ask Elli.

"You made me cut four instances of every harmony," Elli reminisces, "and there were always, like, four-five parts."

Although we've since cut down the number of harmonies we record — two per part instead of four — the idea remains the same: if you're presenting a song to an Asian pop group, you need to have a lot of parts to represent every singer in the band.

Most K-pop writers abide by the same concept. "We take it to the net here," Melanie shares proudly. "Lindgren and I put everything on the track as if it was a final: backing vocals, even ad-libs. We've noticed labels love ad-lib suggestions, so I always do full takes of those." She demonstrates by effortlessly singing a very high note. "I sing the whole song as if I'm the artist."

"We can easily use 10 hours doing vocals for a K-pop song because of the many parts," Neya confirms. "It's »

super-important to present it as close to final-quality as possible."

Lyrics Matter

When we write for the US, we might spend two hours just trying to dial in the lyrics on the second verse: the story development, the clever metaphors, the cerebral wordplay. We know that once the song we're creating is out, the lyric will be in its original form, so we want to make sure it's near perfect. In the Asian market, almost everything gets translated, but the English lyric still matters in one way: its phonetics. I will never forget how surprised Elli and I felt when we first heard the Korean version of our song 'Talk To Me', which had been recorded by the group Red Velvet.

"In the pre-chorus," Elli recalls, "we had this repetitive lyric: 'keep it real, keep it real, keep it real.' When we heard the Korean version, it sounded very similar: 'gidarin, gidarin, gidarin' [*'I waited' in Korean*]. The phonetics of our English lyric inspired the Korean writers' choice of lyric."

"I always give lyrics good phonetics because it enhances the lyric," Melanie agrees. "Melodies just sound better on good lyrics." Although, according to Melanie, you can't always guess what will be kept and what will be translated. "Often," she shares, "the lyrics you think they won't keep, they keep. And vice versa." Sometimes, even the song's title is on the chopping block. "With BTS's 'ON', the original title was 'The Elements', but the label changed it. You never know!"

Nea experienced something very similar with her song 'Hell In Heaven', which had been cut by TWICE. "The song was called 'Paradise'," she recalls, "but they ended up calling it 'Hell In Heaven', which is not even a lyric that was originally in the song. Funny how that works sometimes." Although the label did keep some of her original lyrics: "When I was writing, I added some little keywords in there, just for some ear candy. A lot of those are still in the song."

A Fair Fight

Even years later, I still feel amazed at the way Elli and I were able to dive into the Asian music world without much resistance. We were completely unknown as writers, and we lived in Nashville — not exactly the capital of K-pop. And yet, we were able to get a cut on a K-pop album that went to number one on Billboard's World Albums

Elli Moore (left) and Alina Smith are production duo LYRE.

Producer Daniel Durn.

Katrine Klinth, aka Neya.

chart within the first few months of writing for that market, simply because it was a good song.

"Something like this is unheard of in the Western market," Elli says, shaking her head as we reminisce. "In America, your clout always comes before the quality of your work."

"But in Asia," Melanie says, chiming in on the topic, "it's a fair fight. You can be at your laptop on GarageBand in a town no one's ever heard of, and you can be 15. If a K-pop group gets a hold of a song of yours that they think is great and is better than a well-known writer's song, they don't have a problem telling a 'large' writer that they have something that meets their vision more head-on from a lesser-known person. They simply don't care about clout. It's all about meeting their vision."

Melanie then goes on to explain that she's always admired the professionalism of the Korean A&Rs she's worked with. "They're usually very honest," she explains. "There's never been a time I've been told by a K-pop A&R that I have the single, and then it didn't happen. If they say it's coming out in three weeks, it is. There's no hype. Just be the best songwriter, write the best song, and you will do well."

It's a far cry from the US market, where it often feels like you're playing the lottery when you're writing and producing for pitch — which was exactly the way I felt when I was trying to get country artists to cut my songs back in Nashville. It had shattered my world at the time, but looking back now, I'm so glad my publisher had given me that ultimatum all of those years ago. I would have never been able to write songs with my best friend with no 'big' writers involved and get them cut.

Stretch Musically

Over the years, I've felt beyond grateful for all the career opportunities LYRE has garnered through writing Asian music. But my love of K, J, and Mandopop goes so much deeper.

"You love the challenge," Elli smiles, calling me out. "Being able to do something totally left-of-field and seeing how it turns out."

She knows me well. If I were solely focused on Western music, I doubt I'd find myself mixing jazz with EDM, stacking up six parts of harmonies, or switching tempos mid-song as I often do with K-pop. I wouldn't be forcing myself to study complex chord structures, chipping away at my music-school-dropout solfeggio skills — stretching and growing as a writer and producer.

I had hoped writing Asian music would save my career — and it did. But I would have never imagined that it would save my love of music as well.

DURAN DURAN

JOE MATERA

Nick Rhodes: Making *Future Past*

For their 15th album, Duran Duran collaborated with three masters of electronic music and a dog-fixated artificial intelligence.

From their early ground-breaking music videos to being one of the first acts to sell singles digitally, Duran Duran have always embraced the latest technology. The band's latest album *Future Past*, their 15th to date, is no exception, with production split between Giorgio Moroder, Mark Ronson and Erol Alkan.

History & Energy

The group first began working on the new album in late 2018. Originally planned as an EP, it evolved into a full album before the process was halted when the worldwide pandemic hit in early 2020. Rather than continuing to work remotely, Duran Duran decided to put the project on ice.

"We work better when we're mostly together," says keyboard player Nick Rhodes. "I sometimes work independently because I have a lot more work to do on detail and there are more layers of synths, but when we're writing and forming the shape of the songs, we will always work together. It's only afterwards that we can get out the smaller brushes. Like Simon [*Le Bon*] will go in on his own with the engineer and producer and do all his vocals and figure out his harmonies and things like that.

"The main producer on the album was Erol Alkan, who we've never worked with before. He comes from a sort of indie and dance background and is a great DJ with a phenomenal knowledge of the history of music. I think his brain can only be rivalled by Mark Ronson's. Between the two of them, you can name a track as obscure as you like, and they will know it. I find that quite inspirational. More than anything, I think Erol brought energy to the project, which was infectious, and it really helped particularly with the bass and drums."

In The Dark

Once the band were able to regroup after lockdown ended, they returned to Assault & Battery Studios in Willesden Green to resume working on the album. "We did the bulk of the album there," says Rhodes. "The

For the recording of *Future Past*, the entire band set up in the control room at Assault & Battery, with Roger Taylor playing a Roland electronic drum kit.

studio had loads of analogue gear and a big analogue desk, a heavily modified Cadac G-series, which we used for the drums and things like that. There was also a giant Roland System 700 modular synth sitting in the room, which we used on a few tracks in order to run sequencers with which we started writing.

"We set up in the control room with Roger [*Taylor*] playing a Roland TD-30KV V-Drum kit. And we had sequencers running from the Roland System 700 while I had four different keyboards to choose from. We had another keyboard on the other side of the room, too, in case somebody else wanted to play. And we usually created an atmosphere by having the studio as dark as possible when we were writing, with most of the light coming from the computer screens. It was a great room to work in at the studio.

"Because we usually build the palazzo from the foundations up, we've got to know what the bass and drums are doing at all times to be able to work around everything else. We often re-record the bass and drums later if we want to tweak a little bit of something, but we don't change the rhythm much. If we do change the rhythm, we will then have to rebuild the whole track. John [*Taylor, bassist*] and Roger are a very solid foundation, and I think their work on this album is the best they've done in a long time."

Perfect Timing

The band worked with legendary producer Giorgio Moroder on two tracks, 'Tonight United' and 'Beautiful Lies'. "Giorgio was everything we love in songs: sounds, melodies, arrangements and compacting things," explains Rhodes. "Those two tracks are very Euro-dance, with high-energy pulsing synthesizers and sequencers, synth hooks and super-hooky chorus lines and melodies. Giorgio is one of Duran Duran's absolute superheroes and we'd been wanting to work with him since we first formed the band. When I was a teenager DJ'ing at the Rum Runner Club in Birmingham, I used to play a load of Giorgio tracks. I don't know how we never managed to work with him over the past four decades, so to finally converge was great. We've had this vision in our heads of what it would sound like — the perfect blend of the Giorgio Moroder and Duran Duran sound — and for once it went predictably well. Everything we did just fell into place and sounded just as we envisaged it. He was the consummate professional, too. He'd turn up every day on time with his briefcase, get out his little keyboard and put it on the desk along with his computer and get to work."

On another song, 'More Joy!' the group collaborated with a Japanese girl punk group called CHAI. With borders closed, »

Duran Duran pose in the live room at Assault & Battery with collaborators Graham Coxon (far left) and Erol Alkan (second from right).

this was largely done remotely, though lead vocalist Simon Le Bon did journey to Japan to record the track. "Simon engineered that one, as he spends some time in Japan where he has a small studio called SYN," affirms Rhodes. "So that track was done remotely because they were in Japan doing the vocals with Simon.

"The track itself, though, came about out from a jam we wrote together with Graham Coxon. I think the guitar on that track is so spectacular and it was done completely live; there are no guitar overdubs on it at all. Sometimes when we were jamming on it, I couldn't tell at certain points when I was putting things through effects or when Graham was putting things through effects. I didn't know who was playing what. We loved working with Graham, as he is truly an inventive musician and a great energy in the room."

Jupiter Rising

Despite Duran Duran's emphasis on new technology, Nick Rhodes' own musical contributions drew mainly on his collection of analogue synthesizers. "I barely use much that's digital aside from the samples, particularly the orchestral samples. I do use some samples from Arturia, which I think are the best samples of analogue synthesizers out there. They've really worked hard to get

the things as accurate as possible. They're never going to be the same as analogue, though, because analogue sounds behave differently and synthesizers behave differently too, and they'll be the first people to say that too. But if you want a sort of a low Moog bass or a Fairlight, which also sounds very close because that was digital anyway and only 8-bit at the beginning, they're it.

"I've used the same synths for many years, and I find myself constantly returning to my Roland synths: particularly the Jupiter 8, and sometimes the Jupiter 4. I have been also using the Roland Fantom; though it's digital, I'm astounded by the quality of the reverb in it. It's almost up there with the standard of a Lexicon. It's extraordinary as I've never had a synth that has had that. I also use a Moog Voyager. I think there is no better bottom end sound than a Moog. I also used an Elka Synthex, a Prophet V and a lot of older analogue beasts."

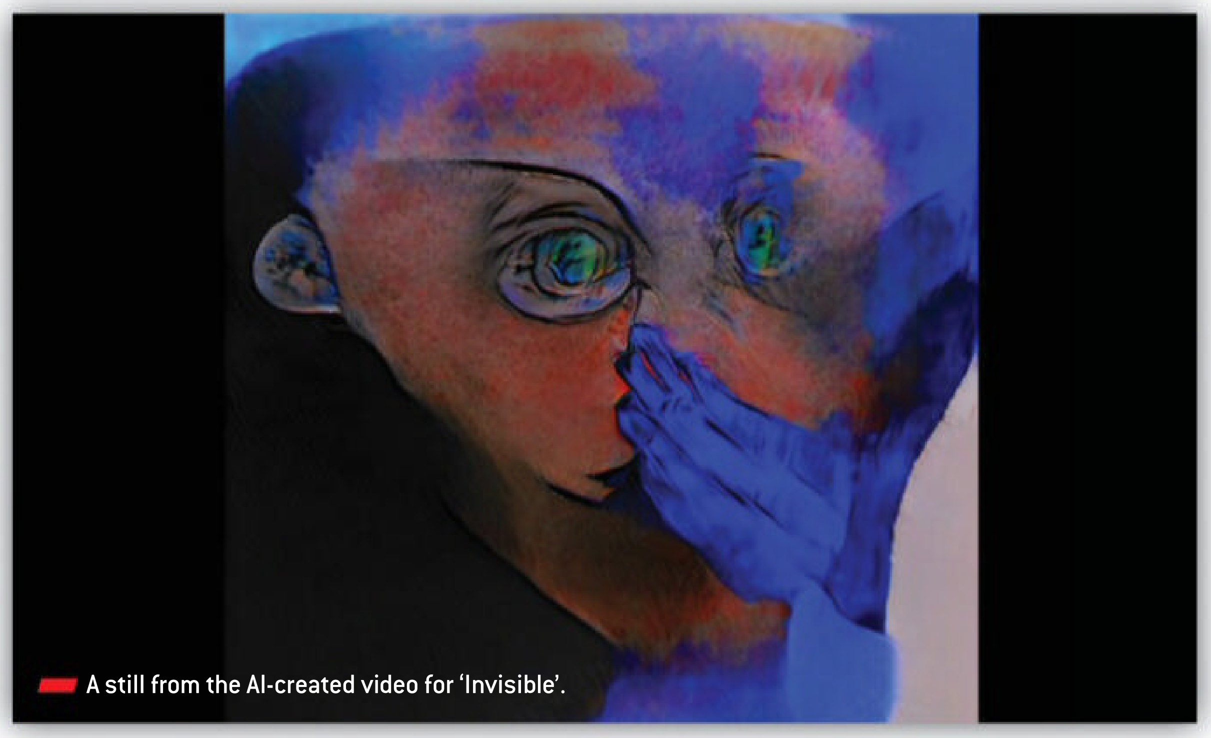
Other Beings

Cutting-edge technology is apparent in the video for the album's lead single, 'Invisible', the first to make use of an artificial intelligence called Huxley.

"Huxley, our AI, is the most extraordinary machine," exclaims Rhodes. "I think people are sometimes a little frightened of AI and what it's going to do, and understandably so. As many of the great thinkers of the world have said, we must never let AIs become weaponised, because the leap in technology

over the last decade has been phenomenal. With Huxley, what was interesting in making the video for 'Invisible' is that we fed in the song, photos of us, videos of us and, very specifically, videos of us mouthing the words to the song. Also, lyrics and references of what the song was about, along with some art references too, they all went in as well. After a while it learns like it is a child. Then the AI mapped all the measurements of our faces and took a load of photos and images and made its version of what it wanted it to look like. The blacked-out faces was a shock! We didn't tell it to black out our eyes, that's just how it came back. I guess putting multiple layers of different images must have darkened the eyes.

"There were some strange things that happened along the way. At one point it became completely obsessed with dogs, and we didn't know why. For about three days all it was doing was making dogs and we thought 'This is going to go wrong!' as there was no way to control it. So they then had to invent a director who we called Hitchcock to go in and give it a rule and try to persuade it by asking, 'Can you make different types of animals?' Then it started making all these different types of animals and strange creatures. It was a very fascinating experience working with another type of being, because it was much more than a computer. It was programmed to follow the same processes as the human brain. It's based on the part of the brain that creates and dreams, rather than just the cognitive part that is logic and fact. The people that worked on it are all students of a prominent neuroscientist, Karl Friston."



A still from the AI-created video for 'Invisible'.

Change & Convenience

Through their 40-year career, Duran Duran have successfully adapted to many such

changes and developments in music technology. Looking back, Nick Rhodes pinpoints several that he believes were turning points. "There's been a few of them that were game-changers, such as sampling, for one. If you listen to an album like *Fear Of A Black Planet* by Public Enemy, you can hear what they were doing then with samples, making something really remarkable with them. And this was before anyone knew you had to credit people and that you weren't supposed to just to take people's things, which I'm sure they did very innocently.

"The Fairlight CMI, an Australian invention, was very significant in the beginning of proper sampling and digital synthesis. There weren't many of us that had one at the time. I remember it was a very elite club as it was very expensive, but we had one along with Kate Bush, Peter Gabriel, Trevor Horn, Art Of Noise. The Akai samplers were another. Those Akai samplers made a real big difference. And the Linn Drum along with the Roland TR808 really revolutionised the sound of music through the early '80s. We used drum machines from the very first album.

"I miss analogue tape, because I like the compression and I like lots of things about it. But there's a convenience to being able to do things with a computer now and the use of digital effects and how you can mix and keep everything. I remember the very first automation for mixing desks. When we did our first album, *Duran Duran* [*1981*] there was me, producer Colin Thurston, the tape operator and whoever else was in the room, and there were like eight hands on the desk. My job might have been to turn on the echo at a certain point on the vocal, and I had to catch it in real time while somebody else had to ride up the string sound at the end of the song and somebody else had to mute the drums at a certain point. There was no automation. It was actually really exciting doing mixes that way, but it was a real drag when you got them wrong, because with some of them you had to do them 20 times to get them right.

"We take it for granted now with Pro Tools, and I also think people take it for granted when they cut things up, because it's so easy to just sing a chorus and then say, 'Right, now we've got it.' We've tried very hard to never do that, as we try very hard to actually make it more of an experience and a performance. Simon, in particular, is not a fan at all of that cutting and pasting. He always likes to do all his vocals and the backing vocals for each chorus, and you will very rarely hear him say, 'OK, you can take that and put it in the other place.'

"But where would dance music be without cut and paste? When we used to make 12-inch versions of our very early songs, the night versions as they were called at the time, with songs like 'Planet Earth' and 'Girls On Film', we used to play them from start to finish. We would arrange the song and then go, 'OK, we need another 24 bars there, so you do this with the drums and I'll play this,' and we'd literally rehearse an arrangement and then play it for nine minutes or so to a click to keep everything tight. There wasn't a way to cut things up. Editing with tape was such a laborious process it just didn't make sense to make 12-inches like that. The first 12-inch that we made by cutting tape was 'Is There Something I Should Know', which I did with producer Alex Sadkin."


For two songs on *Future Past*, Duran Duran collaborated with legendary producer Giorgio Moroder.

Immersive Mixing

Rhodes believes the key question about any new technology is whether it will enhance the music or whether it's just a gimmick. The group were faced with this decision when it came to mixing the new album, which is being issued both in stereo and in Sony's new Reality Audio 360 immersive format. "I played around with it in the studio with Josh [*Blair, engineer*] and it's really good for headphones," states Rhodes. "I would call it more like an enhanced stereo for around the head, as it's not so much literally 360. You can put things in the middle in

Astronomia

When the pandemic halted proceedings on *Future Past*, Nick Rhodes embarked on a side project. He teamed up remotely with British artist and singer/violinist Wendy Bevan, he in London and she in Los Angeles, and together they completed a four-album, all-instrumental suite.

"When we closed down the Duran Duran project at the start of last year, I went into the studio on my own and started putting together some musical ideas which I sent to Wendy in LA. We decided we'd make a few instrumentals and perhaps put them out so we could introduce her and my work together, and then hopefully get to do an album a few months later. Things began developing as we were sending each other files back and forth. I'd open them up in the morning and find that she had sent me these wonderful soundscapes with violins and ethereal voices, lots of reverb and effects, and it was quite inspirational because it had no real structure to it. It was truly freeform. I'd been used to working in the confines of writing songs where you need a verse and a chorus and certain structure. And this had none of it, and no words either.

"So I rose to the challenge by sending her my pieces, which were a little more structured, as I wanted to see what she would do to them. After we had about eight of them done, they were sounding so beautiful that we decided to carry on. And we roughly based the theme of them on the Universe — which has been plundered a thousand times by every film-maker, musician and artist — and the turbulence of the past year, and blended that in with some mythology. By the time we had made an album, we could see that there was no end coming to the pandemic, so we just carried on. We decided to make 52 tracks and put 13 tracks on each of four albums that would go out on equinoxes and solstices and have all four released this year, and

then we'd make a box set for vinyl. It was fun for me to have all these sounds and melodies and sound effects to work with and with orchestra too, because we've got a lot more orchestral sounds on these recordings. In the studio I've been using Spitfire Audio and EastWest sounds and the sounds are truly sensational and the samples are really blissful."

front of your face and move them over to the back of the head, and you can swirl things around the head and you really do feel it, which you wouldn't think with just stereo in headphones that you would get that sensation.

"We wanted to do the album in that way because it's a different way to listen to things. You will hear things in a way you wouldn't if you just listened to it on regular headphones. Because the mixes are basically made from stems, you can pick something and you can, for example, make the snare drum move across diagonally from behind your head to a corner over in front of your head. We had so much fun doing that."

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The analogue I/O includes a pair of premium microphone inputs, two high-impedance instrument jacks, two stereo headphone outputs, a stereo monitor output and a second stereo line out pair. Plenty for recording a small ensemble, then, but with some serious expandability available thanks to its stereo S/PDIF I/O and multichannel ADAT input. All told, there are 14 inputs and 10 outputs on offer, making the Zen Q Synergy Core suitable for almost any recording scenario.

But it's on the inside where things get really interesting. As well as boasting Antelope Audio's acclaimed conversion and AFC (Acoustically Focused Clocking) technology, the Zen Q Synergy Core features Antelope's unique combination of DSP and FPGA chips. This proprietary Synergy Core platform is able to host a huge collection of Antelope Audio Real Time FX, with zero latency and at no cost to your CPU usage. As standard, the Zen Q Synergy Core ships with a library of 37 bundled effects and processors, which includes analogue-modelled compressors, preamps and EQs, guitar amp and cab emulations, plus Antelope's signature AuraVerb reverb effect. As part of this generous competition prize bundle, however, the winner will additionally get licences for every single Real Time FX processor in the Antelope Audio collection — that's over 50 Real Time FX, including some of the biggest names in signal processing! To name just a few examples, this bundle includes Auto-Tune Synergy, based on Antares' world-leading vocal tuning technology; a model of the BAE 1073 microphone preamp; emulations of some of the most revered British and American FET, VCA and optical compressors; and accurate recreations of classic console EQs. All told, the additional Real Time FX are worth over $2000!

To be in with a chance of winning this fantastic prize, all you have to do is visit the URL shown, and answer the questions there, by Friday 5th November 2021. Good luck!

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How I Got THAT SOUND

Martin Landquist

A-ha 'Lifelines'

JOE MATERA

Martin Landquist is a Swedish producer, remixer, songwriter and composer of film music. His credits include A-ha, the Cardigans, Kent, Meja and many others. Martin has been a BMG songwriter since 2014, and is currently working on the fourth album by his electronic project Nåid, due to be released in 2022. Here Landquist reveals how he got his favourite sound on A-ha's 'Lifelines', the 2002 title track to the Norwegian synth-popsters' seventh studio album.

"On the upbeat of the intro of 'Lifelines' you will hear an Ebow sound that runs through each chorus. I have always felt that this line really captures the melancholy of the song. The sound was not revolutionary, but its ability to survive to the end of a very long and complex recording process was simply astonishing!"

Lonely Voice

"I was looking for a part that could melodically interact with the string arrangement without taking too much space. I tried different instruments but the Ebow found its natural place in the mix. It also had a solitary feel that went so well with the sentiment of the lyrics. I played and recorded the guitar line in my studio in Stockholm while working on A-ha's original demos. I used a Marshall JMP1 and lined it straight into the computer. It sounded very bland and I was trying to make it sound more synthesized without it taking too much space. A-ha has always had an element of electronic in their sound, and I was looking for that effect without losing any of the feel of the guitar line.

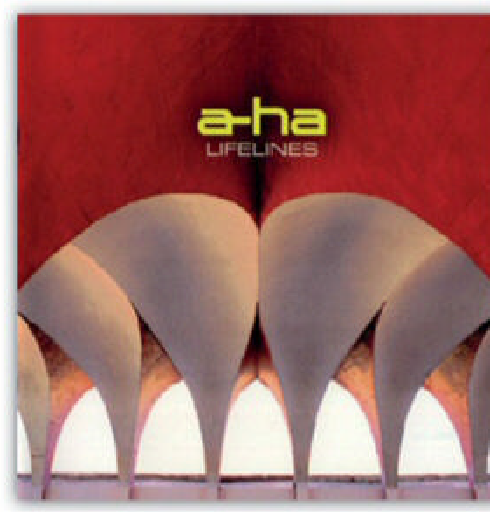
"We ended up feeding the guitar track into a Roland RE-201 tape delay. We hardly put any effects on. We just slightly distorted the sound using the inputs of the delay. That signal was then fed into my [*EMS*] Synthi AKS input. The filters of the Synthi were amazing and we found a slight modulated setting that worked perfectly with the line. The Synthi has a built-in reverb and two little speakers. I added a little reverb and then miked the Synthi up using an AKG 414 Microphone. Finally, this signal went through a Calrec PQ1740 [*channel strip*] and then back into the DAW. It was quite a journey! And the journey continued for months, travelling to multiple studios across three different countries: Sweden, Norway and England. Musical parts were dropped and added en masse but this guitar line stayed in there until the end.

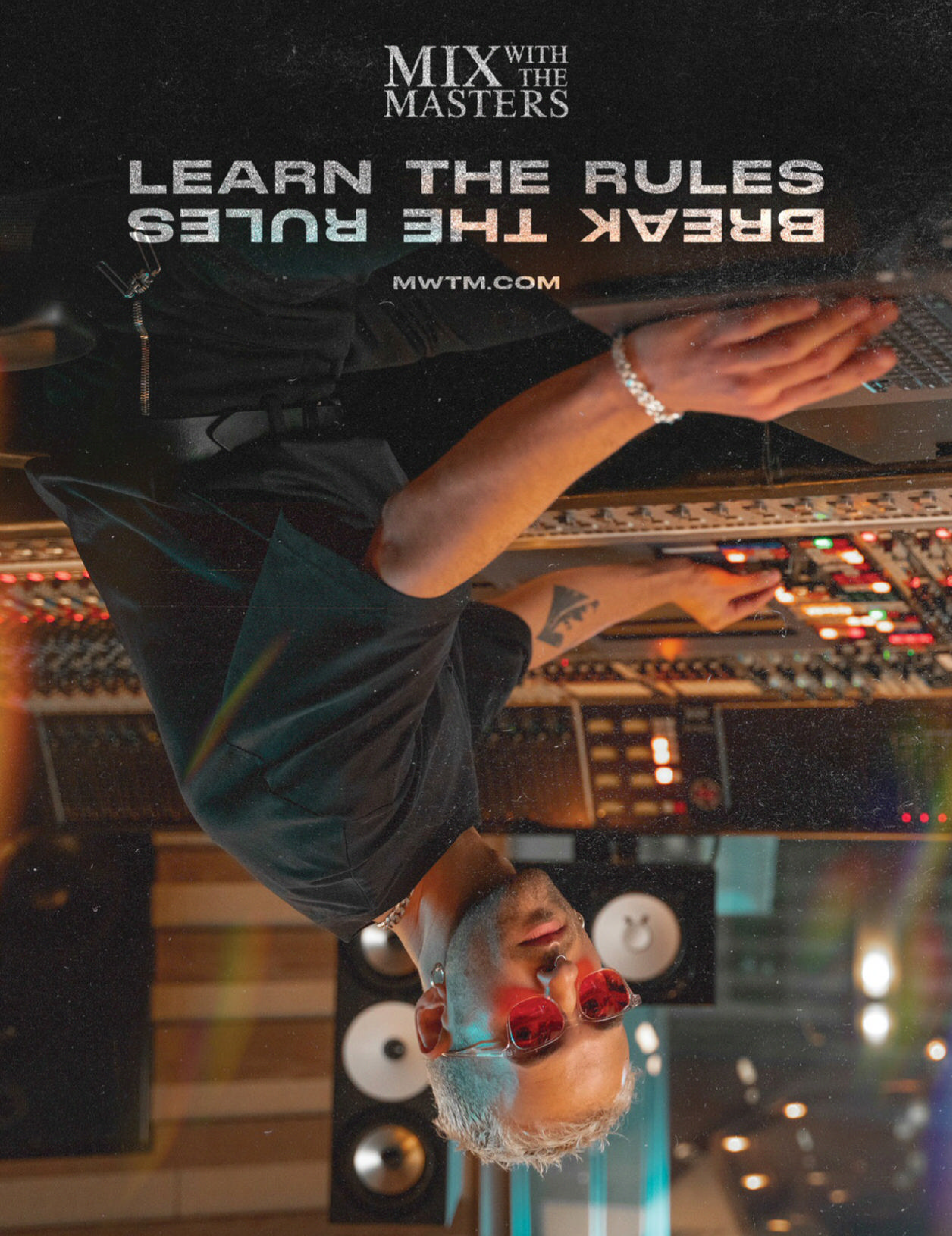
"It was a complicated process recording the album. The members worked more or less separately with different producers. Songs were assigned to the respective producers, so we all worked simultaneously in different studios. To do the vocals we flew to Oslo and brought the files with us from my studio. So it was quite a challenge to mix the album since the songs came from many different sources. But Michael Brauer and his team did an amazing job mixing the album and making it sound whole. And still today I am amazed and happy that this little guitar made it all the way!"

Hear The Sound

W https://open.spotify.com/track/3kPuHm2aXRJvpoKOCtjYI0?si=2bbd368452854511

W www.youtube.com/watch?v=DCkbfyk6XGc

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INSIDE TRACK

SECRETS OF THE MIX ENGINEERS

Alessandro Marcantoni

Italian band Måneskin were unlikely winners of the 2021 Eurovision Song Contest, but even that couldn't prepare them for what came next. Mix engineer Alessandro Marcantoni tells the story of their viral single 'Beggin''.

PAUL TINGEN

Måneskin broke through this year when the glam-rock quartet surprised everybody by winning the Eurovision Song Contest with a rap/rock mashup called 'Zitti E Buoni'. The song immediately went to number one on Spotify, and to the top of several hit parades in Europe. A second single released a few months later, 'I Wanna Be Your Slave', was also successful all over Europe, and was further propelled upwards by a later duet version with Iggy Pop.

So far, so normal — but then something strange happened, and it perfectly illustrates the random weirdness of the Internet, especially TikTok. A song that Måneskin recorded in a few hours in 2017, at the very start of their career, went viral on the social video sharing platform, thanks to a number of glow-up and karaoke challenges.

'Beggin'' promptly went to number one on the Spotify global chart, and started to climb the charts everywhere in Europe. Eventually it went to number one in nearly a dozen countries, reached number six in the UK and became the band's first entry in the US charts.

By the middle of the summer, 'Beggin'' had become the band's biggest worldwide hit. From interviews it was clear that the band had mixed feelings about a cover song they recorded hastily at the start of their career superseding their self-written and more expensively produced recent releases, and they refused to promote it. Yet at the time of writing, 'Beggin'' was still flying high around the world.

The X Files

Let's look at what happened in 2017. Speaking from Metropolis Studios in Milan, engineer and mixer Alessandro Marcantoni has the full story. He spent

Alessandro Marcantoni at the current incarnation of Metropolis Studio, Milan.

only two days recording four tracks by an up-and-coming teenage band. According to Marcantoni, it all started with the band performing at *X Factor Italia* that year, eventually coming in second.

"They performed two originals and several covers on *X Factor*, and Sony Music, which is a partner of the *X Factor*, each year selects some of the artists that appear to record an EP. Måneskin had already recorded their two original songs elsewhere, and then came to Metropolis to record four more covers, all of which appeared on their *Chosen* EP. Here at Metropolis we have recorded acts that appear on the *X Factor* for the first 13 seasons."

The four covers that Måneskin had selected to record were songs by Black Eyed Peas ('Let's Get It Started'), the Killers ('Somebody Told Me'), Ed Sheeran ('You Need Me, I Don't Need You'), and the Four Seasons ('Beggin'). A fifth cover that appears on the *Chosen* EP, 'Vengo Dalla Luna', by Italian rapper Caparezza, was recorded and mixed by Marcantoni at Metropolis a month earlier.

'Beggin' was written by Bob Gaudio and Peggy Farina and became a minor hit in 1967 for the Four Seasons. It was remixed in 2007 by French DJ Pilooski, and his version reached number one on the UK Dance chart. That same year, Norwegian hip-hop duo Madcap released a new version of the song, with added rap sections. It became a big hit all over the world, reaching number five in the UK and number one in a number of other European countries.

Exactly what prompted Måneskin to cover 'Beggin' is not known, but theirs is a high-energy, uptempo funk-rock version starting with an a cappella by singer Damiano David. His huge, raspy voice is the most defining characteristic of the song, while the musical arrangement is basic: just drums, bass, and a single guitar. According to Marcantoni, it was the same with the other four songs.

"They had just performed these cover songs at the *X Factor*, and they wanted to record them just like

»

'Beggin'
Written by Bob Gaudio
& Peggy Farina
Produced by Luccio Fabbri

Metropolis Studio in 2017, where Måneskin recorded 'Beggin'.

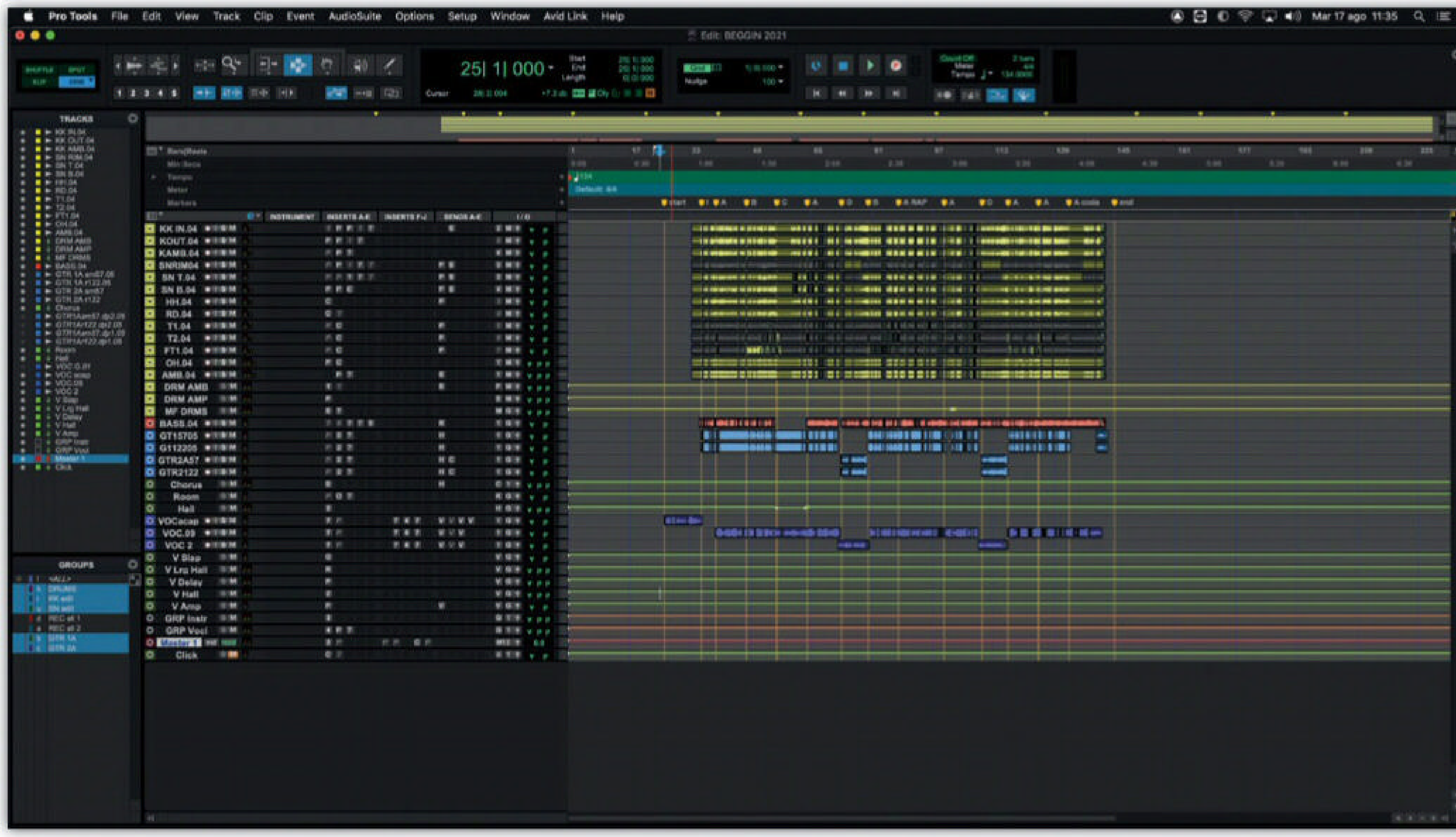
their performances, so very simple. I tried to do some production by getting them to add some instrumental overdubs, but they were not into that. They performed all songs live in the studio, and later overdubbed the vocal. While they all played together, we focused on one instrument during each take. So during these band takes we first tried to get the drums, and then the bass, and then the guitar. Damiano sang the vocals by himself, later on."

Metropolis

Before elaborating on what went on at Metropolis in 2017, we need to go back further in time to set the scene with regards to the studio, and Marcantoni. Metropolis (no relation to the London studio of the same name) first opened its doors in 1990, with three recording spaces, and quickly became one of Italy's leading studios. The studio has moved twice since then, each time reducing the amount of rooms as the demand for recording studios declined.

Metropolis' second location had two studios, with a Yamaha DM2000 desk and Genelec and Dynaudio monitors in Studio 1. It was here that Marcantoni worked with Måneskin. Metropolis has since moved to a third location, with only one studio. The operation is now fully in the box, with two Avid S1 controllers, a Dangerous Music Monitor ST+SR, monitoring from ATC SCM100ASL Pro and Amphion One18, plus an Amphion Amp700, and large collections of outboard, microphones and musical instruments.

Alessandro Marcantoni joined Metropolis in 2003 as an assistant



The 'Beggin' Pro Tools session contains just 21 audio tracks.

engineer, and quickly went up the ranks to become senior engineer. He'd played several instruments as a child, and performed with various indie rock bands as a teenager. He then studied Drama, Arts & Music at the University of Bologna, and did a Sound Engineer course at the SAE Institute in Milan. Marcantoni still works for Metropolis, but is also active as a freelance engineer in other studios, and as a front-of-house engineer in large venues in Europe. He's also occasionally involved as an educator with IULM University and the SAE Institute, both in Milan.

"I know most of Italy's top engineers," says Marcantoni, "and have learned from them. But I am mostly influenced by American and British engineers and mixers. I work with all kinds of music, and I like a lot of different music, but my roots are in growing up with rock, in all its variations. But of course, I do my best to remain up to date with the latest sounds in pop and hip-hop.

"In general I'll work with the song and the tracks that I have, and make everything sound as good as possible, without thinking too much about genre or time periods. With Måneskin there clearly were '60s and '70s influences, and at the same time I wanted to make their recordings sound current. They are often linked to the sound of Franz Ferdinand, for example."

Quick On The DAW

Back in 2017, when Marcantoni spent one day in the then Metropolis Studio 1 with a promising teenage band, it was a routine session for him. He and the band had a brief to record four cover songs, which he mixed the following day. When asked how long the entire project took, he burst out laughing: "With the *X Factor* I have very few days for each project! But each working day was definitely more than eight hours, so we probably spent two or three hours recording each song, and I would have spent up to two to three hours on each mix. The first song took longer to record, and to mix, as I was setting up the sounds. After that, because the instrumentation and recording chains did not change, things went a lot quicker. But if I had felt that I had needed another day to get the result we wanted, I would have reorganised my schedule to free an extra day.

"I don't tend to work with templates. I create each session from scratch, whether it's a mix session or a recording session. However, if I record the drum set in our studio, I'll start with the settings I have for that from a previous session, just to speed things up. I may have done a session with another drummer the day before, so when Maneskin came in, Ethan [*Torchio*] spent some time tuning our drums, and we worked on getting the sound he wanted. But whatever template I have is purely there to speed up my input list, and to have my cue list for headphones ready. It's not for the sound."

Working from memory, old gear lists and photos, and the track names in his mix session for 'Beggin', Marcantoni retraces his steps, both for the recordings and the mixing. "For the kick I would have used a Shure Beta 52A on the inside, a Neumann U47 [*FET*] on the outside, and a Royer Labs R-122 placed close to the kick drum pointing to the snare, for a mono drum kit track. On the snare I used a Shure SM57 at the top, and a Shure Beta 57 at the bottom. I duplicated the top snare track to create another effect setting for the rimshots.

"In addition I had one Neumann KM-184 for hi-hat and another KM-184 for the ride cymbal, a couple of AKG C414 XLII mics as overheads, and two Neumann U87s for ambience. The toms had Beyerdynamic Opus 87 mics. I would have avoided the Yamaha desk for the recording chains, so the mics would have gone through external mic pres. I most likely would have used API 512C's for »

Vocal effects included the UAD Galaxy Tape Echo, Lexicon 224 reverb, and a send to a bus containing Avid's SansAmp plug-in.

the kick and the snare, and Focusrite ISA 828/430 for the other drum tracks. They would have gone into the Apogee Symphony MkI, and the Avid HD I/O.

"I recorded the bass DI, using the Manley Voxbox channel strip, which I still often use for DI. That sounds great because it adds a tube sound at the recording stage. I will have used an Empirical Labs Distressor while recording the bass. For the guitar, I used a Shure SM57 and a Royer R-122 on one cabinet, which I think was our Fender Reverb Pro. The two guitar mics will have gone through the API 512C, and I do not use compression while recording electric guitars. The vocal mic was a Brauner VM1, recorded through an API 512C. I think I used another Distressor to keep the spikes under control."

Post-production

Marcantoni comped the tracks as he was recording them. "I knew I would not see the guys again the next few days, as they were busy with the talent show, so I made sure we did the comps while they were in the room, so they could have their input. The final drum track was comped together from three different takes. The drummer had played with a click in his headphones, but I did some minimal time adjustments where they had started to drift.

"We spent a bit of extra time doing additional vocal takes to fine-tune Damiano's English pronunciation. For the songs in Italian we did maybe only three takes, but in this case there were nine takes. The extra takes were not complete takes, but purely some sections. I didn't use any Auto-Tune or Melodyne, but did some minor pitch adjustments using Pro Tools' Elastic Audio — maybe some words or parts of words — and I may have done some minor timing adjustment. But Damian is exceptionally talented and it's very easy for him to sing in tune, so there wasn't much to correct."

Marcantoni also began rough mixing during the recording: "I try to get the sounds right from the moment I record them. I then work on instruments or sections of instruments during the recordings, trying to get them to sound pretty good. At that point it's easy to incorporate feedback from the artist or band. I'll do some more mixing by the end of the day, and I normally end up with something that sounds pretty refined. I may get the sounds to 80 percent of the final mix."

The Mix

Marcantoni went into his room at Metropolis the next day to get his mixes to 100 percent. Clearly there was some polishing going on, but as he had to complete four mixes in one day, Marcantoni's polishing was minimal compared to what's done in an average mix today, which tends to take a day per song, and sometimes far longer.

The mix session for 'Beggin' is indeed exceptionally simple, with the audio consisting of 13 drum tracks, one bass track, four guitar tracks, and three vocal tracks. Even this grand total of 21 tracks was reached only by splitting five of the original 16 audio tracks; Marcantoni wanted to separate some parts onto different tracks to be able to treat sections differently. He did this with the rimshot snare, guitar and vocals.

Marcantoni added a number of aux effects tracks, namely three for the drums, three for the bass and guitars, and five for the vocals. All instruments go to an Instrument Group aux and the vocals to a Vocal Group aux. Finally there's a master track, and the click, leading to a full total of 36 tracks.

"For me balancing levels is the main thing in a mix," Marcantoni elaborates. "But I of course also work on the sounds, and any sonic problems that may come up when I'm mixing. There are many greyed-out plug-ins in the inserts, for example the Avid Pro Compressor, Pro Expander, Channel Strip, and EQ3 7-band.

I used these for rough mixes during the recordings with no latency, and I then disactivated them and added new plug-ins during the mix. With the drums, many instances of the Pro Compressor and Channel Strip remained on the audio tracks, and I often used the EQ3 7-band for basic EQ.

"The main sonic treatment on the drums came from the aux effects tracks: the Drums Ambient, which had a UAD EMT140 for some plate reverb with a small decay and some predelay, and the Drums Amp, with the SansAmp for parallel distortion. All drums go through the MF Drums group track, which has the UAD SSL G-channel and Avid EQ3 7-band, for some compression and minor EQ adjustments.

"I kept the bass in place with the UAD 1176LN E, and it also has a send to the Room aux, which has a UAD Ocean Way room reverb. I added compression to the guitar tracks with the UAD LA-2A S, and there are sends to the Hall aux, with a UAD Lexicon 224, on the two main guitar tracks. The two guitar tracks that I pulled out for the B sections have sends to the Hall and to the Chorus aux, which has a UAD Studio D Stereo Chorus."

Vocals

"All vocal tracks have the same inserts: Avid EQ3 7-band, UAD 1176 LN, Mäag EQ4, and again the 7-band. The main difference is in the plug-in settings and in the sends. The a cappella at the beginning of the song is sent to the Very Large Hall aux with the Avid ReVibe II, as well as the Hall aux with the UAD Precision Delay Mod, and the V Hall which has another instance of the Lexicon 224. I wanted the a cappella to sound larger than the other vocals, which have only the 224 and a send to the V Slap aux, with a delay from the UAD Galaxy Tape Echo.

"The main vocal tracks also have a send to the V Amp track, on which I had the SansAmp, for some more edge. The singer has a very raspy voice, which I think sounds great, and the SansAmp enhances this. All vocal tracks go to the Group track, on which I have two EQs and the UAD Precision De-Esser. I use volume automation a lot on individual tracks, so if I use de-essers on them, it can emphasise the esses. For that reason I prefer to have the de-esser on the subgroup at the end of the chain. It makes it easier to control things."

Alessandro Marcantoni: "They had just performed these cover songs at the *X Factor*, and they wanted to record them just like their performances... I tried getting them to add some instrumental overdubs, but they were not into that."

Instrument Group & Master

"The instrument group track has the Brainworx bx_digital V3 M-S EQ, and the only thing I did with that is make the instruments wider. I wanted to open up the stereo image, because the song is almost in mono. Only the drums are in stereo, and although I used two mics on the guitar and hard panned them, they still picked up the same part. It's not a double guitar, so sounds pretty mono.

"Both the Instrument and Vocal Group tracks are sent to the Master track, on which I again have the bx_digital, again to widen the image. I guess I felt I had to do it twice, because once was not enough! I used the Precision Maximizer and Precision Limiter for volume when I sent the mixes out for approval, but I took them off when I sent the mix to the mastering engineers, Pietro Caramelli and Claudio Giussani at Energy Mastering."

Caramelli and Giussani added their own bit of final polishing. Naturally, none of those involved expected one of the songs to become a worldwide hit four years later, but clearly, the quick work that was done by Marcantoni in 2017 was of high enough quality to holds its own in the charts of 2021.

TalkBack

Badsista

WILLIAM STOKES

The world of production has never been more diverse, more accessible or more exciting than it is now. Burgeoning producers who might previously have never had the opportunity to step foot inside a recording studio and prove their worth are making staggering records in all kinds of environments and cultures, and with all kinds of gear — expensive or rudimentary. Badsista, aka Rafaela Andrade, is one. Now a globetrotting producer and DJ, her work with artists like Lei Di Dai Linn da Quebrada and the Multishow Brazilian Music Award winner (in Portuguese: Prêmio Multishow de Música Brasileira) Jup do Bairro has seen her grow into one of Brazil's most eminent electronic music producers. These days she lives in the Cangaíba district of São Paulo, where she is currently working on a project with MC and singer Brisa Flow, as well as on an album of her own.

"It's nice to see other realities," she says. "You want to ask me about my favourite gear and things like that, but for us in Brazil it's not so easy to buy things. The taxes are really high, and equipment is only a reality for the people who have the money. I only started to buy my things last year, like my KRK monitors. A friend gave me a microphone. I have a keyboard controller that another friend gave me as payment to work on her album. Right now I have a proper room to work in, but in the house I used to work in I had to use the living room. There was a lot of noise. It's another reality." She then pauses for a moment before adding, "we can use the word 'ghetto', I think."

At the moment I can't stop listening to
Ahh... The things I'm doing with Brisa Flow! Because, she is writing in Spanish and we have this big connection doing music. We really understand each other. I'm listening to it a lot. I'm really enjoying it. I'm also listening a lot to Brisa Flow's last album — it's really good. I'm also listening to Amy Winehouse, too. I never get tired of it. I wish it was possible to make an album through a medium, or something like that! It's so sad. I wish we could have another album from her. I thought she was an amazing singer, and I really like the way she wrote her lyrics. The way that she was talking about her life. I really like people who write like that. It's not difficult to understand what they are trying to say.

The project I'm most proud of
So, last year I worked with music director Jup do Bairro who is a great, great friend of mine. We used to work together in Linn da Quebrada's band. Linn da Quebrada and Jup do Bairro are both *travestis* — I don't know how to say that in English — trans-girls? They were the two people who believed in my kind of sound. In the kind of sound that we could build together. It's so special that they're still alive, you know. Because in Brazil the trans population are living in danger all the time. If you go out in the street, if you take a bus... things can happen to you. At any time. So it's really special that they're still alive, and that they're singing about themselves — not only about being transgender or whatever — about love, parties, about self-confidence, about transformation. I produced Jup do Bairro; we spent two weeks together. We spent two weeks producing this EP with five or six songs together. It was really special because last year she won a lot of prizes with just that first EP. And it's not even a whole album! We have an awards here called the Prêmio Multishow de Música Brasileira and she won the Super Jury: New Artist Award. It was a big, important thing for us. She won other awards, too... right now we're starting to work on her album but I'm also doing this other stuff with Linn de Quebrada and Brisa Flow. I think that was one of the most important projects that I did because I believed in her, she believed in me, and we trusted each other. So it was really important to me.

The first thing I look for in a studio
I love monitors! But, the atmosphere is really important. I don't like lazy people. I think that the energy is really important when you're in somebody else's studio. I had an experience with another artist when we went to a friend's studio and spent about three hours trying to get one line right. Sometimes the energy just doesn't work out at all. For me, it's all about the music. It's because of the music that I have this equipment now; because of the music that I can send some money to my Mum; it's because of the music that I pay my rent. But, when I go to the studio with someone who doesn't care about the music so much, who cares more about the image, it gets me really bored. I don't like that. I'm here for the music. I really love that people are still supporting me and buying my music, but I don't care so much about the exposure.

The person I would consider my mentor
Lei Di Dai. She's been a dancehall singer for more than 15 years, here in São Paulo. She's a Rastafarian. It's because of her that my first EP is so dancehall. When I met her I was really disappointed with my life. I was sad. I used to play the acoustic guitar and sing in bars for four hours at a time, and earn \$20 or less. When I got to know Lei Di Dai, I had showed her a lot of the dancehall beats I was doing at that time and she asked me to work with her. It was really special because she basically took me and inserted me into the electronic and underground scene of São Paulo. Within one month, everybody — producers, DJs — was getting to know me because

Photo: Thaís Oliveira

of Lei Di Dai. She said "OK, if you want to do this then I'm going to help you." And she helped me with everything. Especially with contacts. She was talking to everyone about me. And she knows a lot of people here in São Paulo! I met her in 2015, and I think the first time I travelled to another country was 2018. She was making connections for me, and teaching me how to work. When I started to travel to Europe I met a lot of people, and it was really crazy because they had such easy access to equipment. Here for us, the taxes on audio equipment and equipment in general makes it so expensive. That's why it's so special to share the knowledge and share the contacts and the network. I know that if Lei Di Dai hadn't done that for me, I would still be uploading music to Soundcloud and sending emails.

The biggest misconception about the role of the producer

This is a tough question! I think the misconception is that the producer is like a machine, where you put a nickel in and then you have a beat. Sometimes there's inspiration and sometimes I'll go two or three weeks without opening up my Ableton Live. Sometimes I've had studio sessions with a friend or an artist where I go and I don't feel any inspiration at all. And we're there trying to make it work and it's not going to work. Last week Brisa Flow spent three days or four days with me, I think, and we just worked one day. The other days we stayed here and watched movies, talked about other stuff, listening to music, did cooking... I think that's part of it. Part of getting to know each other and feeling close to each other, because if you feel close to someone, the process of doing music gets easier. These kinds of things are important.

My go-to reference track or album

It's kind of lo-fi house, but that track from Mall Grab, 'I've Always Liked Grime', it's really, really good. I used to play that track a lot and still want to play it because it really hits me. It was like a new world when I started to listen to that kind of music. What else... some tracks from Rihanna, especially from the *ANTI* album, the last album. It has a good bass. This is really important. There's also a track from Mariah Carey, 'H.A.T.E.U.' — that one is really cool because the beat is so simple yet so... concrete.

My desert island studio item

I think I would take my guitar. There are things that I can do on the guitar that I cannot do on the keyboard. I'm looking for a guitar controller, with MIDI or something. That would be nice. That would save my life! I play more guitar than keyboards.

The producer I'd most like to work with

Anyone...? I'd have to think about it. Kasimyn from the group Gabber Modus Operandi. We met each other in Berlin at the CTM festival, and I think they're really insane with their producing. They mix a lot of Indonesian percussion and melodies with gabber, and this is really cool. I don't know why we haven't produced together yet!

The advice I'd give myself of 10 years ago

I would say: keep calm. Things are going to happen. Keep studying your stuff. Stay a nerd. You're going to make money. I would say, you don't have to feel this imposter syndrome. In the beginning, when I started to get known, sometimes I felt like I didn't deserve that. I felt like I was just pretending. When they said I was good, that I was one of the best producers in Brazil, I thought they were lying. But right now I feel like I am one of the best producers in Brazil. Because I feel more comfortable about my ideas and I feel stronger. About my work. I know what I want to do, I know what I like, what I don't like. People like us, from the places we came from, the path to gaining self confidence is longer to us. We have to do a lot of things to feel like we deserve it. A lot of artists and producers who came from where I came from feel this way. And we're trying to change that.

GOLDEN GEAR

Studer Revox PR99 Open-Reel Tape Recorder

The final Revox tape recorders left the factory almost 25 years ago, yet the enthusiasm and support for these classic machines remains as strong as ever.

HUGH ROBJOHNS

There will be few 'baby boomer' studio engineers who won't go a little misty-eyed at mention of the legendary Revox A77: one of the first all-transistor, high-quality, open-reel tape recorders that non-professionals could afford. And, of course, those of slightly younger years will undoubtedly have much the same emotion for its successor, the elegant B77. However, it is this latter descendant's close cousin, the PR99, that I really want to reminisce about here! Forty years ago the PR99 was launched to span the gap between the Swiss manufacturer's Revox consumer machines and their more expensive professional Studer recorders.

Background

Although the A77 (and later B77) was designed fundamentally for the 1970s (and '80s) domestic hi-fi market, it has been suggested that most of them were bought for professional or semi-professional applications. They certainly became familiar sights in musicians' home studios, educational facilities,

and in radio production offices, as well as making regular appearances in professional recording studios and location recording trucks, too!

The simple reason for this enormous breadth of service was that, despite their notional consumer origins and relatively affordable prices, these were extremely well-engineered, rugged and reliable machines which, when correctly aligned and maintained, achieved remarkably good technical specifications and gave surprisingly little away to pukka professional machines. These Revox recorders could also accommodate 'professional' 10.5-inch tape reels and run at 15 inches per second, when most comparable domestic machines could only manage 7-inch spools and 7.5ips!

Revox's A77 was constructed around a strong aluminium die-cast chassis, just like its Studer cousins, to ensure a stable and accurate platform supporting the transport motors, tape platters, guides and heads. And instead of sharing a single motor and relying on belts, pulleys and cogs to drive the reels, the A77 featured three powerful direct-drive asynchronous AC motors, with the capstan motor being electronically controlled via a tacho-sensor for absolutely speed accuracy — most consumer machines of the era relied on a synchronous motor whose speed depended on the (varying) mains frequency. In the A77 everything was controlled electronically, using relays and solenoids, allowing 'light-touch' transport buttons and versatile remote-control options, while the audio electronics were arranged on modular plug-in circuit cards for easy access and servicing.

Studer Revox also designed and built — to exemplary standards — their own 'Revodur' tape heads, which allowed their recorders to deliver a significantly flatter frequency response than most of their contemporaries could manage. Moreover, head life was maximised through gentle tape paths with modest wraps, resulting in less tendency to develop flats and grooves in the heads. One amusing Revox A77 advert from the '70s stated that the company guaranteed four parts for just a year: the capstan, the pinch-roller, and the record and playback heads... while the remaining 842 parts were guaranteed for life!

From A To B

In its 10 years of production from 1967 to 1977, the A77 model evolved through four Marks, and around 450,000 machines were built! In 1977 it was succeeded by the B77, which built upon the great successes of its forebear while maintaining the same fundamental design concepts, mechanical engineering and build quality. The audio circuitry was largely derived from the earlier model, but the B77's transport control system was completely redesigned principally to use digital logic instead of relays. The B77 achieved slightly better technical performance across the board, and most say it sounded it a little better. It was certainly more generously equipped with features, including sound-on-sound via an internal track dubbing facility.

Around 220,000 B77s were built altogether, from a catalogue of 61 different model variants, and at the height of production, Studer's factory in Regensdorf, Switzerland could churn out around 175 new machines every day! Different model options started with selecting the head format. The 'professional' options included mono full-track, two-track (with a wide 2.0mm NAB-standard guard band), or stereo (with a narrow 0.75mm DIN-standard guard band), using Studer's acclaimed 'butterfly' heads to minimise magnetic crosstalk. These three formats ran the tape in a single direction, of course, while the standard 'consumer' option employed the stereo quarter-track head format to record a stereo track in one direction, whereupon the tape could be turned over and a second interleaved stereo track could be recorded in the opposite direction. Additional options applied to the erase head, which could span the full track, or be split to erase each half-track separately, or the two erase gaps could overlap to ensure the guard band was wiped as well.

> "One reason for the continuing popularity of the PR99 is that virtually every part is still available."

Another key decision when specifying a new machine was the preferred equalisation standard. The A77 featured a switch to select NAB or IEC (CCIR) replay equalisation (although the record format was fixed at the factory). However, the B77's record and replay EQ was determined during manufacture so the appropriate NAB or IEC standard had to be specified on the order. Another factory specification set the two operating speeds: any adjacent pair could be provided between 15/16-inch and 15ips. Other model options included a playback-only version, machines with deleted controls, rackmounting rails, added sync and remote-control facilities for slide projections, and much more besides.

Enter The PR99

Despite this plethora of factory configurations, some of which were obviously already aimed at professional users, Studer Revox announced the PR99 model in 1980. This was an obviously 'more professionalised' version marketed primarily towards »

The most eye-catching feature of the MkII PR99 was the digital tape timer.

the rapidly increasing numbers of local radio stations which were appearing across Europe at that time. These new broadcasters tended to have relatively low budgets, yet still needed workhorse tape machines in great numbers. Studer Revox met that need with the PR99 which was promoted as an 'affordable Studer', neatly bridging the capability gap between the domestic B77 and the nearest professional equivalent at the time, the Studer B67.

The most obvious difference between the B77 and the original PR99 is the latter's raised transport chassis, bringing it flush with the control panel to give much easier access to the heads for rapid lacing and tape editing. Less obvious was the addition of a second damper arm in the tape path on the exit from the headblock (the B77 had only one damper arm on the entry to the headblock). The PR99 also featured transformer-balanced input and output XLR connections to integrate properly with professional mixing consoles, and front-panel buttons selected calibrated or adjustable I/O alignments. A Tape Dump button was also added to disable the take-up spool motor, allowing unwanted sections of tape to be 'played off' during editing without the take-up spool spinning wildly. Another useful feature was the appearance of a pair of Sync buttons, which replayed audio from either channel of the record head rather than the replay head, enabling synchronous recording or overdubbing onto the other track. Standard PR99s came with 19-inch rackmounting rails, but there was also a version mounted horizontally in a wheeled trolley, and another in a flightcase with built-in monitor loudspeakers for portable location recording applications — although this is probably the rarest version to leave the factory.

In most other respects, though, the PR99 and the B77 were still very close cousins, and this frequently created 'operational issues'. In particular, the profusion of front-panel controls with options for input selection, track dubbing, I/O level adjustments, monitoring modes and so on, often lead to embarrassing user errors! Remember, these machines were typically being used by non-technical radio presenters and broadcast journalists rather than experienced studio managers or engineers.

Outside Influence

ASC, a small British pro-audio manufacturing company with close ties to the UK radio broadcasting industry, picked up on this problem and, in 1983, launched a much-simplified version of the PR99 in which all the 'non-essential' controls were removed, with a new grey panel fitted to hide the missing knobs and buttons. But ASC didn't just take things off the PR99: they also installed an elegant digital tape timer to the left of the head block. This was a really useful modification, as the simple four-digit mechanical counter carried over from the B77 was far too basic for use in a professional setting. An additional benefit of removing most of the front-panel controls was that it created space in front of the headblock to fit a large, traditional 'EMItape' splicing block (similar to an Xedit 'Editall' for those to the West of the Atlantic), allowing easier and faster editing.

These ASC PR99s proved hugely popular in the UK, and perhaps that's what spurred Studer Revox into launching their own Mark II version of the PR99 in 1985. This new updated model likewise incorporated a smart digital tape timer, with zero locate, address locate, and auto-repeat functions, and a rather small but functional splicing block complete with integrated cutter to the right of the headblock. Going beyond ASC's updates, Revox also added a varispeed facility (ranging over ±2 semitones) controlled by an on-off button and range knob above the head block, and a Cue-Edit function operated by a mechanical slide switch just in front of the head block. This last feature defeated the tape lifters and replay head hum-shield when in Stop mode, and activated the replay audio circuitry to allow manual audible tape cueing (or scrubbing, as it's often known). A less visible but still quite important improvement expanded the range of treble EQ adjustment in the replay amplifier, allowing compensation for greater head wear. Possibly of even greater significance is the fact that Studer Revox changed suppliers around this time and started using high performance Beyschlag and Philips components, which many claim resulted in worthwhile audible improvements.

My Own PR99

I bought a well-used ASC PR99 MkII from the BBC in about 1998 — it was one of many literally being thrown out, as the BBC considered it pretty much obsolete and disposable equipment by then. It had had a hard life, but it was a half-track stereo machine with butterfly heads and came with a full service manual, so I was confident of being able to restore it to good health. It's still going strong today, although it has been re-capped, had a couple of new bearings and a new pinch-roller. About 15 years ago it had major surgery to remove the record amplifiers and bias oscillator, install a second set of repro amplifiers, and the record head was replaced with a quarter-track replay head. I can now switch between professional half-track stereo and consumer quarter-track stereo formats by using the Sync buttons. Not a standard mod, obviously, but it works well and suited my needs at the time!

Many options could be specified when ordering a PR99. This unit, for example, lacks the mic preamps.

Many claim this Mark II revision was the best version of the PR99, but four years later in 1989 Revox brought out a Mark III model, recognisable by its grey control panel. Most of the changes introduced in the Mark III were really aimed at reducing manufacturing costs, perhaps driven by a need to remain competitive as the digital audio revolution was building rapidly in the broadcast world. New digital recording systems like Sony's PCMF1 and the then-new 'R-DAT' recorders cost a fraction of the price of a PR99 and, although impractical for editing at that time these formats were quickly gaining popularity for recording applications, not least for their longer recording times, less costly media, and more compact and lightweight physicality.

Other aspects differentiating the Mark III include the deletion of the microphone inputs, input selector knobs (in the centre of the control panel), and the Sync buttons and associated sync replay amplifiers (so it was no longer possible to overdub synchronously between tracks). Internally, the electronics were significantly reworked, too, with extensive application of FET analogue switches and op-amps throughout the audio circuitry. On the upside, wider adjustment ranges were provided for treble EQ, bias and erase drive current, presumably to cope with higher-output tape formulations. As a result, though, the Mark III's electronics cards were no longer interchangeable with Mark I/II PR99s.

Revox Or ReVox?

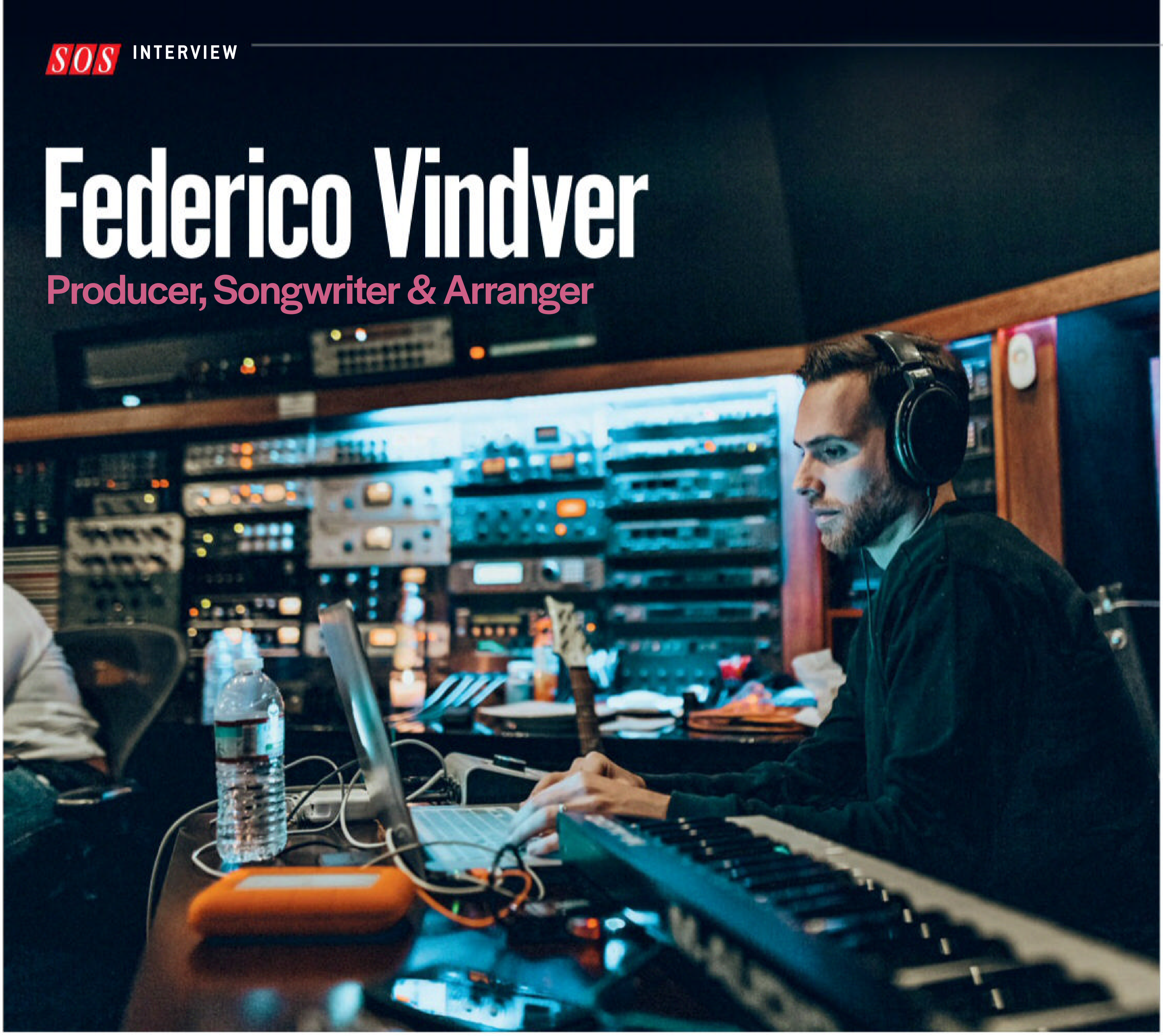
The Swiss manufacturer's name is often seen written as ReVox, but in this article I've stuck with Revox partly because it's simpler to type, but mainly because it's much easier on the eye! The actual company logo used small capital letters throughout, with the V in a larger type size: reVox.

Still Rolling

The last B77 Mark II and PR99 Mark III machines left the Regensdorf factory in 1997, ending 46 years of Revox open-reel tape recorders. That means that, today, even the youngest PR99 is over 25 years old, and the oldest models are around 40, with most having endured hard lives in broadcast studios. Nevertheless, there are always more Revox machines in the for-sale ads than any other brand of tape recorder, and they often change hands today for considerably more than their original purchase price. Such buoyant prices only confirm the impressive longevity and continued desirability of the PR99. The most sought-after models are good-condition stereo half-track machines fitted with DIN butterfly heads.

One reason for the continuing popularity of the PR99 is that virtually every part is still available from a collection of specialist companies that enthusiastically support these revered machines. The same is generally true of its siblings the A77 and even the later 36-series tape machines! Consequently, bringing these elderly machines back to perfect working order is generally quite feasible for an experienced technician with the right tools and test equipment — albeit at a cost; running any open-reel tape machine is inherently an expensive business these days!

The PR99's mechanicals are generally very robust and long-lasting, while elements that suffer normal wear and tear like the motor and guide bearings, the capstan shaft, pinch-rollers and brake linings are all fairly easy to obtain and replace. As for the electronics, the biggest risk by far is failure of the many electrolytic and mains-filter capacitors, which is why most service technicians automatically replace original components as a standard part of any service or renovation — hopefully before their looming failure can do any serious damage! Trimmer pots on the electronics cards can also become intermittent due to dirt and corrosion, or just through excessive use, but again these are easily replaced. Thankfully, there are very few custom or unique circuit components in a PR99, so most electronic repairs are straightforward and inexpensive. Studer Revox's service manuals are very thorough and easy to follow as well, so routine maintenance and alignment presents little challenge to anyone familiar with electronics and tape recorder servicing — you just need to make sure the manual covers the specific boards in your particular machine because, as we've seen, there were quite a few versions over the years. Modifying or converting machines from one configuration to another isn't usually too much of a problem, either, or obviously far more involved than routine maintenance, and there are even companies offering brand-new front-panel and case metal work (in fancy colours, too) as well as wooden cabinets to rejuvenate tired-looking machines. I think it's a fairly safe bet that there will still be Revox PR99s in use at the turn of the next century!



Federico Vindver

Producer, Songwriter & Arranger

TONY BACON

His remarkable versatility has made Federico Vindver the go-to collaborator for Timbaland, Kanye West, Coldplay and more.

In 2019, Federico Vindver was working with Kanye West on the album that became *Jesus Is King*, when Kanye announced a new rule for everyone in the studio: "We can never have more than four sounds at the same time." Rick Rubin would sometimes use a similar directive, and it's one that certainly concentrates the mind and, indeed, the ears.

Federico considered the options on the piece they were developing. Kick, snare and vocals were already there, so there was room for only one more sound. Maybe lose the kick for a bass? Maybe make it all voices? Whatever the solution, it had to be restricted to four sounds.

"That was a situation where simplifying helped a lot," Federico recalls. "Like, how do you make a chorus or some other section really pop if you only have this very limited amount of options? Sometimes you have to simplify more. Sometimes we'd even go to a cappella sections. For example, 'Use This Gospel' on that album has a Kenny G sax solo, and so we had all these combinations of how to do that solo. Do we put a pad behind it, or a piano, maybe? In fact, we made it kind of an a cappella sax solo for the most part. It really brought your attention to it! Imagine, though, if it had big drums and pads and bass — it would not have had the same shocking effect. That was definitely a situation where 'less is more' applied."

Something From Everyone

This sort of adaptability has been the key to an unusually diverse career. "Something that defines me as a producer," he says, "is that I love to work with artists who are completely different from each other." He's not joking. He's worked with everyone from Mariah Carey to Tee Grizzley, from Christina Aguilera to Chance the Rapper, from Muse to Justin Timberlake.

"One time I was working with Polo G, a great young rapper from Chicago, and at the same time I was finishing a song with Josh Groban and Kirk Franklin, completely the other side of the spectrum. I thought, well, nobody under the age of 50 would probably buy Josh's record, and at the same time I'm working with a guy who nobody over the age of 18 might buy his record." He laughs at the thought, enjoying the diversity.

Although he laments that he doesn't get to spend as much time there as he'd like, Federico Vindver has a well-equipped studio of his own.

"And then I'll go to Nashville and I'll make some white American records, I'll go to Atlanta and work on more street hip-hop records, and from there to Miami or Colombia to work on some ghetto Latin records. And I'm working with mega-superstars who are gazillionaires, then back to new artists who have no money. I go from country to hip-hop to rock to anything."

Rick Rubin comes to mind again. "Yes, he's one of my heroes. He has meditation-music albums with Middle Eastern chanting, he has heavy metal, he has hip-hop. That's the kind of producer I'm trying to be, somebody who can tackle very different projects. It keeps it fun for me. I was just doing a Christmas album with Meghan Trainor, all orchestral, big-band, Nat King Cole-sounding things, and from that I went to something weird and electronic with a Latin artist. Then I did some reggaeton records. That's what I love about music. I love it all!"

Jazz & Bad Soundcards

Federico was born in Argentina. When he was 19 he moved to Spain and then to Mexico, to a gig playing piano at a big hotel resort there. He practiced assiduously during the day, and in his early 20s he majored in jazz performance at the Frost School of Music at the University of Miami. "It was a comprehensive programme that also offered conducting, orchestral arranging and computer technology," he says. "That four-year course really helped me out on so many levels."

He was playing keys with a variety of bands, mainly to make ends meet, and that inevitably led to some recording sessions. "The producers didn't really know much about music. They were more like engineers working the computer. I was always very computer-savvy; I'd taken a year of computer science in Argentina, too. So I'd see these guys and think man, I can do better — because I know music. All I've got to do is learn this Pro Tools thing."

His budget didn't allow that, exactly, but a friend gave him a beat-up PC and he acquired copies of Acid Pro and Fruity Loops, plus a "really bad" soundcard, and he began making tracks. "The bands we were playing with would say, well, your really dumb setup actually sounds better than the studios we're going to, so we're going to do it with you. And that's how it all started."

He studied the music he liked, curious how people like Timbaland, or Pharrell and the Neptunes made their tracks, and attempted to emulate them. He started touring, notably with Lauryn Hill and then Marc Anthony, Jennifer Lopez and Ricky Martin. "Even then, I was trying to produce whoever I could on the side. And I'd always be on my laptop. If I wasn't making tracks, I'd be learning plug-ins, learning music, listening, always doing more at the same time."

Eventually Federico realised he liked the studio more than touring, so he moved to Los Angeles to try to become a full-time producer. The biggest problem he faced at first was how bad his computer was. It would freeze every 20 minutes or so — which at least taught him how to work fast. He had a Yamaha Motif keyboard he used for gigs, and he'd record that. Quickly.

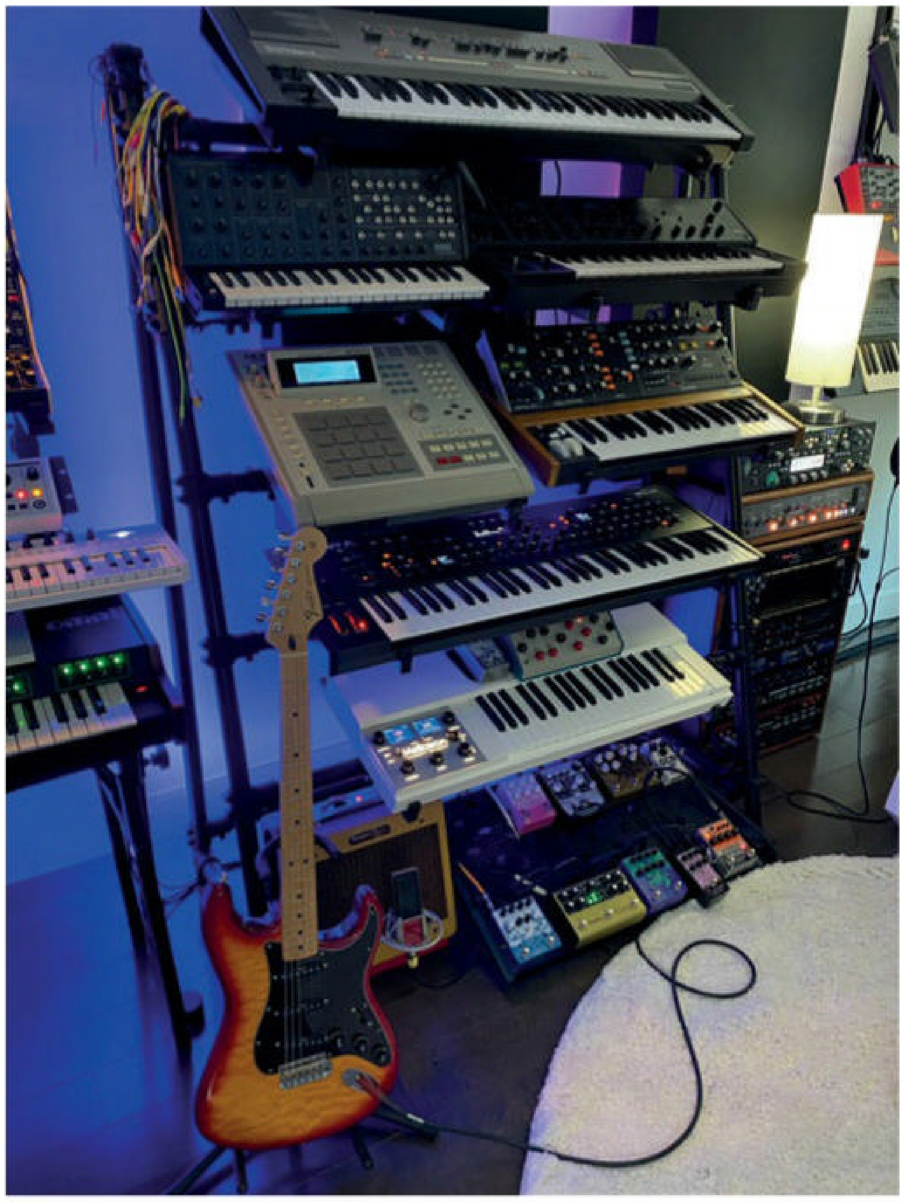
Around 2006, Federico graduated to a MacBook Pro and Logic, which came with some very usable plug-ins. It was a big improvement, and his music-making and recording moved steadily forward. He met Timbaland during a session for a band called New City and began working more with him, and that in turn led to Kanye West.

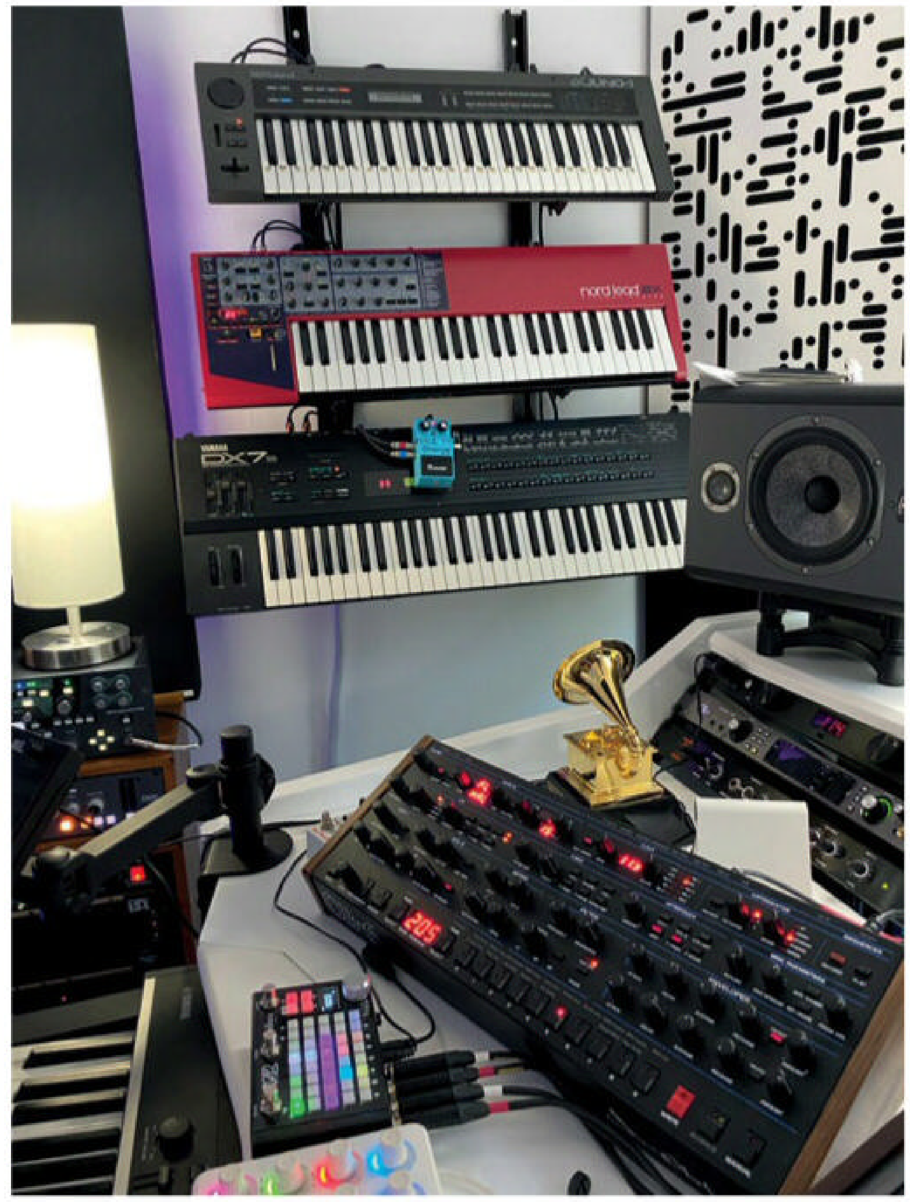
Early Music

Kanye wanted to work with Timbaland, but their respective schedules never quite coincided. One time, though, Tim and Federico had been working on some sessions in Miami and were just about to return to LA when word came down that they should stay. Kanye would be with them tomorrow — at 9:00am. "This," says Federico with a smile, "was the first time I'd heard somebody say '9:00am' in the last 20 years."

They figured they might as well get there at 8:30, just in case. "And by 8:45, Kanye entered the room. It was like hey, how are you? Amazing," Federico remembers. "Then he played us a bunch of music, and we started working with him, at first with Tim, and then we began going to his studio in LA, and we ended up working with him for, I would say, a year and a half, two years. I'm still doing stuff with him, too, nowadays, but it was really a nice working relationship. He works with so many people, so you get to meet a lot of talented producers and engineers."

The sessions would turn into Kanye's 2019 Christian/gospel album *Jesus Is King*, and Federico gradually became what he describes as the everything guy. At first, he was presenting ideas along with





This diverse rack of instruments reflects the variety in Federico Vindver's work. From top: Roland HS60, the 'home keyboard' version of the Juno 106; Korg MS20 and Yamaha CS10 analogue synths; Akai MPC3000 sampler/sequencer and Behringer Poly D synth; Sequential Prophet X polysynth; and M4000D Digital Mellotron.

Visible in this shot are Federico's Roland Alpha Juno 1, Clavia Nord Lead and Yamaha DX7 synths (on wall), with his Oberheim OB-6 desktop module in the foreground.

some of the other people in Timbaland's camp — sometimes full beats, other times straightforward chord progressions, then maybe samples from records. He recalls that in the first week alone they worked on ideas for 40 or 50 songs, and gradually he became closer and more keyed into Kanye's working methods. Which can be... different.

"I would have maybe 20 little pieces of music prepared for him to play," he says, "and he would freestyle stuff on them, so that was one thing. Then also I was working on his vocals a lot. They would record his vocals and give them to me. He might do his freestyle thing for 20 minutes on something, so I would take that and try to condense it into what felt to me like a song. I would try to find the hook, try to find the verse, I'd put it together that way, and then he came in and would make notes. And sometimes he'd say he really liked the edit, but there was a part he did that I didn't use that he really liked. So could we go back to it? He'd remember everything, which was incredible. And so, of course, we'd do it that way."

Play It Again, Federico

Another strand of Federico's work on the *Jesus Is King* album was to replay samples. "Kanye would say he liked this or that sample, but maybe he wanted to take the drums out of it, mute the bass, whatever. So I would do replays. I can play every instrument, so I'd be in a room with guitar, bass, keys, drums, percussion, and I'd replay the whole sample note by note, putting it together so it sounds exactly like it. After a while you do get better at that, but it's definitely challenging. Sometimes you need to do some research about what they used on this record, who the band was, and it can take time to recreate all the necessary sounds."

On other occasions, he has needed to dissect a sample and rearrange it in a new way. "Kanye will have songs that are — how can I put it — in a very unfinished form. They might have him freestyling over a sample, so maybe I want to put drums on this, and I have to redo the sample so we have that, change the key, perhaps, or put in a vocoder on his vocals, create some harmonies, all kinds of things to finish the record. So another role I took on was the finisher. That's why I got credit on almost every song on the *Jesus Is King* album, because everything was at some point coming through me, to give the final touch. Some of them I started from scratch with him, some of them he already had and it was me doing finishing touches."

There would be sessions to record real instruments. "He would say, 'OK, I want this to sound like marching-band horns.' So I'll go into a horn-section session, record a bunch of songs, and I'd have to rearrange them right down. My background, writing for big bands in the jazz programme at Miami, came in handy there."

For another piece on the *Sunday Service* album, Kanye decided he'd like to record a choir of 150 people. "There was no studio in LA that could accommodate 150 people to record," Federico recalls, "so we found a rehearsal studio where we could do it. Then the problem became how to mic all these people for an effective recording. We tried so many combinations of techniques,

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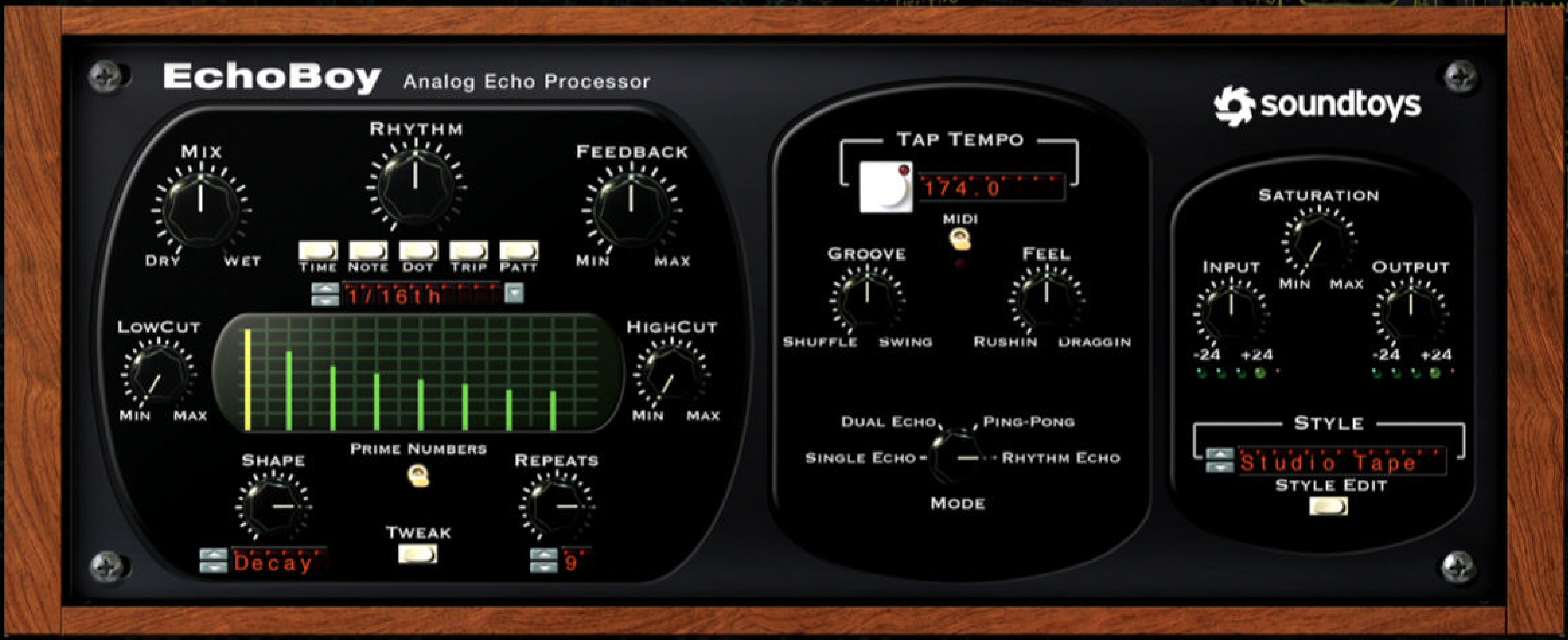

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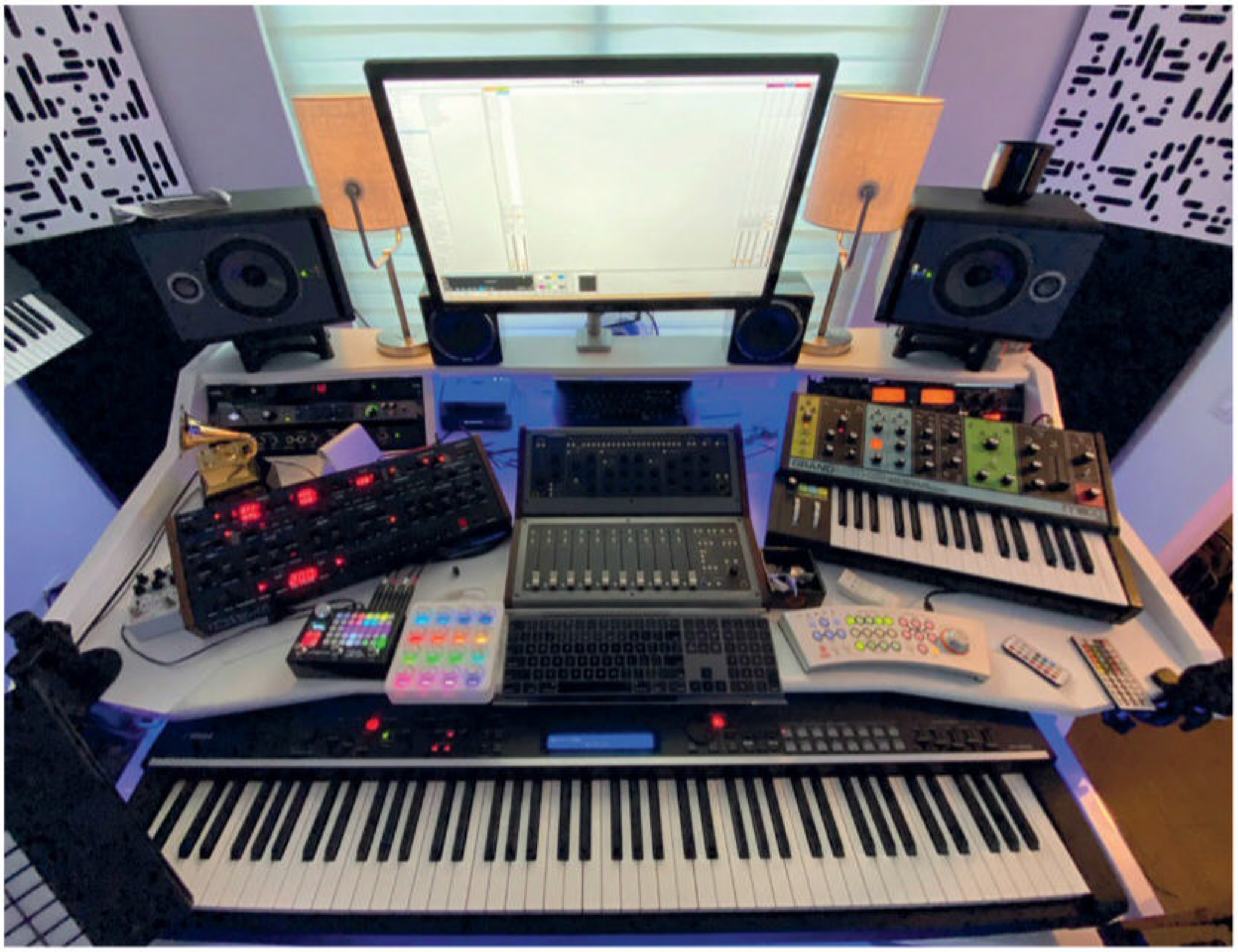


Federico Vindver does much of his work in Ableton Live. Softube's Console 1 and Console 1 Fader control surfaces sit above his controller keyboard, with his beloved Moog Grandmother to the right.

but Kanye felt it never sounded as big as he wanted. He said it sounded too much like a studio, but he wanted to feel like he was there."

Eventually, Federico came up with a scheme. "I used a series of microphones with different pairs in front of the choir. The first pair was at the very back of the room and very wide — as wide as I could. Then there would be further pairs of mics that got closer and a little less wide, making a pyramid shape as the mics got less wide and closer to the choir. We also put spot mics on the different sections of the choir. And having all those mics gave us enough control over how we wanted it to sound. So," he adds with a contented sigh, "we nailed it. Working with Kanye, it can be everything from miking that 150-piece choir down to a song that just needs a better kick drum."

Everyday Life

One of the many lessons Federico has learned from working with Kanye is how much benefit can come from the luxury such an artist has to spend time (and, necessarily, money) considering multiple options. Eventually, Federico notes, you start to see how malleable music is. He recalls a similar example from when he worked with Coldplay on their 2019 album *Everyday Life*. There was a song where Chris Martin and the band felt they still hadn't nailed the string sound they wanted for it.

"We spent a few days looking for that string sound. OK, sampled strings? Well, not so good. Real strings? Still not there yet. Oh, Oberheim strings? Ah, almost there. Eventually, I played this sound, and I think it must have been the simplest sound I've ever done — like a filter saw wave pad with a very long attack, so it sucked into the sound, ver-oomph. I added some street noise on top of that, and the street noise had the same envelope as the saw wave, giving the impression like when you reverse a sound. And Chris was saying, 'Oh, that's the sound!' He loved it."

Then came what Federico describes as the genius moment. "Chris took that sound and wrote a whole other song with it! And the original song, the one we were looking for the string sound, never even made it to the album. The new song he wrote with the sound was 'Everyday Life', which actually gave the name to the album. It just shows how these genius people — like Kayne, like Chris — can work creatively in a far less structured way than most people."

Forward Planning

Federico had come to Coldplay through Timbaland, who'd worked with the band before. Federico and Angel Lopez went into a studio in Malibu with Chris, expecting Tim to turn up, but he was sick and couldn't make it. "Chris said 'OK, I guess it's just the three of us today. Let's try to make something.' And it was magical," Federico says. "We kept on working for a few days, then Tim came in and did a few days, too, then went back to Miami. At the beginning it was mostly writing stuff with Chris, and then eventually he said he had this new Coldplay record they were doing, and he wanted to see if we could hear anything."

There followed sessions where Chris opened up the working versions of songs, and they would add sounds, change things around, try different approaches. "A few songs made the album where the first versions had been jams that we did together, like 'Champion Of The World' and 'Orphans'," Federico remembers.

He was impressed with Chris's ability to think well ahead. During the first few days they worked together, Federico was playing him some of the sounds he'd prepared in advance, a system he uses regularly. "For me, Coldplay was ambient, pads, big sounds, big drums, certain guitar sounds, things I know they use, so I prepared literally hundreds of sounds ready for these sessions. One of my sounds I played, Chris was like, 'That's it! That's going to open the album after this one!' I'm like, What? Why are you thinking about two albums from now? And this to me was another genius move, right there, to be able to think so far in advance." Federico says the result is due to become a short introductory piece opening Coldplay's forthcoming album, *Music Of The Spheres*, due for release later in 2021.

For the sound in question, he drew upon techniques he's developed over the years, particularly exploiting Ableton Live, his prime working environment, and what he calls its

Another eclectic selection of instruments! From top: Behringer Neutron, Pro One, Model D and Wasp Deluxe synths; Moog Slim Phatty module; Arturia Keystep Pro controller; ARP Omni MkII string machine and (left) Makenoise Strega and 0 Coast modules.

Kitchen Church

At one point during the production of the 2019 *Jesus Is Born* album by Sunday Service, Kanye's gospel group, Federico Vindver found himself in a cabin in Wyoming — the successful producer's life is never dull — pondering a problem with a recording of a live gospel choir that Kanye wanted to use. Try as he might, Federico couldn't get it to sound as if the choir was in a small church, the intention for its use on 'Sweet Grace'.

His solution was to put two C414s in the cabin's kitchen, blast the recording through some speakers, and resample it. "And surprisingly, with all that kitchen reverb, it came out great! It gave that reverb sound I just couldn't get with a plug-in. On 'Sweet Grace' you hear the drums and everything, and a big part of it was going through these 414s to get that ambient sound — and then the Sunday Service gospel choir recorded on top of that, too. More proof that you have to resource whatever you can to make things sound the way you want them."

modular-like qualities. "I can use all kinds of MIDI sources to modify any parameter on anything I'm doing — a kind of sound-design dream. For this sound in particular, I was assigning Ableton's key tracking, so the higher or the lower you play on a keyboard, that will control something, in this case delay time. So if I played a higher chord, it's going to move my delay, and that created a crazy wobble in the delay, because I'm changing the time and changing the pitch. And it's interesting because it does this depending on what I play."

He estimates that 85 or 90 percent of his work today takes place in Ableton, with maybe a little Pro Tools and some Logic. "Sometimes I'll even use FL Studio. I like to change things up — it makes you think differently. Once I learn how to do something, and I feel really comfortable and like I'm mastering something, it's almost like, well, I'm never gonna do that again! Let's go to an uncomfortable place."

Outside The Comfort Zone

When we speak, Federico is preparing sounds for some sessions with the Mexican pop duo Jesse & Joy. The songs are all written, so he's working on his palette of sounds ready to present at the studio. "I do find it's good to separate your sound-design and sound-searching days from your music-making days," he says. "When I make music, I don't want to be searching for sounds. I want to have everything ready to go. So when I'm bored and I feel I'm not able to do anything good musically, I go into that mode of sound-searching, maybe downloading new plug-ins, or going to my synths to record some stuff." His current faves from a large collection of synths are his Moog Grandmother, a reissue Prophet, and some of the Make Noise modular stuff.

He tells me about another of the curveballs Kanye West threw into the studio one day. "Kanye said he wanted to record a choir of 500,000 kids. And he was serious, too. I thought to myself, well, getting 500,000 kids is not going to be that easy. But this kind of thing makes you think! It's almost like a Zen riddle. What's he trying to do, you know? Eventually you find your way around it. But I love that he really takes you out of your comfort zone. Whatever is easy, he will not do it. And the lesson to be learned is that living outside the comfort zone will bring unexpected results, that it will push the limits of what you can do."

Federico works hard to push those limits. It can be important, he says, to let your subconscious mind take over. "And that's what's hard in producing. I'm big into ergonomics. When I work in my studio at home, which unfortunately doesn't happen too often, I try to carry that philosophy. I have a MIDI Fighter set up so when I hit one of its buttons, it will do something I do often — for example open a reverb, maybe, or perhaps open a delay — so I don't have to go looking for a plug-in when I'm working."

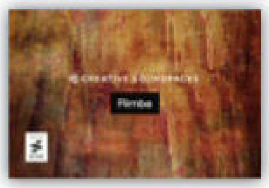
This circles back to his youthful jazz studies and performances. "I like to improvise when I'm creating," he says, "I like to feel like I'm playing an instrument. If it's going too slow, if I have to go on the computer and edit things, it's not going to work for me. I have to feel the immediacy. All my synths, all my drum machines, everything is plugged in, everything is ready to go. I have it set so it's automatic — my hand hits a button and it brings what I want."

He reckons that most of the time, he'll go with the first thing he does. "If I love it, I keep it. If not, I go to another sound or another idea. I love to keep that energy, that thing of not really being sure what I'm doing — but we'll see. It's important to let yourself go, because it brings new ideas, it brings unexpected results. And I've become friends with the unexpected."

SAMPLE LIBRARIES

Orchestral Tools Rimba

Sine Player



★★★★

Though media composers love creating colossal orchestrations, there comes a time when you have to think small. Intimate travelogues, sensitive wildlife documentaries and sun-drenched holiday ads don't need bombastic brass and battering drums. Such productions require a light touch, and that's where Orchestral Tools' Creative Soundpacks come in. In contrast to the epic sweep of the company's symphonic titles, these budget-priced mini-libraries focus on smaller, more intimate sounds such as living room piano, 'forest percussion' and the other-worldy verrophone, combining a subtle organic approach with a sense of exoticism.

The latest sound pack is Rimba, a 6.6GB sample collection which compresses down to 2GB installed. It runs exclusively on OT's Sine Player, an efficient if somewhat limited sample engine undergoing an update at the time of writing. Recorded drily from two mic positions at the company's T-Rex Classics studio, Rimba's nine instruments are presented *au naturel* and also in 10 processed patches featuring various layering, echo, filter and distortion treatments.

As the title suggests, the main instrument is a very decent Musser M250 concert marimba sampled at multiple dynamics over its full A2-C7 range. Played with medium-soft mallets, it sounds clean, resonant, incisive and sweet-toned, though resonance is noticeably reduced in the top octave (a defining marimba characteristic). Finger hits provide a nice alternative timbre when played at higher velocities, hot rod hits (which suffer from round-robin dropouts on their top Ab and Bb notes) could pass for a xylophone in the upper register, and the swishy brush hits work well for light rhythm patterns.

Also included are a melodious, soft-toned log drum (aka tongue drum) and a cheerful South African kalimba, both ideal for programming bubbling, jungly tuned perc grooves. The remainder of the instruments are unpitched: the hefty, cajon-like bass thump of the Jicara de Agua (an upended pumpkin gourd floating in a larger, water-filled gourd) makes an excellent kick sample, while a custom-made pair of toms with birch wood playing heads and small wooden boxes turn in a slender selection of single-hit knocks and scuffling quiet rolls. Meanwhile, gentle rainstick shakes and bamboo wind chimes add a relaxing and therapeutic head-shop ambience.

Exhausted by all this nature, I dipped into the processed patches and enjoyed the ethereal 'Mountain Alder' ghost-choir pad and the seismic bass rumble of the 'Oak' patch. The high point was the wonderfully atmospheric pitched rainstick rolls, low chords on which evoke images of life forming in the broiling Cambrian seas.

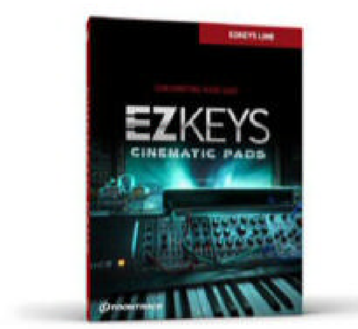
Credit for this eclectic collection must go to OT's Frederik Thyssen, who played the instruments, created the processed patches and devised product ideas for the entire Creative Soundpacks range. There's much to enjoy here: the idea, instruments and price are all agreeable, and although its scope is limited, this sound pack refreshes the parts other libraries cannot reach. *Dave Stewart*

€49

www.orchestraltools.com

Toontrack EZkeys Cinematic Pads

EZkeys Expansion



★★★★

EZkeys is the virtual piano player in Toontrack's excellent 'virtual musician' product range. If you need keyboard-based parts but don't have keyboard-based skills (or a budget to hire them), it's a seriously good tool. Toontrack offer a whole range of EZkeys flavours, each featuring the same underlying performance engine, but containing different sample sets. And if you already own one EZkeys product, you can purchase these additional sound sets at a substantial discount. Initially, these sound expansions focused onf various styles of piano, but other keyboard-based sounds have also been offered and the subject of this review is the latest of these: EZkeys Cinematic Pads.

Cinematic Pads is built around a collection of 17 underlying sampled instruments. Although there is a comprehensively sampled piano amongst these, those samples appear in a reversed format whenever blended into one of Cinematic Pads' 70+ presets. The other sound sources include a number of analogue and modular synths and the exotic Togaman GuitarViol, a bowed instrument that is an amazing cross between a guitar and a viola. The presets are organised into six categories — Atmospheric, Bells, Fat Pads, Modulated, Saturated and Strings — and even a quick browse will reveal the underlying quality of the sounds. There are some wonderful textures here, with plenty of character and depth.

All the usual (very powerful) EZkeys MIDI browser/sequencing options are available so you can, of course, use any of the other EZkeys MIDI pattern content you own with Cinematic Pads. However, these are pad sounds, so sustained chords or slower-moving passages are going to be the most appropriate starting points. The EXkeys front end also includes its usual macro-style sound-shaping controls. Attack, Release, Low-pass Filter, High-pass Filter and Ambience seem to be ever-present, and an occasional additional parameter such as an auto-wah speed control appears in some presets as required.

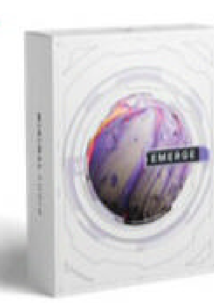
A limited sound-shaping control set is perfectly understandable with the pure piano-based sounds of the more mainstream EZkeys libraries, but it is perhaps a bit more restrictive in an instrument designed primarily to produce pad sounds. After all, lots of other pad-based virtual instruments offer comprehensive synth-style control sets for sound editing or creation. I suspect that control set difference is what will define the most likely users for EZkeys Cinematic Pads, making it most attractive to existing EZkeys users looking to add high-quality pad options alongside their existing conventional piano sounds and EZkeys workflow. Synth-heads who like to dig into filters and modulation options to roll their own sounds might find it less obviously appealing. However, this observation didn't detract from how impressed I was with the sounds themselves. Indeed, if you are already invested in the EZkeys platform, Cinematic Pads would make a great addition. Media composers will find plenty to sonically inspire, but the library would work equally well in any ambient music context alongside other organic or electronic sounds. *John Walden*

$149

www.toontrack.com

Minimal Audio Emerge

WAV



★★★★

Emerge is not one of those 'coffee shop ambience, birds tweeting' collections of sound

effects, but rather a huge collection of overtly processed sounds divided into categories and then subcategories for use in contemporary soundtrack composition. All the sounds are presented as 24-bit WAV files, so you can import them into a sampler or use them as they come.

The main folders are named Organic One Shots, Impacts, Textural Mutations, Atmospheres and Future Synthetics. Look inside Organic One Shots and you'll see subfolders entitled Earth, Ice, Metal, Water and Wood. Earth is full of complex, short sounds, most having a fast attack to produce sci-fi gun noises, metallic thuds and so on. Ice holds more of the same with, perhaps, a slightly colder character. The Metal sounds have more metallic overtones, while Water samples have a more fluid character. Wood offers some slightly darker, impactful sounds. Inside the Impacts folder, the Full Impact sounds are more weighty, often with longer tails and some with a mechanical character. Low Impacts offers sounds with more low end.

Textural Mutations includes folders named Earth, Metal, Water, Wood and Ice. The tonal leanings between sections are similar to those of Organic One Shots but the sounds themselves tend to be longer, lasting several seconds in some cases. The Water sounds are the most 'liquid', with treated recordings of flowing water being evident in many of them. Atmospheres offers Fire, Metal, Water and Wood. There are sounds with slower attacks, crackling flames, alien angle-grinders, lava, bowed metal and so on, some lasting 20 seconds or more.

Finally, we have Future Synthetics, which is divided into Cinematic Basses, Radio Textures Synth Variants and UI Computations and Glitches. The basses have in common plenty of low end but that's about all. There are Doppler-style spaceship fly-bys, mysterious machinery and dramatic swells, but nothing resembling a conventional bass sound. Radio Textures is further subdivided into Micro Static, Signal Ambiences and Static Ambiences. Micro Static offers short bursts of electronic sound, Signal Ambiences similar noises with longer decays and Static Ambiences some interesting crackle-based sounds.

Synth Variants offers Stringed Resonances, Synth Swells and UI Resonances, the common feature here being a greater consistency of sound and pitch throughput the duration of the sounds. Inside the UI Computations and Glitches folder are four more subfolders for Data Computations, Micro Computations, NanoGlitch and Resonances. The Micro sounds are again short bursts, whereas the rest are several seconds in length. You'll find FM-style metallic edginess, glitchiness and granular sounds here.

Many of the sounds have a similar character, allowing a scene to be supported consistently without having to repeat the same sound. The majority are best suited to sci-fi productions, though you may find some musical applications. Each sound effect is named, and my hat goes off to whoever sat down and thought up all those names. *Paul White*

$79

www.minimal.audio

[Audio examples of this month's libraries are available at www.soundonsound.com.]

Q Are passive mics less noisy than active ones?

I record spoken-word narration, close-miked in a small room, with basic room treatment. While I have no noise-floor problems, I've noticed that some of my mics have a lot less noise than others. I've made several recordings of room sound at a quiet time of day, with the gain on each mic set to give the same voice output level from each on the meters of my recorder. It appears that my Beyerdynamic M201 and Royer R101 ribbon mics, which are both passive, are the quietest, giving an ambient noise floor of around -65dBFS, while my AKG C414 XLS capacitor mic set to cardioid is -52dBFS, a difference of 13dB. Am I correct in saying that passive mics are noticeably quieter than active mics?
SOS Forum post

Hugh Robjohns

It depends! Every mic picks up the local ambient acoustic room sound — air molecules in a warm room constantly bang into each other and into the mic's diaphragm, generating noise voltage at the mic's output. This captured ambient noise level depends on factors such as the mic's polar pattern and the size of its diaphragm, but the mic's frequency response also plays an important role; dynamic mics may be perceived to be less noisy than capacitor types simply because of their relatively curtailed HF response, for example.

The source of electronic noise in a passive mic is related to the electrical resistance of the ribbon or moving coil and/or the output transformer, if present. If we assume the total electrical resistance of the generator is 150Ω, the noise voltage due to electrons bumping into each other inside the circuitry (Johnson noise) is theoretically -130.9dBu. Let's round that to -131dBu. (This assumes an ambient temperature of 20C and a measuring bandwidth of 20Hz to 20kHz; change either and the numbers get skewed). Interestingly, if the mic's impedance is closer to 30Ω, as with some vintage ribbons, the theoretical Johnson noise voltage drops to -138dBu but in practice it typically means less sensitivity and thus more preamp gain to achieve the same signal level — which, of course, raises the noise level again!

A DC-biased capacitor or electret mic requires an impedance conversion circuit and the active electronics involved inherently introduces some noise. FETs usually generate less noise than valves, and this electronic noise is usually specified as 'self-noise'. It is typically quantified by replacing the mic's capsule with an equivalent value of capacitor (thus removing any acoustic noise from the equation), measuring the output voltage (which is now only electronic noise), and relating that back to the equivalent level of acoustic noise needed at the mic's capsule to generate the same output noise level. Impedance converters don't usually provide significant gain, so only introduce a few decibels of noise. The most significant source of noise from the microphone is usually acoustic rather than electronic!

In this context it's worth noting that small-diaphragm capacitor mics are inherently noisier than large diaphragm ones; the difference is typically 12-20 dB. There's a variety of reasons for this, such as the diaphragm tension (affected by the polar pattern), and the diaphragm surface area sampling sound waves. A good large-diaphragm mic might have a self-noise figure of 6dB SPL, whereas a very good small-diaphragm mic is more likely to be 18dB SPL or higher.

Many manufacturers only measure the impedance converter's electronic noise because it's much easier to do and gives a better figure than including the acoustic noise contribution too. However, companies like DPA and Neumann usually measure the actual acoustic and electronic noise together by placing the mic in an anechoic chamber: this relates to real life more accurately.

Another factor that affects the perceived noise is the mic's sensitivity. In general, passive mics have a much lower sensitivity than capacitor mics, so output a smaller electrical signal for a given acoustic SPL. That means more gain is required to raise the signal to a usable level, and the most significant factor in perceived microphone noise is usually the mic preamp. Preamp noise is broadly proportional to the amount of gain being applied, but it's not always a linear relationship and many preamps are effectively quieter at higher gains than at lower or middle settings.

A 150Ω termination across a mic preamp's input, as used when measuring a preamp's equivalent input noise.

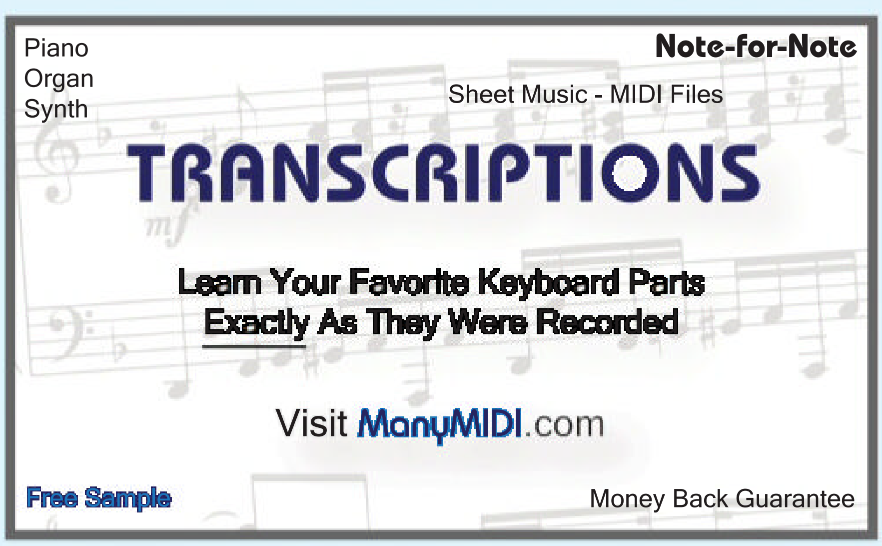
Mic preamp noise is specified as an EIN (equivalent input noise) figure, and typically measured by connecting a 150Ω resistor across the inputs, applying a known amount of gain (usually 60dB), and measuring the output noise level. Whatever that level is (say -67dBu), we then subtract the gain being applied (the 60dB given above) to give the EIN figure: -127dBu. As mentioned above, a perfect 150Ω resistor generates -131dBu (at 20C, between 20Hz and 20kHz) but a preamp contains lots of electronics and even the best will add 3 or 4 dB of self-noise. So a good real-world EIN is around -127dBu. The best designs will do slightly better but only by 1-2 dB and the worst will be many decibels higher.

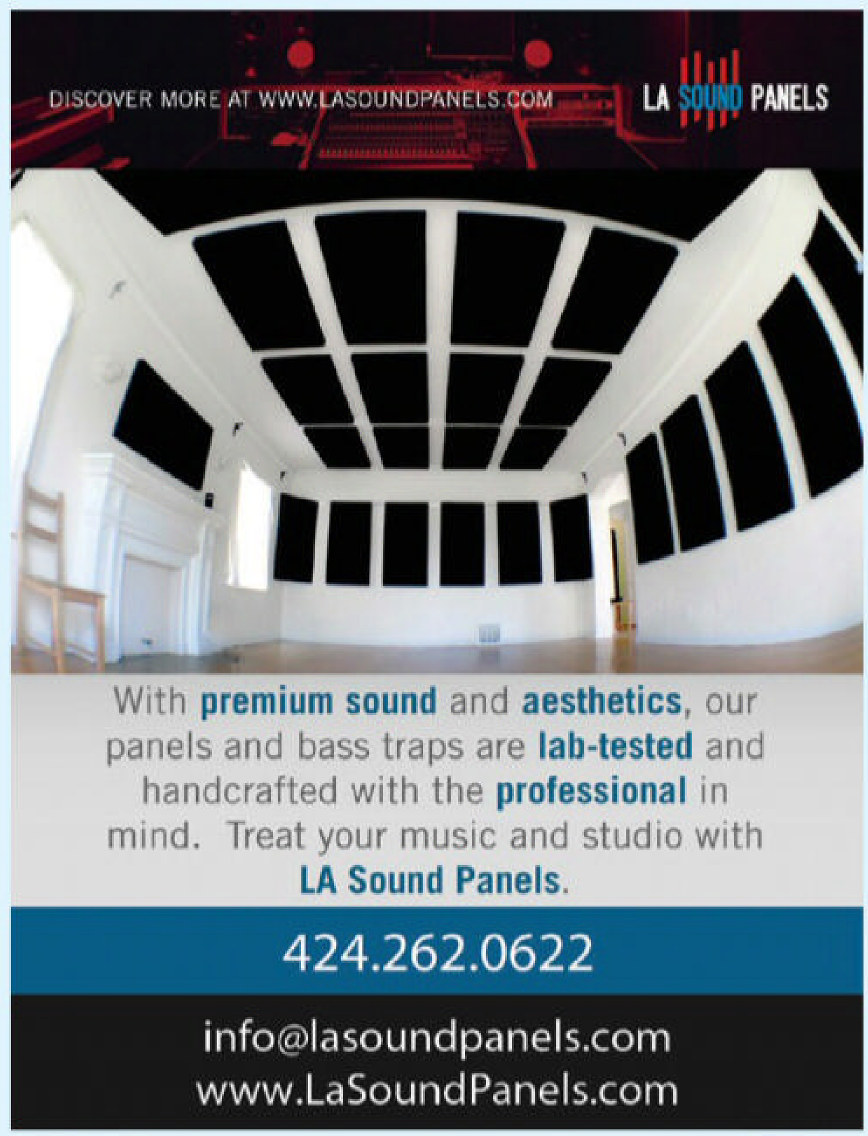
It's possible to make the EIN number appear better than it really is, by reducing the source resistance applied to the preamp's input. Some manufacturers employ 50Ω or even a short circuit (0Ω), giving a much more attractive figure! If the source resistance (and ideally the temperature and bandwidth) isn't specified, don't trust the number!

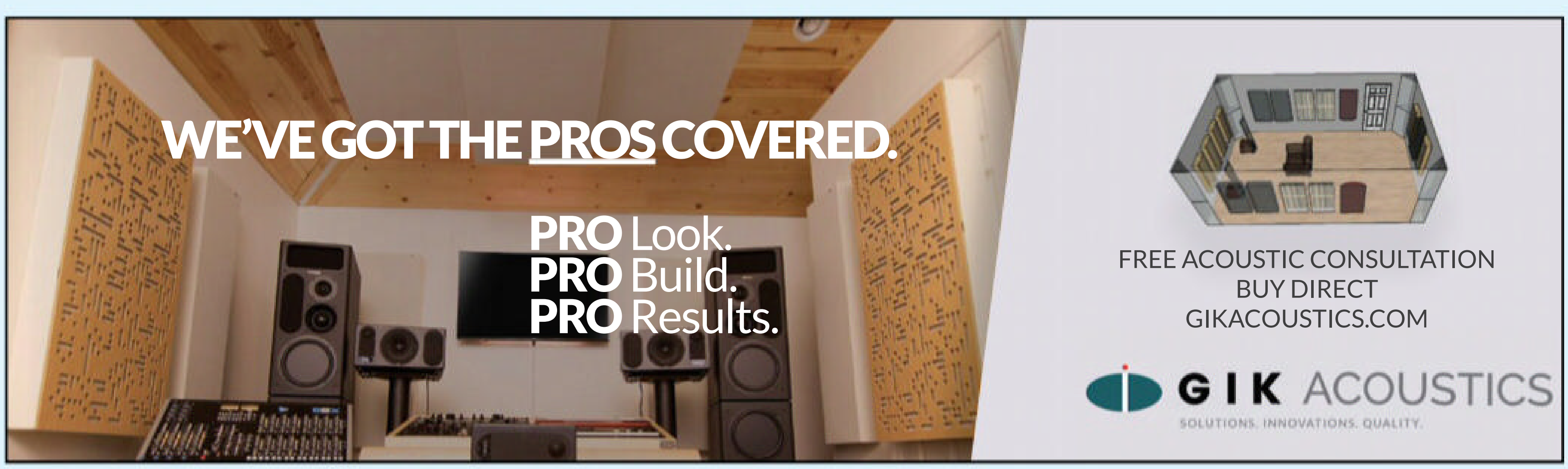
So, assessing microphone noise is complicated because there are lots of interacting factors. Thankfully, in all but the very quietest rooms, and assuming the equipment is decent, ambient acoustic noise usually far outweighs the electronic noise, so we can choose mics for their other important characteristics.

SOS SHOWCASE

David Carson (Regional Sales Associate): david.carson@soundonsound.com







SOS WHY I LOVE...

THE ELECTRIC GUITAR

PAUL WHITE

In my latter few years at school I was the drummer in what was a more than decent local band in my home town of Barnsley, but I also had an interest in electronics, which lead to me building simple treble boosters and fuzz boxes for our guitarist. I managed to pick up some very rudimentary guitar playing tips from him and set about tackling 'House Of The Rising Sun' on a very nasty acoustic guitar with an action like a suspension bridge. However, it kept me interested to the point that I bought a secondhand Futurama electric guitar so that I could test the effect boxes I was building. When I moved to Malvern to study electronics with 'The Ministry', I took my drums with me but soon traded them in for my first proper electric guitar: a very early Gibson SG Les Paul Junior. Since then I've been on a journey of discovery, buying and selling instruments and, more recently, building my own from reclaimed furniture and mahogany window frames. The lovely thing about electric guitars is that you can tweak them to your own preferences in terms of both how they feel to play and in what features you add to them.

In the studio I create a lot of quasi-ambient/chill music under the Cydonia Collective name, in collaboration with local musician Mark Soden and with odd guest contributions from acoustic legend Gordon Giltrap and vocalist Alice Rose Wild. In almost all cases, the electric guitar plays a pivotal role and often takes the main melody part.

By the creative use of pedals and plug-ins, the guitar can produce a huge range of sounds, many of which can't even be identified as coming from a guitar. For example, I converted a budget Epiphone LP Junior guitar to a six-string fretless, which sounds amazing played with an eBow as I can get synth-style pitch slides and infinite sustain, neither of which is natural for a guitar. Eventide's Physion plug-in or the EHX Attack/Decay pedal can add a slow attack to a standard guitar to produce bowed sounds, while adding a shimmer reverb produces a lovely pseudo-string sound.

Of course you can also mic an amp and get the full range of conventional guitar sounds, from clean to full-on rock filth, all with a degree of touch responsiveness that keyboard players can only dream about. If I need a new instrument or need to modify an existing one, my 'man cave' workshop adjoins my studio: already I have put together a fretless bass made from a discarded fretted bass neck and some reclaimed snooker table; a sitar guitar; and a couple of three-string cigar-box electrics, one set up for slide and one for picking. My latest purchase is the Aum Vibraslide, and it will be interesting to see where that takes me!

AD INDEX


YOUR MUSIC TAKES CONTROL
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AKAI PROFESSIONAL
Velocity