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VIRTUAL CIRCLES

Each new technology hype cycle tends to bury its predecessor. The current obsession with AI has, therefore, robbed attention from the previous next big thing.

Virtual reality has been out of the headlines for a while now. But does that mean it was a technological dead end? Or is it quietly maturing in the background, poised to return once we've tired of having ChatGPT portray family members as action figures?

The TL;DR of VR was that you'd put on a pair of ski goggles and be transported into an alternative reality. But in practice, there wasn't actually much to do in the virtual world, and the illusion often proved superficial. The idea that one day we'd all be commuting to work in Mark Zuckerberg's Metaverse now seems even more preposterous than it did first time around.

Virtual reality is thus quite similar to artificial intelligence, in the sense that it has yet to live up to the grander claims that have been made for it. But that doesn't mean both things can't be impressive and useful on a smaller scale. AI might not be building starships or unlocking the secret of free energy, but as Ben McCulloch's feature in this issue demonstrates, it's quite useful for developing plug-ins. And whilst VR isn't teleporting us to our own personal Abbey Road, it is becoming more relevant in a production context.

If reality is ever going to be virtualised convincingly, sound will have a key role to play. And since virtual reality is delivered through wearable devices, the same needs to be true of audio for VR. I've tried quite a few products that seek to manufacture a three-dimensional experience on headphones, and until now, I'd say the benefits have been real but limited. However, Sony's 360VME really does take things up another level. As it stands, it's optimised for the specific task of mixing and QC'ing immersive audio. There's no visual element, and it doesn't support technologies such as head-tracking. But Sony have solved one of the really hard problems, and 360VME is wholly convincing in the way that it presents sound as coming from above or behind you.

There are many other really hard problems to be solved before we can all step into our own virtual Studio 2, or whatever the endgame of VR in audio production is. But at least it feels as though one of the essential building blocks is now in place. And, simultaneously, we're seeing other breakthroughs, such as Neumann's VIS virtual mixing environment for the Apple Vision Pro. Who knows, perhaps soon you'll be turning the pages of a virtual SOS on a virtual newsstand. ■■■

Sam Inglis
Editor In Chief

"Whilst VR isn't teleporting us to our own personal Abbey Road, it is becoming more relevant in a production context."



SOUND ON SOUND

**ALLIA BUSINESS CENTRE
KING'S HEDGES ROAD
CAMBRIDGE
CB4 2HY**

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



WWW.SOUNDONSOUND.COM/SUBSCRIBE

WORLDWIDE EDITIONS



UK/WORLD



NORTH AMERICA

ADMINISTRATION

admin@soundonsound.com

Managing Director/Chairman Ian Gilby

Editorial Director Dave Lockwood

Marketing Director Paul Gilby

Finance Manager Keith Werthmann

PRODUCTION

graphics@soundonsound.com

Production Manager Michael Groves

Designer Mick Reilly

Designer Andy Baldwin

SUBSCRIPTIONS

subscribe@soundonsound.com

www.soundonsound.com/subscribe

Circulation Manager Luci Harper

Administrator Nathalie Balzano

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

Reviews Editor Matt Houghton

Reviews Editor Chris Korff

Production Editor Chris Korff

News Editor Luke Wood

ONLINE

support@soundonsound.com

Digital Media Director Paul Gilby

Design Andy Baldwin

Web Content Editor Callum Hall

Web Editor Adam Bull

Digital Media Manager Atheen Spencer

www.soundonsound.com

twitter.com/soundonsoundmag

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instagram.com/soundonsoundmag

ADVERTISING

adsales@soundonsound.com

UK Media Sales Manager Chris Cope

MARKETING

marketing@soundonsound.com

International Business Development

Nick Humbert

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Schoeps Desert Island Stereo Set

Capacitor Microphones

These pocket-sized mic pairs provide an easily understood entry point to Schoeps' comprehensive catalogue.

SAM INGLIS

Of all the great German microphone manufacturers, Schoeps seem to maintain the lowest profile. Most people who are serious about recording classical music will have extensive experience with Schoeps mics, but in the rock and pop world, they can fly a little under the radar. That's partly because, unlike Sennheiser, Neumann or beyerdynamic, Schoeps are not generalists. They do not make dynamic mics, or large-diaphragm models, and they don't offer a range to suit every budget. Their entire product line is based around small-diaphragm capsules, of uncompromising quality. Historically, moreover, there has been no obvious Schoeps counterpart to the U67 or MD441: no iconic, instantly identifiable object that is ever-present in studio photos or documentaries about classic albums. Apart from the vocal-oriented V4U, Schoeps mics are deliberately made to be self-effacing and visually unobtrusive.

Click & Colette

At the heart of the Schoeps range is a series of no fewer than 20 different capsules. All 20 are available separately as options for the modular Colette system, in which they can be paired with one of a number of bodies such as the CMC 1U and CMC 6, as well as accessories such as pads and low-cut filters. Almost all of the MK-series capsules are also available as

complete miniature mics, courtesy of the CCM range. So, for example, the CCM 2H miniature mic is functionally and sonically the same as the pairing of the CMC 1L body and MK2 capsule from the Colette range. You'd choose the former if you value convenience and compactness, the latter if you want the freedom to mix and match different preamps and capsules.

The Colette range is by far the most comprehensive modular mic system on the market, which is perhaps a double-edged sword. On the one hand, there's no recording task imaginable that can't be handled by at least one of the 150-plus combinations of body and capsule on offer. On the other, if you're taking your first steps into Schoeps' world, the immense wealth of options can be intimidating. I suspect that's one of the major reasons behind the introduction of the Desert Island Stereo Set.

Small Is Beautiful

If you were actually stuck on a desert island, you'd have to keep a sharp eye out

for thieving seabirds, because the Desert Island Stereo Set is remarkably compact. It comprises a pair of CMC 1U bodies, each about an inch and a half long, with your stereo-matched choice from nine of the most popular MK-series capsules. These are supplied with simple standmounting clips in a very robust black Peli case with a carabiner, so you can attach it to your belt in case of albatross attack. At around five inches by three, the case is genuinely pocket-sized, and not much larger than one you might use for a pair of IEMs or similar.

Capsule options for the Desert Island Stereo Set include four MK2 omni types variously optimised to be flat in the nearfield, the diffuse field and distances in between, plus the MK21 wide cardioid, the MK22 open cardioid, the MK4 cardioid, the MK41 supercardioid, and the MK5, which uses a mechanical switching arrangement to offer both omni and cardioid patterns. Schoeps supplied the MK4 version for this review. As you'd expect given the price and Schoeps'



reputation, manufacturing quality is top-notch. There's something awe-inspiring about tiny things made to an incredibly high standard, and the Colette mics are just that.

The CMC 1U body requires standard phantom power, with a range of 30 to 52 Volts specified, and its XLR output is electronically balanced. The CMC 1U / MK4 combo can accommodate sound pressure levels up to 135dB for 0.5% distortion, and if you anticipate recording anything louder than this, you can of course buy Schoeps' inline pads separately. The CMC 1U has a frequency response that is flat from 20Hz upwards; below this, it is intentionally designed to roll off at 12dB per octave. Output impedance is 50Ω, and Schoeps say that the CMC 1U will drive cable runs of up to 400m.

The MK4 capsule, meanwhile, boasts specifications that look almost too good to be true. Its polar pattern is pretty much perfectly cardioid between 250Hz and 16kHz, although it's a shame the graph does not include a measurement at 125Hz. Its on-axis frequency response is ruler-flat from about 180Hz upwards, below which a very gentle roll-off reaches -3dB at 50Hz. Self-noise is a state-of-the-art 14dB (A-weighted), and the system has a comfortable sensitivity of 15mV/Pa.

Pencil Pushing

Applications for a well-matched pair of high-quality cardioid pencil mics are legion, and if there's one to which the CMC 1U / MK4 combo isn't well suited, I didn't find it during the review period. The compactness of the body is a real help when setting up something like an ORTF array; longer mics

such as the old AKG C451 have to be offset vertically to stop the tails and cables fouling each other, but there are no such issues here (although setting a precise mutual angle by eye can be harder with such short bodies). ORTF and some other widely used stereo arrays also demand a good off-axis response, since things that are in the centre of the stereo field are off-axis to both mics. Schoeps' capsules deliver exactly what's needed in this respect.

There are also numerous studio applications for high-quality cardioid pencil mics that don't involve stereo, and these mics fulfil such roles with equal aplomb. It seems to be the fate of all small-diaphragm cardioid mics to be compared with the Neumann KM84,

which has become the gold standard in many engineers' minds. I carried out several such comparisons during the review period. On some sources, such as close-miked vocals, the two were largely indistinguishable, except in that the KM84 seemed more susceptible to popping. On others, notably acoustic guitar, there was a discernible difference, albeit one that was subtle and hard to describe. In most cases, the Neumann classic seemed to exhibit a slightly fuller low midrange, while the Schoeps mics felt more open and extended at the high end: not brighter, as such, but perhaps more natural.

Another favourite German small-diaphragm cardioid mic of mine is the Sennheiser MKH40, now

discontinued in favour of the MKH8040, but still very popular. In a direct comparison, the Sennheiser mic was significantly warmer-sounding on all sources, and less 'sparkly' in the 10kHz region. And in each case my instincts told me to EQ the MKH40 to sound like the Schoeps, not the other way around!

The CMC 1U / MK4 pairing is undoubtedly much more similar in character to the KM84 than it is to the MKH40, and I think both the Schoeps and the Neumann mics manage to achieve the neat trick of measuring completely flat whilst sounding just a touch euphonic. It's not the outright flattery that many large-diaphragm mics provide (and which has to be weighed against their

lumpy off-axis response and other negatives), but a sort of twinkle in the eye that you wouldn't get from, say, a typical measurement mic.

All in all, I was left thinking that the KM84's legendary status is as much

an accident of history as it is a reflection of that mic's undoubted quality. The prominence of the Neumann name, and the KM84's widespread use on classic records, has given it a profile that eludes the more understated Schoeps line. Yet the MK4 capsule has an equally rich history and pedigree. In combination with the CMC 1U or one of Schoeps' other bodies, it achieves technical standards that the older mic can't. And it sounds great, both in stereo and on its own. If engineers decide to elect this as the gold standard instead, I won't stand in their way. ■■■

£ £2795, or £3355 with MK5 capsules.
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“There's something awe-inspiring about tiny things made to an incredibly high standard, and the Colette mics are just that.”



Neve 88R LBC

500-series VCA Compressor



At last, the channel compressor of Neve's flagship 88R console is available in 500-series form!

MATT HOUGHTON

Reviewing Neve gear is always a pleasure, but as they helpfully make various preamps, EQs and processors in multiple forms for different use cases, there's often considerable overlap between products. That leaves reviewers with a conundrum: just what is there left to say? The 88R LBC puts the 88R console's channel compressor into a 500-series module, and is a case in point. Neve have released standalone versions of this processor before, most recently the 88C that I reviewed in *SOS* May 2025 (<https://sosm.ag/neve-88c>). So, after confirming to you that the 88R LBC functions in precisely the way that's claimed, works well with a vast range of sources, and sounds as good as you'd expect, I find myself at serious risk of repeating myself (see what I did there?). Thus, I'll keep this review of what I judge to be a remarkably good compressor short and sweet!

Attack With Speed!

This single-slot, 500-series module offers pretty much identical functionality to the dual-channel 88C,

but aesthetically the knobs and layout owe more to the 88R console. At the top, a nine-LED meter can indicate up to 50dB of gain reduction (with finer resolution in its top half, naturally). To each side there's a tri-colour LED to indicate the input/output levels. That's slightly less visual feedback than you get on the 88C, but a design decision that contributes to a clean, open layout, despite the tight confines of the 500-series format.

Below are pots for the threshold (+20 to -30 dB), ratio (1:1 all the way up to Lim, with 3:1 at the vertical position), release (0.03 to 3 ms), and make-up gain (0 to 30 dB). These are joined by four toggle switches. One at the top right engages a side-chain high-pass filter (80, 125 or 300 Hz, or off), while another engages an automatic make-up gain facility — a time-saver we're accustomed to in plug-ins, but rare in the analogue world, where it's rather harder and more expensive to implement.

A third switch enacts the automatic release mode. Yes, many compressors have such a facility, whereby dual time-constants allow it to recover faster after aggressive transients and more gently at other times, but Neve's approach is notable for employing three 'layers', which tends to make it effective on a wider range of material than most; in fact, I almost always agreed with Neve's 'set and forget' suggestion. The final toggle switch, Fast, implements an ultra-fast attack setting, and for the Lim ratio this doesn't let anything whatsoever peep above the threshold.

Now, the eagle-eyed among you may have noticed that I haven't mentioned an attack knob. That's because there isn't one! Instead, Neve use something they call "adaptive attack technology". Conceptually, it's similar to an automatic release: the use of two separate circuits enables the compressor to respond at different speeds to more aggressive and gentle transients. I wrote about this in my 88C review. In short, my initial reservations about 'letting go' were unfounded; it works brilliantly.

Finally, at the bottom, we have push switches for In (bypass) and Link. The latter allows you, assuming you have a suitable 500-series host rack, to link the side-chain bus of multiple 88R LBC modules, for use on stereo and other multi-channel material.

Key Takeaways

In terms of its action and sound, there's really very little I can add to my already glowing assessment of the 88C but, in summary, the technical specs are everything you'd hope for from a modern Neve design, with vanishingly low distortion that translates into a superb sonic performance. The adaptive attack and 'triple-layer' auto release work very well, and can make setup for most sources exceptionally quick and easy. Were you to have several of these compressors in a rack or 500-series host console (in which scenario there should be no power problems, since the modest current draw is well within the VPR Alliance spec), that speed could prove a real advantage during a busy tracking or analogue mix session. Neve have done a grand job of packing all this into the 500-series format too, with an intuitive layout that leaves plenty of space for fingers around the controls.

Yes, you pay a little premium for the Neve name, but it should hold its value well too, and that money is genuinely buying you a great-sounding, 'all-rounder' channel compressor with some really cool and practical features. **///**

summary

A lovely, clean-sounding VCA compressor-limiter whose unique features make it versatile and exceptionally quick to set up.

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JOHN WALDEN

AutoTune Pro 11 (reviewed in SOS June 2025: <https://sosm.ag/antares-autotune-pro-11>) remains Antares' flagship product, offering both graphical/manual pitch correction and harmony generation alongside automatic pitch correction. But they've long offered more affordable versions that concentrate on the automatic correction side of things, and replacing AutoTune Artist is the new AutoTune 2026. The most obvious change is a GUI redesign, though the controls and settings are largely familiar from earlier versions.

Simplicity

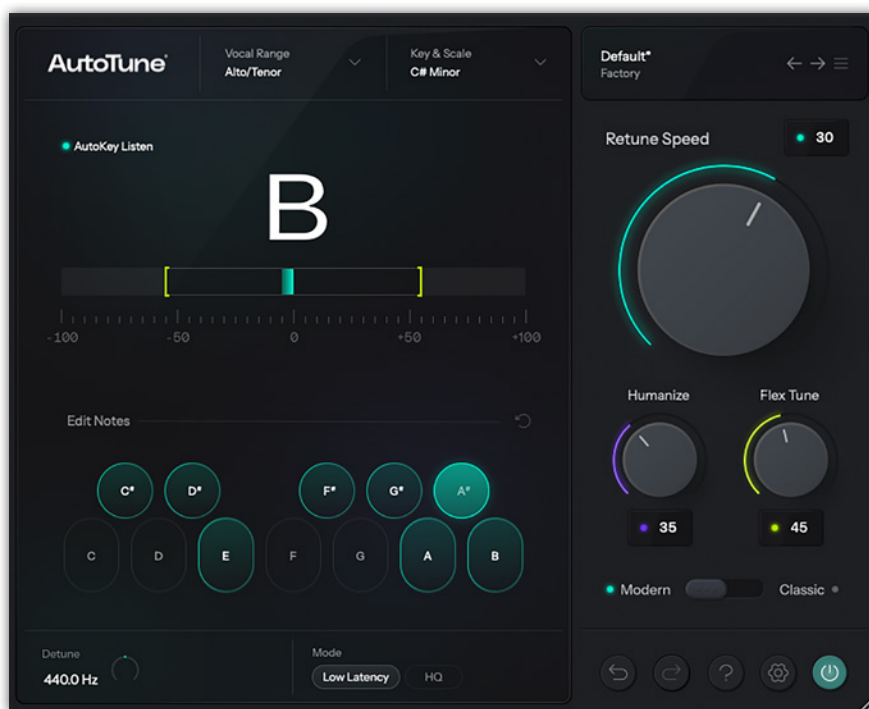
You can pick from two modes, Modern and Classic, the former promising more natural results and the latter offering the older AutoTune 5 algorithm, which apparently remains popular. Other than that, the main control set has been streamlined to just three parameters: Speed, Humanize and Flex Tune.

The retuning Speed should be self-explanatory, while Humanize allows AutoTune to pitch-correct sustained notes more naturally at higher Retune speeds. FlexTune (available only in Modern mode) constrains the amount of pitch correction being applied (something that's nicely indicated by coloured brackets in a redesigned pitch-correction meter), so more of the original vocal gestures can be retained. You can set the voice or instrument type (for example, soprano, alto/tenor or baritone/bass) and the key and scale type manually, but AutoTune can Auto Detect the voice type and, via the included AutoKey plug-in, suggest the appropriate key/scale combination after listening to your audio input. A drop-down list offers more scale options, and the new GUI makes it easy to add/remove notes to constrain the notes being corrected.

With so few controls, dialling in your pitch correction really is very easy, but those new to AutoTune might find it useful to explore the new collection of artist-designed presets.

On Test

AutoTune 2026 is, frankly, a breeze to use, and does its job with a minimum of fuss. The highest-quality setting delivers the best results at the cost of 58ms of latency, but as with AutoTune Artist, there's the choice of low-latency



Antares AutoTune 2026

Pitch-correction Plug-in

Whether you want natural pitch correction or the classic pitch-glitch effect, AutoTune 2026 aims to make things easy.

operation (Cubase reported 2.3ms in Modern mode and 0.8ms in Classic mode). So you could easily use this plug-in in real time while tracking, if that helps your singer to perform with confidence. I ran a few different vocals through AutoTune 2026 and AutoTune Artist, and in most cases I'm not sure I would have been able to reliably tell the difference in a blind test. But since AutoTune 2026 uses fewer CPU cycles to match this already high quality bar, that's a win!

On the down side, the Artist version's Transpose and Throat/Formant options aren't included, and neither are some of its more exotic scale selections. Also, some might be deterred by the cost of the perpetual licence, though there are subscription options and a free 14-day trial is available. But the bottom line is that AutoTune 2026 works well.

It captures the essence of the AutoTune automatic pitch-correction experience in a modern, very accessible GUI. Whether you want to gently nudge the pitching of a decent sung vocal or make a feature out of more extreme pitch correction, those new to automatic pitch correction will find that AutoTune 2026 does exactly what they will expect of an 'industry standard'. **///**

summary

AutoTune 2026 delivers Antares' famed automatic pitch-correction process in a modern UI with a streamlined workflow, although, for a perpetual licence, that quality and convenience does come at a price.

£ Perpetual license £282 (includes one-month subscriptions to Auto-Tune Unlimited and AutoKey 2). Subscriptions also available. Price includes VAT.

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Throne Of Tone

Dual Overdrive Pedal

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PAUL WHITE

Warm Audio make a number of pedals based on existing or classic circuit designs, and they've generally impressed. As the name of this one might suggest, the new Throne Of Tone owes its inspiration to the Analogman King Of Tone, a somewhat boutique dual overdrive pedal, whose designer cites Marshall's Bluesbreaker pedal as his own inspiration. The original King Of Tone allowed for an extra degree of user tweakability compared with the Bluesbreaker, though, courtesy of internal switches and trimmer pots — and the Throne Of Tone helpfully puts all of these on the outside to improve access. All but one of these controls are on its top panel. The exception is the high-voltage switch, which is on the rear panel, and when active this doubles the internal voltage from 9 to 18 Volts to allow more internal headroom — something that's particularly useful when using the clean boost setting, since that too breaks up eventually if the gain is set high.

Game Of Tones

Both sides of this pedal are identical, with rotary controls for Volume, Gain, Tone and Presence. The last of those shifts an internal filter from 500Hz to 2.3kHz to add extra brightness and bite. Below the knobs are three switches, the first of which changes the voicing from King to Blues. Both produce perfectly decent tones, but Blues is the one that offers that bit more bite and clarity. Apparently, the Blues setting uses TL072 op-amps and is voiced to emulate the Bluesbreaker pedal, whereas King uses JRC4580 op-amps and is voiced to emulate the King Of Tone.

Next comes a three-position switch for selecting Distortion, Boost or Overdrive. Boost provides a nominally clean boost, while Overdrive handles lower-gain drive sounds. Distortion delivers more dirt and saturation of course, but it still never gets excessive. Warm Audio have also included insert send and return jacks, so that other effects can be placed between the two drive channels where desired. A graphic EQ could be a very interesting addition, for example.

These modes are created by changing the clipping topography of the circuitry, which uses two op-amp stages per channel. Clipping is produced by diodes in the feedback path of the second op-amp stage, and also by separate diodes to ground following that stage; switching allows the two sets of diodes to be taken in or out of circuit. The first

op-amp stage provides the gain, and the tonal shaping is handled by a passive filter network at the end of the signal chain. A third toggle switch selects between Low or High gain modes. A third toggle switch between the two channels allows their order to be switched, and each channel has its own footswitch with red status LED.

Diode In Your Arms Tonight

It's common practice for guitar players to stack different overdrives, and having essentially two independent drive circuits in one pedal makes this very easy. The way the two channels interact allows for some very expressive drive characters, one option being to push a clean boost into an overdrive, though it can also be rewarding to stack two lower-gain overdrives as an alternative to a single drive pedal set to a higher gain. As each pedal has its own bypass switch, the player can choose between three different drive settings during performance by using either channel on its own or both in combination.

Overall, the natural character of the guitar and amplifier still come over strongly, while the range of drive goes from just adding a little hair up to classic rock, but not as far as modern metal. Coaxing your own optimal tone out of the pedal can take a little experimentation, as the ways in which the two channels can be set up and combined provide a lot of flexibility, but having all the switches where you can get at them makes life much easier than having to go inside the case for internal settings.

As is so often the case with highly regarded pedals, the original King Of Tone has an associated mythology regarding its 'rare as unicorn droppings' diodes and their influence on the sound. While I have no way of knowing whether the same diode types or close equivalents are used here, I definitely have no argument with the clipping behaviour of this circuitry.

OK, this pedal may be viewed by some as a clone of tone, but it's affordable, versatile and very solidly built. If having different shades of overdrive in one pedal appeals to you, then the Throne Of Tone may well scratch your itch. **///**



summary

More than a mere clone, the Throne Of Tone offers takes on both the Bluesbreaker and King Of Tone pedals — and puts all the tweaking options on the outside where you can reach them!

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CHRIS KORFF

HEDD Audio are best known for making studio monitors fitted with 'folded ribbon' tweeters, or Air Motion Transformers. Indeed, their founder Klaus Heinz has spent his life developing this technology — first studying under the inventor of the AMT, Dr Oskar Heil, before founding ADAM Audio and developing the idea into the ART (Accelerated Ribbon Technology), and then setting up Heinz Electrodynamic Design (HEDD) to take his ideas even further. Arguably, Klaus Heinz is responsible for more ribbon tweeters gracing studios than anyone else.

So it perhaps wasn't surprising that HEDD Audio's debut pair of headphones used a heavily modified AMT driver. Indeed, the HEDDphones were the first ever headphone design to employ AMT drivers, and also the first product to deploy AMTs across the full audio bandwidth — an impressive feat of engineering given that they're normally designed to only reproduce frequencies above 2kHz or so.

Those headphones were expensive and enormous, but when Sam Inglis reviewed the original HEDDphones in *SOS* July 2020, he deemed those trade-offs more than worthwhile, praising them for their neutral, forensic sound, and their outstanding ability to resolve detail and reveal non-linearities in source material.

The HEDDphone Two, released in 2023, met with similar praise, but while it was a worthy update to the original design it was still fundamentally the same concept: an AMT driver operating at full range as a headphone driver. For their third headphone release, though, HEDD have made a significant departure...

Mortal Coil

The new HEDDphone D1s use moving-coil dynamic drivers, and they are consequently a much more familiar shape and size. They're also significantly more affordable than the HEDDphone Twos — around a grand cheaper, in fact. Even so, they are not as conventional as they might appear: HEDD say that these are the first ever headphones to employ a Thin-Ply Carbon Diaphragm (TPCD), a material developed by Swedish company Composite Sound. As far as I'm aware, this partnership marks another first for HEDD, who in addition to making AMT tweeters, have always manufactured the moving-coil bass and midrange units for their own monitors — and even developed their own proprietary materials for the

cones. So why get a third party involved?

Composite Sound are a subsidiary of Oxeon, who make carbon-fibre structures for the automotive and aerospace industries, among others. Perhaps surprisingly, their name has appeared in *SOS* before: Oxeon make a carbon-fibre material called TeXtreme, which high-end monitor manufacturers Ex Machina Soundworks employ for the midrange cones in all of their monitors. And Composite Sound, formed in 2016, seem to have been founded as

a spin-off brand specifically to serve loudspeaker (and now, apparently, headphone) manufacturers.

The Thin-Ply Carbon material itself is woven from extremely thin 'tapes' of carbon, which Composite Sound say results in increased stiffness and reduced weight compared to conventional 'yarn'-based carbon fabrics. More impressively, they say that their manufacturing process allows them to vary the thickness and stiffness of the material in a very targeted way, in order to (for example) suppress resonances, or delay the onset of cone break-up. So, rather than following the traditional approach of creating a cone and then damping it where necessary (adding mass in the process and therefore reducing the speed at which it can move), they can incorporate only such damping as is needed by varying the properties of the diaphragm itself. Composite Sound call it a 'Metamodal' material, and say that it can be "engineered for a tailored modal behaviour —



HEDD Audio

HEDDphone D1

Open-back Headphones

HEDD's third headphone release loses the AMT driver in favour of a conventional moving coil, but the company's trademark high-tech approach is still very much evident.

frequency response linearity, improved pistonic motion, moving mass optimisation — as needed”. All of which are very useful qualities for loudspeaker and headphone driver diaphragms.

There is a lot of dogma in the audio industry, and companies that specialise in specific technologies — electrostatic speakers, say, or compression drivers — are often stubbornly wedded to them. HEDD Audio, being one of the leading exponents of AMT tweeters, and having pioneered their use in headphones, could be forgiven for sticking with them. But the decision to work with Composite Sound looks like a very pragmatic one, if you assume that they first decided they’d like to make some less enormous headphones, concluded that they’d therefore need to be moving-coil designs, decided that they should still be extremely good headphones, and then rang around to see who could make the best possible material for their diaphragms. That strikes me as a commendably rational approach. And it should be noted that HEDD haven’t entirely outsourced the tech for the D1s: other than the TPCD, the rest of the driver is custom-made in-house at their factory in Berlin, along with the rest of the headphones.

There’s more to headphones than their drivers though. In the case of the D1s, those TPCD drivers are housed in open-back circumaural earcups, which feature generous padding to place the drivers just over an inch away from your earholes. The drivers are protected on both sides by perforated metal grilles, and

on a B&K 5128 HF-HATS rig, with diffuse-field HRTF compensation (ie. corrected for the effect that the B&K crash-test dummy’s anatomy has on the measurement microphones). The graph shows a largely flat response between 20Hz and 10kHz, bar a broad boost around 100Hz and some wrinkles from 3kHz up, before falling to about -20dB at 20kHz.

In Your HEDD

All of which is very well, but what are they like to use? The first thing I noticed when putting them on, before playing anything through them, is that they’re quite ‘huggy’ — somewhere between ‘snug’ and ‘tight’ — and the contouring of the headband wasn’t quite a match for the contouring of my own head, which meant that the bumps applied a noticeable amount of pressure on either side of my pate. On the plus

In fact, tonality-wise the HEDDphone D1s are impressively neutral, with no part of the spectrum sounding over- or under-represented. The big deal that HEDD and Composite Sound make of the low mass of their TPCD led me to wonder whether the high end might prove a bit attention-grabbing, but in practice, there’s nothing incongruous about the top end (unlike the K712s, which are a little on the hypey side). If I did notice a fast transient response, it was in the lower mids, where double basses and toms seemed rather more tangible and punchy than on both my K712s and the much more expensive K812s. And on material recorded in stereo, I also got a sense that the image was more precise than on the AKG models, perhaps because of the TPCD’s superior ability to resolve timing information between the left and right channels.

In fact, the AKG K872 provides an interesting point of comparison. Although it’s a closed-back design, it boasts highly engineered dynamic drivers, with diaphragms of 53mm in diameter. HEDD don’t specify the size of the TPCD, but it looks quite similarly sized to me. What’s more, both designs place the diaphragms a similar distance from the wearer’s ears. To my ears, both models sound quite similar (notwithstanding the closed-back design of the 872s, and their concomitant excellent isolation and low-mid emphasis), but in terms of neutrality and transparency, I think the D1s win out. Which is no small feat given that they cost around half the price that the K872s did when new.

Finally, and it’s quite a difficult thing to notice when trying out new headphones, the D1s actually sound... slightly boring. Not in a dull or muffled way, but in the way that a good pair of studio monitors in a properly treated room sound. I found myself listening to the music, rather than the headphones. More so than any of the other headphones I had to hand during this review, the D1s didn’t really have a ‘sound’ as such, they just presented audio in a neutral, natural and transparent way. Which is all anyone can ask of a good pair of studio headphones! **///**

summary

HEDD’s first-ever moving-coil headphones are a fine example of quality engineering translating to excellent sound reproduction. They’re not cheap, but they easily hold their own against other premium models.

£ £599 including VAT.

W hedd.audio

“The HEDDphone D1s are impressively neutral, with no part of the spectrum sounding over- or under-represented.”

the earcups are attached to the headband by way of a yoke that both swivels on the vertical axis and rotates along the horizontal. There’s not enough movement to allow the D1s to fold up, but they do come with a rigid zip case, which can also house the included braided Y-cable. The Y ends of said cable have an L and R on the mini-jacks to let you know which is which, and you need to observe those markings if you don’t want a reversed stereo image (the earcups don’t do the clever thing of only outputting the correct side no matter which cable you plug in, as some headphones do). There’s a corresponding L and R on the inside of the headband, which has sliding metal strips to allow for differently sized heads, and is well padded at the top. The padding on both the headband and earcups is covered with perforated, breathable velour, which is comfy to both ears and a bald head.

Specs-wise, the D1s are again much more conventional than the AMT-based HEDDphones. They have a sensitivity of 100dB/W (compared with 89dB/W for the HEDDphone Two), and a nominally 32Ω impedance (41Ω for the HEDDphone Two). HEDD give an unqualified (and therefore not very useful) frequency response of 5Hz-40kHz, but they also show a proper and much more helpful response graph, as measured

side, though, they felt very secure and I was confident that the fit, in terms of the position of the drivers relative to my ears, was spot-on and repeatable. I can’t say the same for my usual open-backs, the AKG K712s, which are very loose-fitting and allow for rather too much movement. What’s more, and I guess as a result of both the snug fit and generously padded earcups, the D1s offer more isolation than you’d expect from open-back headphones. Sound leakage outwards is also considerably better than on the 712s, which you can hear remarkably clearly from across the room. However, the D1s don’t exhibit that slightly claustrophobic or tubby sound that closed-back cans usually do, to varying degrees.

MATT HOUGHTON

With a background in aviation electronics, Pete Hamstead came to the guitar world with different ideas from most designers, but arguably his most important was to design analogue amps that boast much better noise performance than the classics, and his recipe works: I've played through a couple of their amps and as well as kicking out a great tone, they're whisper-quiet in terms of noise — to the extent that when not playing through it, you really wouldn't know the amp was on!

Red Lines

This is all achieved through component choice and skilled circuit design (as opposed, for example, to any noise-suppression processing such as expansion). Naturally, Hamstead soon went on to bring his expertise to bear on analogue stompbox design, and we've already been impressed by a couple of his designs: the Signature Analogue Tremolo (reviewed in SOS July 2016: <https://sosm.ag/hamstead-signature-tremolo>) and the Odyssey multi-mode analogue overdrive (SOS August 2018: <https://sosm.ag/hamstead-odyssey>).

Now, in collaboration with the team behind *That Pedal Show*, Hamstead have returned with a larger, more ambitious pedal called the Redwing. But although billed as an analogue 'stereo modulation pedal', that description doesn't really convey its full scope. It features two bucket-brigade delay (BBD) lines that are used in various ways to bring us a whole smörgåsbord of tasty effects, ranging from chorus, vibrato and flanging, to a sort of rotary-speaker emulation and even ring-mod-like textures.

As you might expect of Hamstead, they've taken an analogue purist's approach: the signal path and the controls are entirely analogue, and that's a decision that has pros and cons. In the pros column, the sound is great, the technical performance is superb, with vanishingly low noise levels, and the controls are every bit as immediate and intuitive as you'd expect of a decent analogue pedal. As for the cons, there's no MIDI, no tempo sync, and no preset switching system. For some, that last one could well prove a deal-breaker, simply because this pedal is capable of performing so many different duties,



Hamstead Soundworks

Redwing Stereo Modulation Pedal

This all-analogue affair takes modulation well beyond the bread-and-butter chorus and flanger effects.

and for playing a set on stage I'd tend to view it as a single-purpose pedal, even if that purpose might vary from show to show. The sound and versatility definitely appeal, though, and at home and in the studio, you obviously have more time to dial in your settings and experiment.

Form & Function

The build quality is solid and weighty; it inspires confidence. On/bypass switching is handled using GigRig's 'silent' OptoKick optical footswitch, which

should prove reliable and, as long as you're gentle, can be operated quietly, with no click. The main knobs on the top panel are Gain (which appears to be a post-effect master volume, and doesn't apply when the effect's bypassed), Speed, Width and delay time (labelled Manual), pretty much as you'd expect. There's also a Blend knob for mixing the wet and dry signals, and Regen, which sets the feedback intensity. An active EQ section comprises separate Bass and Treble knobs that can cut or boost, but apply only to the wet signal — so, in conjunction with the Blend control, you can use them to rein in any effects that you like but which nonetheless tend to leap out a bit too much.

Some switches, though, allow you to take things a little further. There are three LFO modulation waveform options (Triangle, Sine, and Asymmetric, which is essentially a sine wave that's flattened out

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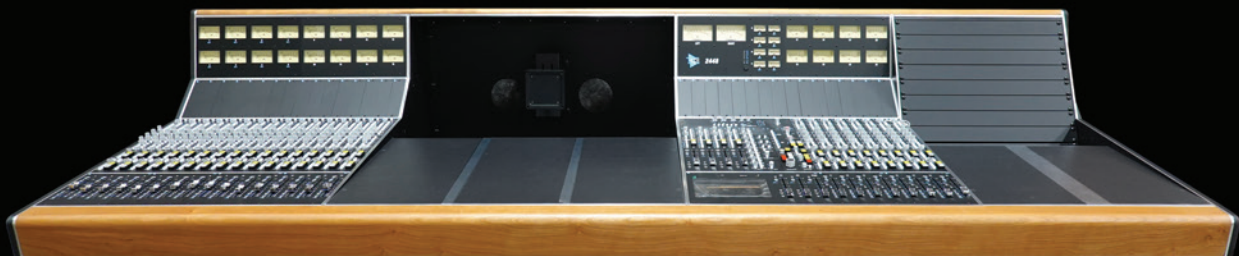
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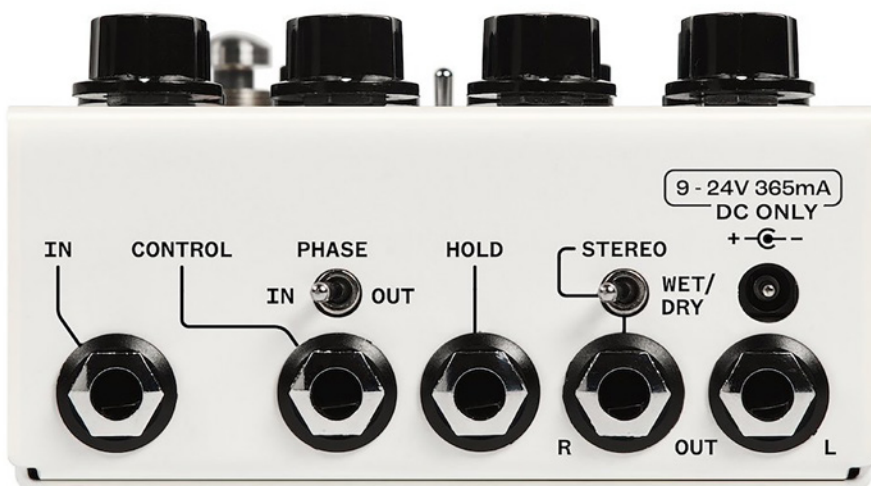
» on the top part of the wave, to cause a 'lurchier' sort of feel), along with a smaller knurled knob to adjust the Shape. With the latter you can move the centre point of the waveform back/forwards, so you can, for example, dial in a classic sawtooth shape. The LFO is Hamstead's own design by the way, not an off-the-shelf chip.

The Speed knob has an associated switch for Hi/Lo speed ranges. This increases the range of effects possible without the pot seeming too sensitive and finicky to the user, and the Hi setting is what allows this pedal to be taken all the way into Leslie and ring-mod sort of territory. Finally, an Offset switch and another small knob allow you to offset the right-side delay line, making it possible to achieve something like the classic through-zero open-reel tape flanging. Obviously, you can't move either delay line earlier in time than the dry signal!

On the rear, the Redwing has a single mono jack input, but the dual jack outputs can be used in a couple of ways. A switch determines whether they're used in tandem for left-right stereo, or one for the dry signal and the other for the wet. There's also a Control input for an expression pedal that acts on the Speed parameter. A Hold jack supports an external footswitch, and this can be used to drive the feedback loop into oscillation. A Phase In/Out toggle inverts the polarity of one of the two delay lines, for more complex, spacious-sounding effects. Finally, a DC power inlet accepts 9-24 V, and requires a 365mA supply (not provided).

Impressions

Plugging into the Redwing, the first thing I noticed was a joyful sonic richness, and in general it sounds every bit as good as I'd hoped. Obviously, you do need to spend a little time dialling in the actual effect you want, and there are some recall sheet-style examples in the manual to help get started. The dual BBD lines open up possibilities far beyond what a standard mono modulation pedal can deliver and since, in stereo mode, you can offset one channel and polarity-invert



the modulating LFO for one line, you're able to produce all sorts of luscious textures.

The Redwing really shines when used in fully stereo setups. I don't tend to run dual amps or cabs myself, but when testing it over my studio monitors with modelling amps, the sense of width and motion was impressive: a constantly shifting, luxurious image that feels warm and immersive, but simultaneously clear — never noisy or muddy. It has serious potential, then, for keyboards, synths or, perhaps, even vocals, not just guitar. But for all this glorious stereo sophistication, the Redwing is perfectly happy being used as a mono-in, mono-out stompbox on your guitar pedalboard.

I can't claim to be a frequent user of chorus effects either, but I admired and enjoyed what I found here; even subtle settings seem to have a sense of depth and vitality. Flanging is much more 'me' territory, and when it comes to that effect this pedal is oh-so satisfying. The Redwing is capable of restraint or wildness, and pretty much everything in

■ You have the choice of fully mono operation and not only a stereo output, but separate wet and dry mono outputs.

between. You can dial in familiar effects — the gentle 'whoosh' of an Electric Mistress, say — or something far more distinctive. That includes, with the Offset control engaged, what I reckon is a convincing approximation of tape-style flanging. What's more, since the Gain control only applies when the effect's engaged, you can compensate for any sense that a flanging effect is sucking your part out of the mix when you engage the pedal — an occupational hazard with modulation effects that can render otherwise brilliant effects unusable!

I found the vibratos equally impressive. And if you push Regen into higher values and set the Speed switch to Hi, the pedal begins to veer into much more experimental territory: those metallic, ring-mod-like sounds I mentioned above, and self-oscillation — the latter can be really engaging.

Verdict

So, do I love the Redwing? Absolutely! Should you get one? That depends on what you require of it. If you need a Swiss Army knife of a modulation pedal to experiment and record with, and the sound and range of effects is all important, it'd be a superb choice; it may not be inexpensive, but I really can't think of a better option, and I'd heartily recommend it. If, on the other hand, you need a stompbox that you can quickly configure differently for each song in a live set, then, gorgeous as it sounds, you should probably look elsewhere. ■■■

summary

An immaculately designed and built all-analogue pedal, the Redwing is capable of a surprising range of modulation effects, from classic choruses, through whooshing flangers, to swirling rotary speakers and metallic ringing. As long as you don't require instant preset recall or MIDI control, it's a joy to hear and use.

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“The sense of width and motion was impressive: a constantly shifting, luxurious image that feels warm and immersive, but simultaneously clear...”

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ACS Eclipse 500 In-ear Monitors

IEM pioneers ACS bring their bassist-optimised hybrid technology to the universal-fit market.

DAVE LOCKWOOD

Regular readers may recall, last year I found a set of custom-moulded ACS Emotion Dynamic hybrid IEMs to be the perfect solution to the challenge of playing bass in a silent-stage setting. These just-launched models employed a setup of twin balanced armatures for midrange and highs, plus a balanced-armature super-tweeter, combined with a custom-made dynamic driver to give the impression of more 'weight' in the lows courtesy of the dynamic driver's ability to 'move more air' than balanced armatures. Of course, given that we are talking about a dynamic driver small enough to fit inside an IEM along with all the other drivers, the vent tunnels and tuning filters, we are not talking about moving very much air... But a little goes along way when it is coupled directly to your ear canal!

They worked very well in the specific application for which I acquired them, and having got used to them, I am not sure that I don't now prefer them to my existing, multi-balanced-armature set, at least for live-performance applications. At £895 a set, though, clearly these

are not within easy reach for everyone. However, with the release of the new Eclipse 500 model, ACS have brought the same technology to the universal-fit IEM market at a lower price point.

A Winning Combination

The Eclipse 500 has the exact same combination of dynamic and balanced-armature drivers as the Emotion Dynamic hybrids, but now mated with a choice of three sizes of foam tips. Achieving a proper fit is all-important for getting a good low-frequency response with IEMs, and getting the correct size of tip and, crucially, positioning it optimally is obviously the first essential in all IEM usage. But the shape of ears does vary significantly from one individual to another, resulting in some people struggling to achieve both a stable fit of the body of the IEM in the outer ear and a good coupling to their ear canal at the same time. I'm actually one of those people, having a slightly unusual ear canal size and shape, and have sometimes struggled with the positioning of universal sets.

Included with the Eclipse 500s, however, is a choice of different lengths of the coupling nozzle that mates the foam

tips to the body of the IEM. The difference is barely discernible to the eye, with the largest being only 1.5mm longer than the shortest, but it will often be enough to make all the difference. I'm not aware of anyone else offering this particular mode of customisation, but it is a very good idea, and should ensure that more people are able to achieve the proper fit and positioning needed to get the full performance out of these IEMs.

So, what do they sound like? In addition to the standard foam tips, the review models were supplied with a pair of my personal custom mould tips as well so I could make a direct A/B comparison — well, as close to A/B as you can get given the time it takes to manoeuvre a pair of IEMs out your ears and insert another set! To me, they sound substantially the same when using the custom moulds and almost indistinguishable with the foam tips too, once I had found the optimum set and the ideal nozzle adaptor length. Clearly, there are more potential user variables with the latter setup than custom moulds.

The Eclipse design differs from the Emotion model in construction, having an aluminium body versus the all-silicone body of the custom-mould version,

and the sound-bore arrangement is slightly different, too: the custom set has the dynamic driver and an ambient filter combined into a single bore while a second conveys just the balanced armatures, with both bores then venting directly into the ear canal. In the Eclipse version, the dynamic driver, balanced

“Properly fitted, the Eclipse 500 hybrid set can offer much the same functional benefit as the Emotion set...”

armatures and ambient filter all have separate bores, which then combine into a single outlet via both the custom and foam tips. In theory, any of that might be expected to make some sort of difference, but I’m happy enough to say that, properly fitted, the Eclipse 500 hybrid set can offer much the same functional benefit as the Emotion set for

any performer needing a bit more sense of depth and weight in their IEM mix. And, crucially, the Eclipse includes the same ambient filter as the upmarket model, maintaining airflow into and out of the ear, whilst still achieving the required degree of attenuation and a good low-frequency performance. I really don’t know how they make that work!

IEMs For All

There is no denying that IEMs have changed the landscape of live music performance in a lasting way. Some people love them, some people hate them, while others just accept them as a necessary ‘part of the job’ whenever they are required. But they are certainly no longer just the province of the fully ‘pro’ performer. With ever increasing numbers of ‘silent stages’ or ‘low-volume venues’, if you work in any kind of ‘paid to play’ situation, sooner or later you are quite likely to really wish you had some.

One last thought occurs to me on these: custom moulds may be the pinnacle of the ‘pro’ world for optimum

fit and performance, but ears change as we get older, and sooner or later, those pricy custom moulds will no longer seal properly and have to be replaced. The serious semi-pro or amateur performer using universals, however, may be able keep a universal set in use for far longer by periodically switching to a different tip size, or, in this range, using a different nozzle length. A price tag of £699 is still not in everybody’s ‘affordable’ bracket, but with these at least you can look at it as a long-term investment in both your contemporary stage-readiness and your hearing health, too. **///**

summary

ACS’s Eclipse 500 hybrid model offers the same benefits as their upmarket Emotion Dynamic, at a lower price point, by incorporating essentially the same design and driver line-up into a universal-fit IEM.

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HLabs EQ169+

Single-channel Analogue Equaliser



Inspired by Studer's classic 169 console, this handmade EQ is just as suited to mixing work as it is to tracking and mastering.

MATT HOUGHTON

Back in March last year, I reviewed HLabs' CS169 (<https://sosm.ag/hlabs-cs169>), which is an all-analogue channel strip based on the classic Studer 169 mixing console. I enjoyed it very much. The EQ169+ is a standalone EQ that draws on the same heritage and, as with all HLabs' products, is entirely handmade at the company's facility near Lyon, France. The company already offer a less costly EQ that puts two channels of the CS169's EQ section in the same box, but the EQ169+ is a mono device that boasts not only more bands but some clever features that I'll explore in detail below.

Controls & Features

The EQ169+ is a single-channel device that's housed in a 1U 19-inch rackmountable case. It has balanced XLR inputs and outputs (the maximum input and output levels are each +22dBu) and an IEC14 power inlet on the back panel, along with a ground-lift switch. On the inside, you'll find circuit boards populated entirely with through-hole components. There's nothing inherently wrong with surface-mount devices, but this traditional approach bodes well for maintenance and repair on site (should that ever be necessary; the build quality inspires confidence!).

On the front, the yellow and white legending contrasts pleasingly with the panel's dark blue-grey, and it all looks

very stylish. You could argue that some of the labels are a bit on the small side, but they're cleanly printed and this, along with the high-contrast colour scheme, makes everything perfectly easy to read. Colour is used to differentiate the controls. The band gains all have yellow caps, with black lines aligned to an integrated pointer. As with other knobs, a line down their side makes it easy to discern the setting from any viewing angle. The frequency selectors have white caps, again with black lines. The only departure from the 'white for frequency, yellow for gain' scheme (I know

There's a reassuringly chunky power on/off toggle on the right of the front panel, as well as a smaller bypass toggle, and I was pleased to see the power switch protected by two solid-steel cheeks that protrude from the front to create a visually striking feature.

In terms of frequencies, the low shelf has a 60/20 Hz frequency switch, while the options for the high shelf are a conventional 20 and a rather airier 35 kHz. Despite those particularly high and low frequency options, the shelves' gentle slopes reach well up/down into the audible range, where they can interact usefully with the other bands to fine-tune the overall EQ curve. The high-pass filter can be set to off or 12, 16, 20, 25, 31, 42, 55, 78, 110 or 160 Hz, making it pretty

"My ears detected an almost indefinable hint of silky smoothness: while this EQ sounds clean, it never struck me as sounding harsh or clinical."

not why) is the master output level control, on the far right, which has a white cap.

The control layout makes good, intuitive sense, with the knobs for two Presence bands flanked by both high and low shelving bands, and with an additional high-pass filter on the far left.

These knobs operate 11-position switches and that, in combination with the tolerances of the chosen resistors and capacitors, means you have the precision needed for accurate recall and for channel matching of multiple units; HLabs tell us that different units are gain-matched to be within $\pm 0.1\text{dB}$.

versatile. The Low Presence (low-mid) band offers 150, 200, 300, 400, 500, and 700 Hz, and 1, 2.3, 3.5 and 5.6 kHz, while the High Presence (high-mid) band can be set to 270, 390, 560, and 820 Hz, or 1.2, 1.8, 2.7, 3.9, 4.7, 6.2, and 7 kHz.

The EQ band gain knobs again turn 11-position switches, this time for five cut and five boost settings (the 11th position being unity), but the options change according to your choice of gain range, which presumably is why the numerical settings aren't specified on the panel. I think that's the right call; too much information would arguably be worse

»

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» than the clean turn-and-listen approach adopted here. To set the range, there's a three-position toggle switch to the left of the bypass toggle. In Original mode, you have $\pm 12\text{dB}$ of adjustment on the Presence bands and $\pm 15\text{dB}$ on the shelves, whereas in Mastering mode, the presence bands' control range is $\pm 4\text{dB}$ and that of the shelves $\pm 5\text{dB}$, so the switches give you far finer control over a more restricted range. A third choice, in the switch's middle position, is Hybrid mode, which combines the shelving options of Mastering with the greater mid-band ranges of Original. The output level isn't affected by the range control, and always offers you $\pm 3\text{dB}$ of gain.

In Use

The EQ169+ is incredibly quick and easy to dial in, thanks in no small part to the stepped controls. On the face of it, to inexperienced engineers who've grown up with software tools, stepped controls might appear to limit your options, but there's real merit in this approach. Just being able to reach out, tweak the controls and listen, in full confidence that you can return exactly and immediately to the previous setting, means you can easily compare and contrast settings. That's great for mastering and other work with complex material, but it's also a way of working that every student of audio should experience; a really effective way to train your ear

and develop your listening skills. In any case, there are plenty of frequency options to play with, and generous overlap between bands means that you can perform boosts and cuts close together, or dial in some pretty complex curves — between the HPF, the low shelf and the Presence bands, you have ample control over the punch and power region of a kick drum, for instance.

The stepped controls also made using two units on stereo material a breeze, but not just for mastering. For example, the review pair was a joy to use on a drum bus, and worked well on the stereo mix bus too.

Some EQs do offer you more than the $\pm 12\text{dB}$ of gain available in the mid bands, of course, but honestly, how often do you really need a broad 15dB boost/cut? While there's plenty of scope to shape any sound, I should note that the bandwidths are not adjustable, and I'd suggest that this EQ is much more about broad tonal shaping than the surgical removal of problems.

That you can restrict the gain range to make possible really focused adjustments is wonderful, and is one of the keys to this EQ being so supremely versatile — it's super easy to configure an appropriate gain range for whatever tracking, mixing or mastering application you have in mind — and this is something that's been managed without crowding the front panel, or cramming too many options into the range of any one control.

It's not long since I had the CS169 channel strip in for testing, and I opened up some projects I'd used that on for comparison. There are more EQ options on the EQ169+, of course, but the sonic character is just the same — just as impressive. What this translates to in subjective terms is an EQ that sounds clean and effortless, as you'd expect of a unit whose quoted frequency response ($1\text{k}\Omega$ load) is within $\pm 0.1\text{dB}$ from 5.4Hz to over 25kHz , a THD spec of 0.005% at 0dBu , and which has a signal-to-noise ratio in excess of 108dB . Still, as was the case when appraising the CS169, my ears detected an almost indefinable hint of silky smoothness: while this EQ sounds clean, it never struck me as sounding harsh or clinical.

The Bottom Line

This isn't the only seductive-sounding, high-quality analogue EQ today, and competition is fierce. But there are few models whose roots lie in the Studer consoles, and the EQ169+ has lots to commend it. If you want to hear it in action on a range of material, check out the examples on HLabs' own product page (<https://hlabs.audio/product/eq169-plus>). But the long and short of it is that this device is easy to use and it sounds superb; if I had to use no other EQ for tone-shaping ever again, I wouldn't complain. Given the sonic quality, the versatile features, and that the units are hand-built in Europe, the asking price represents decent value too. ■■■

summary

A lovely analogue EQ, whose design has roots in the circuitry of a classic Studer console, and which boasts not only a great sound but also some really useful practical features, such as the ability to adjust the gain range to suit different tasks.

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■ The Gain Range switch lets you toggle between settings that might be better suited for tracking and processing individual sources, or for mastering and bus-processing work.



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PAUL WHITE

With most of Electro-Harmonix's effects pedals, you use just a few intuitive controls to perform a specific task. But Oceans Abyss goes way beyond this remit. While it isn't exactly a budget unit, you get a lot of processing power for your money, and I'd say it finds itself in the rarefied company of pedals such as Strymon's Big Sky, Source Audio's Collider, Boss' RV-500 and IK's AmpliTube X-Space.

With dual stereo reverb engines and additional effects blocks, you could argue that the Oceans Abyss is in fact a dual-engine multi-effects pedal. Its main focus, though — at least where the 128 presets are concerned — is very much on reverbs, with the other effects generally playing a supporting role. It can create reverb effects that go way beyond the natural, yet it can also handle all the more conventional reverb scenarios. You can use any of the other effects without reverb if you want, though. For example, you could use one processing block to replicate a Small Stone phaser or Memory Man delay. Furthermore, as each engine has its own bypass switch, you could load different effects into each and access them separately during your performance. The audio quality has also not been neglected, and the pedal's converters run at 96kHz/24-bit.

In Control

Physically, the pedal is fairly wide (181 x 146 x 57mm WDH), and is housed in a robust folded-metal base that mates with an extrusion forming the front, top and rear of the case. The Oceans Abyss is controlled by a combination of knobs, sliders and three footswitches. The footswitches are, sensibly, located well away from the sliders, which have integral LED indicators in their caps that flash while they're being moved. The footswitches are used for bypassing the A and B effects engines, overall bypass and, by pressing both together, for preset up/down scrolling. The switches may also be used to enter tap-tempo data, with various customisation options offered in the Settings menu.

As with EHX's POG 3, a turn, rock and press NavCoder works in conjunction with a monochrome OLED display. The latter makes use of graphics where applicable to show the signal path, settings, patch saving, virtual fader positions and so on. Control settings are aided by those



Electro-Harmonix

Oceans Abyss

Digital Reverberation Pedal

If you're looking for more than reverb from your reverb pedal, you really should check out EHX's latest powerhouse...

illuminated slider caps (green on the left, blue on the right), again similar to those used for the POG 3, along with dedicated knobs for Pan, Pre-Delay and Time. The faders adjust Blend, Low EQ and High EQ, and there are buttons for Edit, Moment (which changes the related footswitch from latching to momentary action) and Tails. Reverb engines A and B have identical control sets, with master buttons for Preset, Home and Settings just below the display. The NavCoder, which is mounted centrally, is quite sensitive to movement, so you'll need just a little practice before you get used to it.

As for editing, the Home button shows the signal path, and this can be navigated using the NavCoder. The blocks, I/O

ports and links blink when selected. When a block is selected the NavCoder can be used to scroll through the list of available effects and when a connector is selected, rotating the NavCoder scrolls through the connection options. A Live mode allows for direct control using the panel faders, to make quick changes to the sound. And being able to see the various blocks and their connections on the display makes more detailed editing much easier — once you've learned the three-letter abbreviations used on screen.

On the rear panel are pairs of jacks for the inputs and outputs, which allow the pedal to be used in mono or stereo. There are also send and return jacks (TRS for mono or stereo connections) for inserting an external device into the signal path, a jack for connecting additional footswitches and a jack for an expression pedal. The footswitch input supports up to three additional footswitches. There's MIDI in and out on five-pin DIN connectors and a USB-C port for connection to a computer for running EHXport software, similar to that already available for the POG 3. This was not available at the time of writing, but should streamline preset editing and handling. The comprehensive MIDI support includes Program Change, Control Change and MIDI Clock sync. Power comes in via a standard barrel

connector, with the pedal requiring 9.6V DC/500mA. A suitable PSU is included in the box.

Reverb Algorithms

There's a choice of 10 reverb types, and each can be panned independently — great for dramatic stereo effects when recording or playing live through a stereo system. While only one reverb can be run in each of the A and B sections, the remaining effects blocks can be distributed as you see fit, and can be used in multiples if needed (there's one preset that uses eight analogue delay blocks to simulate a BBD reverb). Also, note that each effect block is stereo, so that opens up a lot of routing possibilities.

The reverb algorithms fall into the categories of Room, Hall, Spring, Plate, Reverse, Dynamic, Auto-Infinite, Shimmer, Polyphonic, and Resonant. Routing allows for numerous series or parallel configurations including running one engine's output into the other to cascade all the blocks.

Dynamic mode allows for ducking, swelling or gated reverb, while Auto-Infinite mode will appeal to those wanting their sound to build into an infinite drone; in effect, this mode listens to your playing and crossfades to a new reverb wash when new notes are detected. Two modes determine how the crossfade operates. In X-Fade mode it crossfades smoothly from the previous reverb wash to a new one, while Swell mode mutes the previous wash, fading in the new one. Polyphonic mode includes up to two pitch-shifters running into a Hall or Plate reverb, while Resonant mode also offers two sub-options: Chimes mode is a reverb coloured by tunable resonators, while Filtered uses a configurable, self-oscillating filter to treat the reverb tail. Those working on chill-out, cinematic or ambient music will find a lot to explore within these less-conventional reverb algorithms.

Starting out with what I'd call the bread-and-butter Plate, Room and Hall reverbs, these come across as serious, studio-quality effects, and the same is true of the excellent Spring and Plate emulations. The more adventurous may want to explore the Shimmer and

Resonant reverb modes before jumping into the effects section. Shimmer is very smooth and creates perfectly that impression of a hidden string machine mirroring your playing. Even more so if you add chorus.

If the range of reverb treatments is impressive, the pool of 12 effects modules can transform the sound. These include some excellent delays and modulation effects. As explained, multiples of any block type other than reverbs can be combined. So you could, for example, combine several delays to create your own multi-head tape delay emulation, or cascade overdrive modules to build a fuzzbox. The delays offer digital, analogue and tape variants. The remaining effect categories are Modulation (chorus, flanger, phaser or tremolo), Graphic EQ, Saturation/Overdrive, Bit Crushing and Volume control. And if that still isn't enough, those stereo effects-loop jacks let you connect external pedals or processors. Bypass can be set to buffered 'with tails' for the reverbs/delays to continue to die away naturally once the pedal is bypassed.

The factory presets provide some insight into what this pedal is capable of, but there are so many possibilities that you'll probably want to fine-tune and re-save the existing presets or build your own from scratch. Effects can range from dreamy and ethereal to glitchy and dramatic, and swells, shimmers and even backing drone pads are within easy reach.

Impressions

I have to say that the Oceans Abyss is beyond impressive. The conversion sounds clean and the conventional reverbs are top-tier, including the very convincing Spring, complete with drippy

twang. (You can even set an external switch to replicate the sound of a spring reverb being kicked, should that appeal!) But if that's all you need, this pedal is most definitely overkill — check out EHX's popular Holy Grail or Oceans 11 if that's what you want.

Rather, it's a reverb workstation that will hold particular appeal to ambient musicians and exponents of shoegaze, or anything for which evolving soundscapes are central. Doubling up on reverbs offers some interesting options, such as running a Reverse reverb into a Hall or Plate, or panning a Plate reverb to one side and a Spring reverb to the other. But once you start exploring the crazier reverb types and putting modulation before, after or between them, life gets much more interesting! Feeding a distorted signal into reverb can also be effective, while for a shoegaze wall of noise, try feeding a long reverb through distortion — and then perhaps flange the end result!

Of course, you must master the editing menus if you're to get the most out of it, and the sheer flexibility of this pedal inevitably means that there's some depth to the menus. Thankfully, EHX have made navigation and editing intuitive, and the individual effects modules have sensibly concise control sets. The promised free EHXport software should make patch editing even easier still. Heartily recommended. **■■■**

summary

The Oceans Abyss is an extremely sophisticated pedal that goes way beyond reverb, and is capable of some truly beautiful sounds. It might better be described as a reverb workstation with integral multi-effects and high-quality, dual stereo reverbs... though its delay and modulation effects are excellent too!

£ £439 including VAT.
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Lauten Audio Kick Mic



Lauten's latest makes crafting the perfect bass drum sound a breeze.

PAUL WHITE

Lauten Audio's Kick Mic is an end-fire capacitor microphone, designed specifically for use with bass drums. In contrast to some conventional kick-miking practices, this mic is intended always to be placed inside the drum, via a hole cut in the resonant head. According to Lauten Audio, this approach addresses common issues such as low-frequency build-up, phase problems from reflections within the drum shell, and unwanted bleed from other kit components.

Physically, the Kick Mic is robust and compact, suitable for both studio and live use. The mic body measures

125 × 50mm and it weighs 290g. An included swivel-mount collar with a thread adaptor (for compatibility with both EU and USA mic-stand threads) allows precise positioning inside a kick-drum port. Unlike some swivel mounts, this one locks in place very securely. The only other accessory supplied is a soft storage pouch.

Requiring 48V phantom power and equipped with a FET preamp stage, the Kick Mic has a tight cardioid polar pattern designed to remain directional down to very low frequencies — around 50Hz. (In contrast, most directional mics tend to become more omnidirectional at low frequencies.) The Kick Mic's KC608 capsule, which is specified to operate over the full 20Hz-20kHz frequency

range, features a 17mm diaphragm mounted within a 40mm element in a rear-ventilated chamber — a topology intended to deliver a fast transient response and a balanced blend of beater attack and drum-body resonance. As with most kick mics, the frequency response is intentionally not flat, but at same time it's not so dramatically shaped as some other kick mics like the AKG D112.

Because of its intended position inside the drum, the capsule is designed to handle very high SPLs of up to 146dB. Self-noise is low at less than 15dBA, which is negligible in the presence of a loud source such as a kick drum. Figures for sensitivity are almost irrelevant for a mic working in such high-SPL environments, and I needed to set a very low mic gain

on my interface to achieve a sensible level without the risk of clipping.

Another key aspect of the design relates to what Lauten call Air Coupling Modes. These are essentially vents in the rear of the capsule, intended to mitigate the effects of turbulence created by the rapid movement of air inside a kick drum as the beater strikes. The aim is to prevent the mic from being overloaded by a build-up of headroom-eating sub-bass energy, which can lead to a muddy or distorted sound. These vents also help the mic to maintain its directionality down to very low frequencies.

“For any given session the ideal mic placement spot can be found very quickly.”

Onboard tonal shaping is provided by switchable 140/80 Hz high-pass filter and 5/12 kHz low-pass filter, offering several tailored response options to suit the drum and musical style. These filters allow the user to tame excessive low end, emphasise the midrange, or soften an overly aggressive beater attack.

Pumped Up Kicks

The Kick Mic's recommended placement inside the drum significantly reduces spill from nearby drums and instruments, while the ability to balance the impact and resonance of the drum by adjusting the position of the mic makes it relatively

easy to achieve a strong kick-drum sound. Gone are the days when engineers had to spend hours chasing a usable miked kick sound! The tight polar pattern also minimises the effect of internal reflections, and with careful positioning and EQ switching it's possible to coax a wide range of bass-drum characters from a well-tuned drum.

For live work, the benefits of quick setup and reduced bleed are substantial, and the mic's high SPL handling capability means even the loudest drummers won't faze it. However, the Kick Mic also performs well with smaller drums — for

example, tested with a 16-inch Tama kick, it still served up a surprisingly weighty recorded tone. As expected, altering the mic's position changes the character in very perceptible ways: the furthest-forward recommended setting places the mic's EQ switches just inside the drum, and yields a balanced, 'general purpose' kick sound. For the small bass drum I found this position to be the one that delivered the most depth and weight. Fine-tuning the placement may be necessary depending on the drum's construction and tuning, so for a large kick drum that tends to be heavy on bass and short on beater click, moving

the mic a little closer to the beater helps achieve a better balance. Further tonal variation can be heard when moving the mic from a centre position towards the edge of the shell. Because the Kick Mic is an end-fire design, positioning is relatively straightforward even when the resonant-head hole is small.

While bass drums may seem like known physical quantities, when you get them into the studio every one seems to behave differently, and that's where the Kick Mic comes into its own — its predictable behaviour means you soon get a feel for what tonal changes to expect when adjusting its position. For any given session the ideal mic placement spot can be found very quickly. The reduction in spill from other sources is also a valuable attribute.

In summary, the Lauten Audio Kick Mic represents a genuine solution to a number of routine problems when faced with miking kick drums. It may not be the only mic able to produce a good kick sound, but its 'better mousetrap' benefits are definitely worthwhile. **///**

summary

The Kick Mic has a number of well-thought-out features that make it a real problem-solver when recording bass drums.

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Universal Audio

Volt 876 USB Audio Interface

UA's Volt range reaches its full potential with a rackmounting, multi-channel USB interface.

SAM INGLIS

Universal Audio have built up an incredibly strong brand by marrying vintage and modern sensibilities. In part by cultivating relationships with other big names from the analogue world, they've generated a strong association with the golden age of tape-based recording — yet, since the late '90s, the products on which they've built this reputation have been digital. The UAD-2 Powered Plug-in platform, the Apollo range of Thunderbolt audio interfaces and the Sphere modelling mics all major on traditional values, but they realise those values through DSP. An all-UA recording system can thus mimic the sonic qualities of tracking to tape, on vintage mics, through a classic console and outboard, but it does so within a modern computer-based environment.

The cost of such a system is a tiny fraction of the gear it emulates, but even so, UA have historically positioned their plug-ins and Apollo interfaces towards the premium end of the market. To cater for those whose budget won't stretch that far, they more recently introduced the Standard Series modelling microphones and the Volt line of audio interfaces. The Standard mics offer basic modelling options through the Hemisphere plug-in, while the Volt range dispense with the Apollos' built-in DSP, and connect via USB rather than Thunderbolt.

Until now, the Volt range has been comprised of small, bus-powered, desktop models with small I/O counts, no digital I/O, and basic hardware input monitoring. If you wanted ADAT expansion, or low-latency input mixing in UA's Console software, you'd have to step up to the Apollo range. That's all changed with the introduction of the Volt 876.

Volt Face

The new model boasts numerous features that, within UA's product range, were previously found only in Apollos.

It's a mains-powered, full-width 1U device that comes with removable rubber feet and rack ears, and its rugged metal case is surprisingly deep and heavy. This is partly because it features a built-in mains power supply, something that will have Apollo owners casting envious glances. On the analogue side, it sports eight mic/line inputs on combi XLR/jack sockets, with the first two located on the front panel. The front-panel inputs offer a high-impedance mode for tracking electric guitars. At the other end of the front panel you'll find two independent headphone outputs, augmenting the eight line outputs on the rear panel.

The 876 is the first Volt to feature digital I/O, and it's pretty comprehensive. You get two sets of optical Toslink in and out sockets, for either stereo S/PDIF or up to 16 channels of ADAT Lightpipe I/O, plus word clock in and out. There's also MIDI in and out on full-sized DIN sockets, while connection to the host computer is made using a single Type-C USB port.

The Volts cannot host UAD-2 plug-ins, and consequently do not implement Apollo features such as Unison modelling



Universal Audio Volt 876 £860

PROS

- Full digital control enables Auto-Gain and seamless integration with UA's Console and LUNA software.
- Preset FET compressors on all eight analogue inputs.
- Can be controlled from a computer even when configured as an ADAT expander.
- Decent specifications and nice loud headphone amps.
- Built-in talkback and speaker switching.

CONS

- Routing within the internal mixer and UAD Console is less intuitive and less flexible than in some interfaces.
- Fixed threshold relative to 0dBFS means there's a limit to how hard you can drive the compressors.

SUMMARY

UA's Volt range just got a whole lot more serious, with an expandable USB interface that can tackle challenging multitrack recording tasks.

mic preamps, or low-latency monitoring through plug-ins in UAD Console. Instead, Volt preamps have a switchable Vintage mode, and on some models, a preset FET compressor based on UA's classic 1176 design. Both features operate in the analogue domain, and both are available on all eight of the Volt 876's analogue inputs.

However, with so many different settings available for each 876 channel, UA have moved away from the analogue, one-button-per-function control paradigm of previous Volts. Instead, a set of eight illuminated buttons is used to select one or more inputs, whereupon a rotary controller and a further set of buttons are used to adjust preamp parameters for the selected inputs. In other words, although the preamps on the Volt 876 don't support Unison, they are nevertheless fully recallable and under digital control. A 60dB gain range is available, calibrated in 1dB steps, and adjacent odd/even input pairs can be linked to maintain gain matching for stereo recording.

To the right of the input controls is a bank of eight five-segment LED meters, and to the right of this, you'll find the Volt 876's 'master section' controls. A main encoder with LED halo governs the output level at the first pair of outputs, which are designated monitor outs. There's also a separate mute button, but it's the other three controls that help to differentiate the Volt 876 from comparable interfaces. The Dim/Talk button simultaneously dims the monitor outputs and engages the 876's built-in talkback. By default, this uses a small mic embedded in the front panel, which is also available as a recording source to your DAW. The 876 also implements a slightly more basic version of the Apollos' speaker switching, whereby it's possible to designate outputs 3+4 as

being an alternate monitor output pair, and use the Alt button to toggle between these and the main monitor outs. And lastly, there's a button labelled Direct, which lights up green when engaged.

Going Direct

The desktop Volts offer a very simple hardware-based implementation of direct monitoring. That approach wouldn't really cut the mustard on an interface that can potentially be handling more than 20 inputs, so UA have brought this into the digital world too, making the 876 the first Volt to be controllable from their familiar UAD Console app.

Within UAD Console, you'll see a new mixer pane that presents all of the 876's digitally controllable features. This means it's possible to control all the abovementioned preamp parameters, and to set up monitor balances that draw on all the analogue and digital inputs, plus a stereo loopback channel. As well as having a fader, mute, solo and pan controls, each input channel also has sends to two dedicated Cue busses, meaning that in effect, you can create three different low-latency cue mixes. These can be distributed between the 876's main monitor outs and its two headphone outputs. However, DAW return channels don't appear in the Console mixer at all, so if you want your previously recorded signals to appear both in the main mix and one or both of its Cue busses, you'll need to arrange this within your DAW, or output to the loopback channel instead.

The Volt 876's Direct button, in effect, is a remote control for the Console channel mute on whichever input(s) are selected using the front-panel buttons. So if, for example, you have input 3 selected on the 876 and toggle the Direct button, you'll see channel 3 in Console

»



Most of the Volt 876's I/O is on the rear panel, with only the first two inputs and the headphone outputs at the front.

» unmute itself, and you'll hear input 3 at the monitor outs and in the Cue busses. If it's routed to an unmuted, armed track in your DAW, you'll hear it 'indirectly' too.

The idea of having dedicated Cue busses is inherited from the Apollo, where it's designed to support more complex setups than can be created with the 876. With this interface it enables integration with LUNA (see box), but in other respects, a simpler architecture might be more intuitive — personally, I'd rather just have DAW returns appear in the Console mixer, and sends routed directly to outputs rather than busses. As it is, the limitations on routing seem a bit arbitrary. For example, each of the Cue busses can be mirrored to exactly one pair of physical outputs in addition to one or both of the headphone outs. Unsurprisingly, the Volt 876 doesn't support surround monitor control, and has no built-in DSP that could be used to host speaker correction profiles, as the Apollos can.

Making Gains

Having made the 876's preamps digitally controllable, UA have taken the logical next step and implemented the Assistive Auto-Gain algorithm that was introduced with the Apollo X Gen 2. It

works in exactly the same way here, giving you choice over the duration of the auditioning period, the target peak signal level, the threshold level at which signal is held to be detected, and so on. As on the Apollos, Auto-Gain can be applied to multiple inputs at once, and will maintain consistent gain levels for linked inputs. You may find you need to reduce the default listening threshold level for quiet signals such as speech, but in general it works well, and is every bit as effective with the Volt 876 as it is with the Apollo.

UA recommend running Auto-Gain with the preset compressors switched off, then activating them afterwards if required. If you ignore their advice, Auto-Gain in effect 'hears' the compressed signal when making its decisions, so the target level you select will influence the amount of compression that is applied. There are three compressor modes, named Fast, Guitar and Vocal. The main variable that changes when you switch between them is the release time; all have relatively fast (sub-2ms) attack times, a fixed 6:1 ratio and 6dB make-up gain, and a soft-knee threshold configured such that the compressor starts to apply gain reduction when the input signal gets within 12dB of the clipping point of the

A-D converter, and doesn't reach the full 6:1 until the threshold is exceeded by several dB. So if you're like me and tend to leave a lot of headroom at recording, for example by setting the Auto-Gain target to -12dBFS, the compressor won't do much at all. To achieve noticeable amounts of gain reduction, you'll need to set a higher Auto-Gain target, or adjust the gain manually. As on the other Volts, there's no visual indication of whether gain reduction is taking place, or how much, so you'll need to be guided by ear. Typically, the compression is pretty transparent in Guitar and Vocal modes; Fast mode is more assertive, though still smooth except on sources with a lot of low-frequency content.

More Volts

One of the features that differentiates the Apollo interfaces from the Volts — and indeed almost all USB interfaces — is scalability. If you want to increase the I/O count of an Apollo setup, you can simply daisy-chain an additional unit over Thunderbolt. This generally is not possible with USB devices, and the Volt 876 is no exception. However, the Volt 876 can act as an ADAT expander as well as an interface, and UA have

»



When the 876 is used as an ADAT expander, Host mode allows it to be controlled from within UA's Console, as shown in this screen capture — note that the same preamp controls appear at the top of the ADAT channels as on the first eight channels.

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» implemented some clever control features in this scenario. The inclusion of word clock in and out sockets is also a positive, especially in cases where you might want to connect multiple expanders to a single host.

For expansion purposes, the 876 needs to be switched into standalone ADAT mode, whereupon it pipes up to eight channels of audio each way over Lightpipe. However, as long as you have a spare port, you can still hook up a USB cable from the 876 to your computer in addition to the optical cables that are tying it into your primary interface. The Host LED will then light on the expander, and you'll be able to control its preamps from Console in exactly the same way as those of the main Volt. With a generic ADAT expander, the Console mixer just shows a fader and cue sends for each ADAT input channel. With an 876 in standalone ADAT + Host mode, it shows all the same controls as you'll see for the

that the true figure was lower, at just under 6ms. A 16-sample buffer size is also available, so you should have no problem achieving decent real-time monitoring. UA also say that low-latency performance is better under the new macOS 26.

The term "32-bit" is rapidly becoming familiar in audio interface marketing materials, and has the potential to cause serious confusion. In a few products, mostly targeted at location recording, it refers to a distinctive architecture where parallel A-D converters are used to generate a 32-bit floating-point audio signal, rather than the more conventional 24-bit fixed-point signal. This has the fairly major advantage that the converters and the audio signal can handle the entire range of possible input signals regardless of gain setting, making the device effectively impossible to clip. By contrast, the Volt 876 is one of a growing number of modern audio interfaces that are touted as "32-bit" simply because

control voltages for modular synths. In terms of the competition, there's pretty much nothing in it between the Volt 876 and Focusrite's Scarlett 18i20 on the specs front; neither can quite match PreSonus's Quantum HD8, SSL's 18 or Audient's iD48, but both are more than good enough for any conceivable real-world application.

High Voltage

So which interface should you buy? Thankfully, that question is getting easier to answer, because manufacturers are realising that their products need to have a unique selling proposition on top of the now-standard eight mic/line inputs and ADAT expansion. The SSL 18, for example, features the ability to operate at 'professional' +24dBu line-up levels, a pair of insert points, and SSL's famed Listen Mic Compressor. The Quantum HD8 has dedicated re-amp outputs, a colossal preamp gain range, and outstanding low-latency performance. The iD48 provides switchable insert points on all eight analogue inputs, which can be used at mixdown and recording without re-patching. What does the 876 offer to differentiate itself?

The Volt 876's ability to be remotely controlled in an ADAT expander configuration is unusual, but not entirely unique. A similar feature is available with the Quantum HD8 and with Audient's EVO range, though its implementation here is particularly slick and thorough. The 876's Vintage preamp setting is likewise comparable to SSL's 4K mode and Focusrite's Air option, and features such as the built-in talkback mic, alternate speaker switching and DC-coupled outputs are also available elsewhere. However, I can't think of another interface that combines *all* of these features in the way that the 876 does. And I'm pretty sure that there isn't another audio interface on the planet that offers built-in analogue compressors on all eight of its mic/line inputs. Admittedly these are preset-only, offering limited flexibility and no metering, but it's very hard to make them sound bad, other than by amplifying any noise on quiet signals.

Finally, another factor that might influence buying decisions is the supplied software bundle. Most interfaces now come with a decent selection of plug-ins, but the 876's UAD Producer Suite hits hard, with goodies like their LA-2A, 1176 and Pultec Collection included.

"I'm pretty sure that there isn't another audio interface on the planet that offers built-in analogue compressors on all eight of its mic/line inputs."

first unit, and supports Auto-Gain (the Direct button and additional unit's monitor section, however, are inoperational). Better still, you can even use an 876 configured thus as an expander for an ADAT-equipped Apollo. In this scenario, the 876's input controls show up in a separate window, but this can be minimised and floated over the Apollo mixer window.

Expanding an 876 or Apollo system this way is not as powerful as cascading multiple Apollos, though, partly because there is no way to access the headphone sockets on the expander units. This seems a shame. On the face of it, I can't see why an 876 used as an expander couldn't at least mirror the first pair of return signals to the headphone outputs.

Bits & Pieces

Another advantage that Thunderbolt interfaces typically offer over USB models is better low-latency performance. With the 876 on my Mac, the 32-sample buffer size delivered a reported latency of just under 8ms at 44.1kHz; oddly, though, Oblique Audio's RTL Utility confirmed

they use the latest generation of converter chips. In this context, the term refers only to the internal operation of the converters, so as far as the user is concerned, the Volt 876 can only ever handle 24- or 16-bit audio, and is very much not immune to clipping.

However, these converters do mean that the 876 has significantly better audio specifications than the desktop Volts. Dynamic range is 116dB on all analogue inputs and 119dB on the outputs, noise and distortion are impressively low, and unlike some affordable USB interfaces, the Volt 876 can both accept and put out a reasonably hot line level. Its outputs are aligned such that 0dBFS produces +20.2dBu, while the line inputs can handle up to +22dBu. Sensibly, although the line and mic inputs both pass through the preamp circuit, the sensitivity of the XLR inputs is greater, and at the lowest gain setting, the mic input can accept a maximum +12.5dBu. The headphone outputs, too, are noticeably more powerful than those on many interfaces, and all the analogue outputs are DC-coupled, so can be used to generate

Volt 876 & LUNA

If you're an audio interface manufacturer, the advantages of offering your own DAW software are fairly obvious. Not only can you bundle this software to drive hardware sales, but you have complete control over integration between the two. Thus, although most DAW software will work well with third-party hardware, it often steps up a level when used with an interface from the same manufacturer.

Universal Audio launched their ambitious LUNA recording software back in 2020, and initially it was closely tied to their Apollo hardware. Since then, UA have added support for generic Core Audio and ASIO interfaces, and have continuously expanded the feature set to the point where it's now an impressively complete package. Best of all, it remains free to download and use, although there is now a Pro paid-for tier that incorporates various LUNA Extensions such as the API Vision Console Channel Strip and Studer A800 Tape Recorder.

To coincide with the introduction of the Volt 876, UA have more or less simultaneously launched version 2.0 of LUNA. This brings some banner features that undoubtedly justify the integer upgrade, most notably the addition of ARA (Audio Random Access) support, allowing products such as Celemony's Melodyne and NoiseWorks' DynAssist to be used directly within the LUNA editing window. The other big step forward is the introduction, in the Pro version, of a comprehensive Hardware Insert system, which handles delay compensation and routing automatically, and allows hardware chains

to be saved as presets. Elsewhere, LUNA's automated instrument detection is much improved in version 2.0, and UA have also built in the means to control the Volt 876 from the LUNA channel strip or mixer.

In essence, this populates the Preamps section of the channel strip with all the same controls you see in the Console mixer. You can adjust gain, change the compressor mode, toggle phantom power and polarity, select instrument mode for inputs 1/2, and so on. It's also possible to run Auto-Gain from LUNA, although this option only appears when a track is record-armed or when input monitoring is active. Channels in the LUNA mixer are automatically populated with sends to Console's two Cue busses, and you can have a further two Cue sends that go direct to 876 outputs.

What you don't get with the Volt 876, though, is the Accelerated Realtime Monitor feature that is available to Apollo users. This, in effect, integrates the Console mixer into LUNA, allowing you to set up low-latency cue mixes without tabbing back and forth between the two applications. I'm not sure if it would have been technically feasible to implement this with the Volt, but its absence does mean that you'll either have to keep your buffer size low while recording, or keep Console open in the background. In the latter scenario, it would be nice to have the option for the 876's Direct button to toggle track mutes in both applications at once, so that a single press would mute the Console channel and unmute the LUNA channel, or *vice versa* — although the switching logic involved might be too complex.



The Volt 876 is smoothly integrated into LUNA. Here, I've launched Auto-Gain on one of the inputs from a second 876 in ADAT + Host mode.

As I mentioned at the outset, Universal Audio's Apollo and UAD product lines are built on the idea of emulating classic analogue equipment in the digital domain. So it's possibly a little ironic that if you want 'real' analogue compression in your audio interface, you can't get this in an Apollo, but you can in the more affordable Volts. And whereas previous Volt models

haven't encroached on the Apollo line at all, the Volt 876 neatly bridges the gap between the two ranges. The Apollos still offer enough, both functionally and in terms of audio specifications, that you probably wouldn't buy a Volt 876 as your primary interface in place of an Apollo x8p. But if you're an existing Apollo owner looking to expand your rig, the 876 is

a pretty tempting option, thanks to its close integration with UAD Console and LUNA.

And if you're starting from scratch, the Volt 876 is a classy, feature-rich design that will do everything you need of a project-studio audio interface and more. **///**

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40 IDEAS THAT CHANGED MUSIC

We continue our countdown of the most significant technological breakthroughs in recording history.

THE SOS TEAM

Sound On Sound celebrates its 40th anniversary this year. To mark the occasion, we've surveyed the history of recorded music and put together our own Top 40, aiming to identify the most significant developments in music tech since music tech became a thing. In the second part, we count down from 30 to 21...

30: Pitch-shifting

We take it for granted now that if you record something a little sharp or flat, you can knock it into shape with a readily available pitch-shifting plug-in. But no such luxury was around in 1955, when Musique Concrète pioneer Hugh Le Caine released 'Dripsody', a composition based around a recording of water dripping into a metal basket, where the tape was manipulated by looping it, reversing it, delaying it and, most notably, by adjusting the playback speed. Halving the speed would lower the pitch by an octave, and doubling it would play back an octave up, but Le Caine took things much further: he built a small keyboard that allowed him to play melodies — a remarkable achievement that now seems astonishingly prescient! Naturally, changing the speed of tape changes both the recording's pitch and duration together, unlike the 'pitch-shifting' we use in our DAWs today. It would be almost 20 years before a dedicated digital pitch-shifter came out, but it was Le Caine who walked so that the Eventide H910 Harmonizer could run.

29: Digital delay

Delay effects in music, such as those pioneered by guitarist Les Paul,

were initially tape-recorder-based. The mid-'70s saw the advent of affordable, analogue, solid-state delay units with short delay times and limited audio bandwidth from Electro-Harmonix, MXR and others, but as digital memory chips became both more capable and affordable, mainly thanks to the growth of the computer industry, the first digital delay units were launched by Eventide, Lexicon and MXR. Even with, initially, limited delay times and low bit depth (often just 8- or 12-bit), they were still more usable than anything that had gone before, as they allowed modulation, looping and even rudimentary sampling. Once 24-bit conversion became available, a digitally delayed signal could be indistinguishable from the source, even at long delay settings. Now, many digital delay units, particularly in the guitar market, offer emulations of some of the lo-fi 'defects' of earlier echo technologies, often felt to be more characterful and musical!

28: Distortion as an effect

Featuring Ike Turner but credited to Jackie Brenston & his Delta Cats, the 1951 hit 'Rocket 88' has been called the first rock & roll record. That may or may not be true, but it's certainly remarkable for the sound of Willie Kizart's electric



The Sennheiser Mikroport was the world's first belt-worn portable radio mic system.

guitar. Legend has it that the loudspeaker in his amplifier was damaged on the way to the session; producer Sam Phillips decided he liked the sound and featured it prominently in the recording. This accidental discovery kickstarted a process that would see the guitar take over as the dominant instrument in pop and rock music, and it wasn't long before devices such as fuzzboxes were introduced to create similar effects in a more deliberate and controllable fashion. And it didn't stop there. Everything from drums to didgeridoos has been distorted to creative effect, and it's hard to imagine what rock or metal music might sound like without distortion.

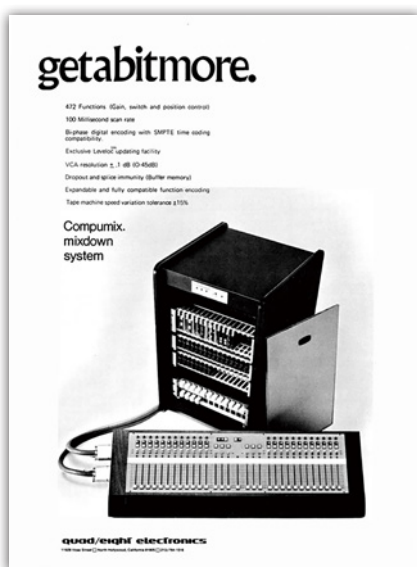
27: Wireless microphones

Cables can be enough of a nuisance in the studio, but in live performance and on television or movie sets, they really

hamper freedom of expression. Several companies explored the idea of using a radio-frequency link rather than a length of copper to connect microphone and preamp in the 1950s. Shure's Vagabond system, introduced in 1953, featured an omnidirectional mic attached to a substantial stand which contained the transmitter electronics. Then, in 1957, Sennheiser introduced the Mikroport, the first 'personal' wireless system with a belt-pack transmitter. Its valve-based electronics could run for a whole 11 minutes on one set of batteries!

26: Audio over IP

SOS readers with long memories may recall Yamaha's mLAN system, introduced around the turn of the century. At the time it was notable for being the first digital audio protocol to use the FireWire standard, but it was also one of the first attempts to create a networked system where multiple devices could communicate. Numerous manufacturers signed up to use the mLAN protocol, but relatively few products emerged. The appeal of networked audio remained, however, and once developers such as Audinate had solved the problems of using standard Ethernet infrastructure for audio traffic, audio over IP quickly made inroads in professional environments such as broadcast and live sound. The ability to distribute audio signals across a network rather than merely in a fixed chain has proved little short of revolutionary.



■ Quad-Eight's Compumix was one of the earliest commercial mix automation systems.

25: FM synthesis

Modulation of one signal by another is a basic principle of radio, and by the late '60s, analogue synth designers had discovered the creative possibilities of allowing one wave to modulate either the amplitude or the frequency of another. Under some circumstances, the latter technique could produce musical sounds with harmonically related partials, but analogue oscillators were too unstable to do so repeatably and reliably across a wide range. However, Stanford University's John Chowning developed a system whereby multiple digital 'operators' — essentially, simple oscillators outputting sine waves — could be combined to generate complex timbres. Having tried and failed to interest US synth companies in his work, he signed a deal with Yamaha. The first commercially available FM synth was 1980's GS-1, but it was the DX-7, released in 1983, that would be FM's breakthrough. Not only did the DX-7 sound utterly unlike the analogue synths that had gone before; with its 16-note polyphony and velocity-sensitive keyboard, it was a completely different playing experience.

24: Mix automation

As productions became increasingly complex in the early '70s, the challenges of mixing grew exponentially. Multiple pairs of hands were needed to make all the necessary fader moves. A mix could not be recalled or repeated, and it was often necessary to piece together a master take from the best bits of several mixes. The advantages of automating the process were obvious, and by the turn of the next decade, Solid State Logic would be firmly established as market leaders. However, many other mixer manufacturers tried their hand at devising a computerised mix automation system, with varying degrees of success. Among the first to gain a foothold in the real world was the Quad-Eight Compumix, as used on Neil Young's 1973 live album *Time Fades Away*. Young himself hated the album so much he refused to issue it on CD until 2017.

23: Portable synthesizers

In its early days, electronic music was very much a studio-based affair. Whether they worked with an assemblage of different devices, as in the BBC's Radiophonic Workshop, or with a modular synth like the early Moog designs, logistical issues prevented synthesists from becoming just another member of the band. All that changed with the invention of compact, complete synths like the Moog Sonic Six, EMS VCS3 and, most influentially of all, the Minimoog. Suddenly, keyboard players could tour with an instrument that was small enough to be flightcased, and simple enough to be reprogrammed between songs. Is it a coincidence that progressive rock became popular at exactly the same time?

22: Direct injection

The idea of directly recording the output of electrical instruments, rather than miking an amp, seems so obvious in retrospect that no-one seems to have thought the DI box an invention worth documenting. Most major studios had custom-built DI systems by the mid '60s, and direct injection often became the default approach for bass guitar in particular, notably at Motown in Detroit. However, pioneering British producer Joe Meek was certainly recording bass direct as early as the late '50s, first at Lansdowne Studios and later at his own studio on London's Holloway Road.

21: The pan pot

Early stereo mixes of pop and rock tracks often sound extreme to modern ears, because the individual sources are all either central or hard-panned. There's still something to be said for this as an approach to mixing, but in the '60s, it was not done deliberately. The designers of most mixing consoles simply hadn't anticipated that anyone would want to artificially synthesize a stereo picture from mono sources at mixdown. As four-track gave way to eight- and then 16-track recording, the scope for miking individual instruments separately grew exponentially — and the ability to place these sources at intermediate positions within the stereo field, and even have them appear to move, was vital. ■■■



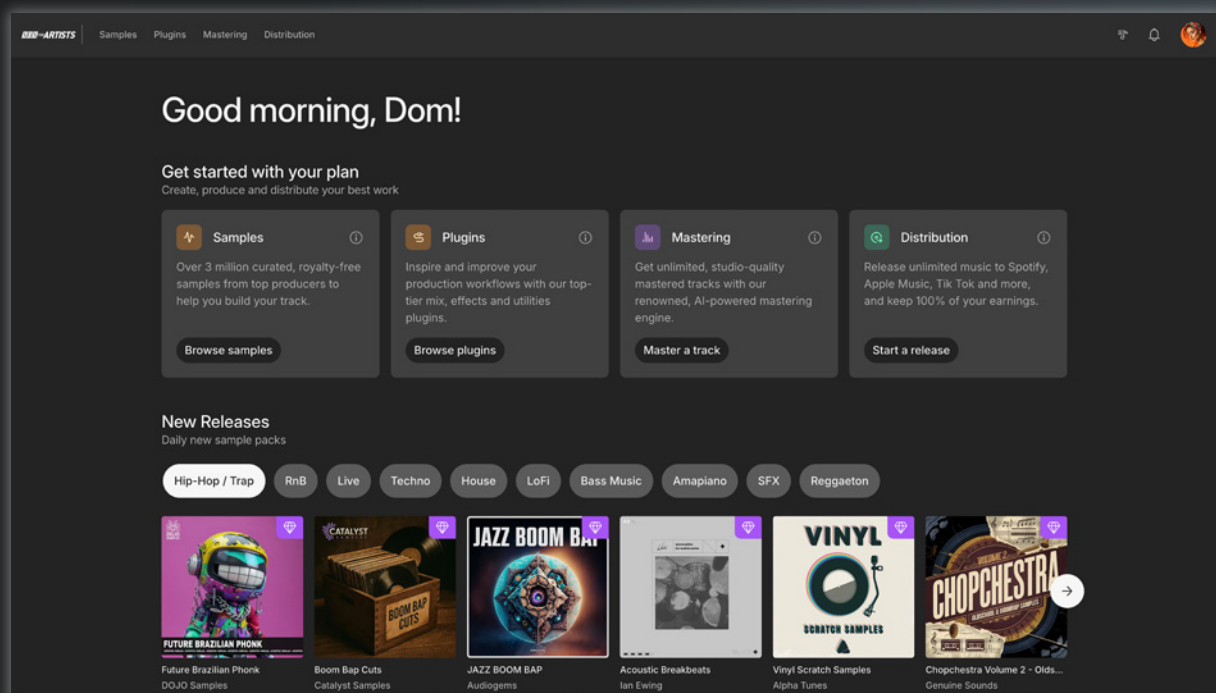
■ Modern manufacturers are still making DI boxes inspired by the famed Motown units.

40
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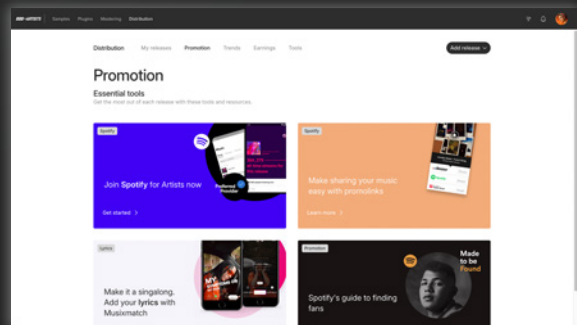
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IK Multimedia

ARC On-Ear

Headphone Calibration System

Headphone correction gets physical with IK's neat desktop DAC.

SAM INGLIS

The benefits of using EQ to flatten the response of a monitoring system are now widely accepted, and competition is hotting up. In 2024, IK Multimedia introduced ARC Studio, a hardware device that can load room correction profiles created in their ARC software. In 2025, Audient countered with the ORIA Mini, the first dedicated, standalone solution for hosting profiles created in Sonarworks' widely used SoundID.

ARC and SoundID will both improve the in-room response of a speaker system. SoundID is also designed to correct the frequency response of headphones, but the ORIA Mini can't host these headphone curves. There are a handful of audio interfaces that can do so, but there's never been a dedicated hardware counterpart

to the ARC Studio or ORIA Mini targeted at headphone listening. If you sensed that you were just about to read the words "until now", you'd be spot on, because that's exactly what IK Multimedia's ARC On-Ear is.

Into The ARC

ARC On-Ear ships in a cute and compact cubic cardboard container, and comes with a handy semi-rigid carry case. The form factor of the unit itself recalls that of the old Focusrite VRM Box from many years ago, being a square-ish unit about three inches across designed to sit on a desktop. It's surprisingly heavy for its size, partly because its shell is made of metal and partly because it has a built-in lithium-ion battery. Audio I/O comprises a full-sized headphone jack on the front, plus a mini-jack labelled Analog In on the rear, where it's joined by an on/off switch and a USB Type-C socket. The top panel features three buttons with logos that illuminate when active, and a large volume control. This is an old-school potentiometer rather than an encoder, and the one on the review ARC On-Ear felt a bit wobbly.

The package includes a short USB cable and an equally short balanced audio cable with mini-jacks at either end, one of which has a full-sized jack adaptor screwed

IK Multimedia

ARC On-Ear

€300

PROS

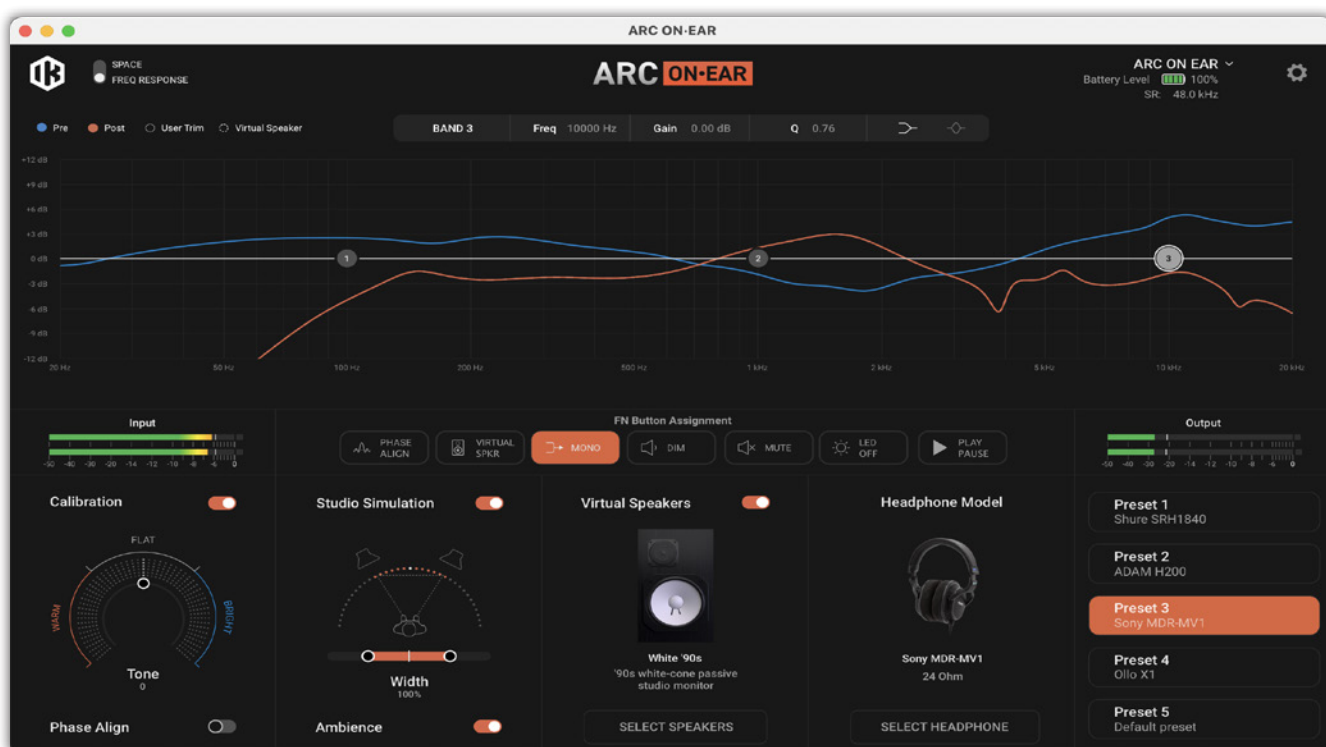
- Headphone calibration and virtualisation in a portable, bus-powered hardware device.
- Capable of standalone operation, and has an internal battery.
- Impressively vast range of headphones supported at launch.
- Editing software is easy to use and well designed.

CONS

- No S/PDIF input, which would make it easier to use alongside another audio interface.
- There is currently no software or plug-in vehicle for IK's headphone correction profiles.

SUMMARY

A well thought-out and highly effective one-stop shop for headphone monitoring and calibration.



on. This enables two separate modes of operation. You can use the ARC On-Ear as a USB DAC for your computer. In this scenario, your DAW software and operating system address the ARC On-Ear directly as an output device, and there is bidirectional communication between the hardware and IK's ARC On-Ear software, of which more presently. Connecting the USB cable to a computer or to a suitable power source will also charge its internal battery.

This battery enables the other mode of operation, whereby the ARC On-Ear is used as a fully standalone device. Audio is piped in through the Analog In mini-jack socket, which is a fixed-gain, line-level input that can accept a maximum level of +10dBu, and the unit is controlled from the three top-panel buttons. This might benefit any diehards still mixing in the analogue domain, and also makes it easier to integrate the ARC On-Ear with a larger audio interface.

Both the controls and the analogue input are always operational, so as long as you have a spare USB socket, you can still have the benefits of bus powering, hardware control and software integration while supplying audio from a line out on another interface. IK say that the ARC On-Ear will deliver around four hours' continuous listening on a single charge, depending on various factors such as playback level. On the output side, the ARC On-Ear uses a 'zero impedance'

headphone amp that will deliver 100mW per channel and a maximum output level of 19dBu. I found it plenty loud enough with all the headphones I tried.

ARC Curves

Once you've registered the hardware and installed IK's Product Manager, you'll in turn be able to install the ARC Studio software. This is a friendly, mostly single-window affair, and I found that almost everything about it was sufficiently intuitive that I didn't need to look anything up in the PDF manual. Which is lucky, because the PDF manual doesn't go into great depth about anything.

The ARC On-Ear hardware can store five Presets. A Preset is configured in the ARC On-Ear software, and is made up of several elements. First and most importantly, there's the choice of headphone model. As this is the first headphone correction product IK have made, I was expecting to find only a handful of headphones supported. Not so. Even at launch, ARC On-Ear includes well over 200 correction curves, for models from more than 60 manufacturers. It was enough to make me wonder whether IK had simply licensed these from another company, but I'm told that all of them have been measured in-house using a proprietary testing system. Moreover, the list is impressively current, with many brand-new models represented. At present, however, IK don't offer the option

▶ The ARC On-Ear software displays the uncorrected frequency response of the headphones in blue, and the corrected response in orange. The latter curve is not flat here because the speaker emulation is active.

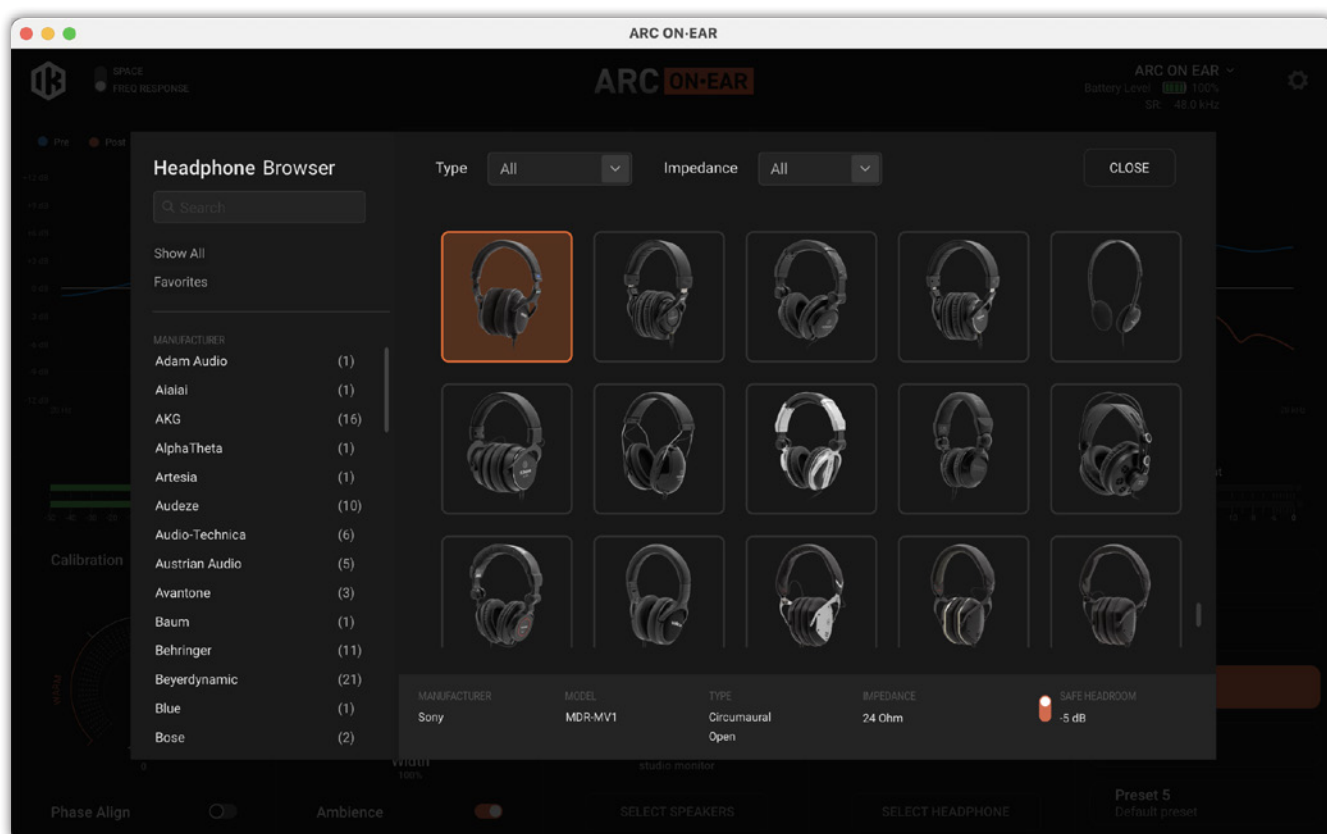
to send in your own headphones for custom measurement.

The correction curve for each set of headphones maps its measured frequency response onto a custom target, which is adapted from the widely used Harman curve. But that's not all. IK have also measured the phase response of each pair of headphones, and ARC On-Ear also offers the option to flatten this, at the cost of increased latency. Finally, each headphone curve also includes a Safe Headroom option that is engaged by default. This reduces the overall gain to compensate for the maximum degree of boost that is applied in the correction curve, to avoid clipping.

On The Lines

Within the ARC On-Ear software, a familiar EQ-type display shows the target response as a flat white line. Deviations from that target in the measured response of the headphones are illustrated on a blue curve, whilst an orange curve shows any deviations in the response of the corrected headphones. If you simply engage the correction and leave all the other settings in their default position, this too is a flat line. However, a rotary Tone setting can

»



ARC On-Ear ships with a library of more than 200 headphone calibration profiles.

» be used to apply a very gentle 'flat tilt' EQ centred at about 650Hz, providing a simple way to push the results towards a warmer or brighter tonality. This setting is reflected in the orange line, as is the effect of a three-band, user-configurable parametric equaliser. It might have been interesting to visualise the phase response, too, but this currently isn't possible.

ARC On-Ear doesn't stop at correcting the response of the headphones. It can also apply a 'studio simulation', either with or without the emulated response of different loudspeaker types. This comes with a Width setting that ranges from 10 percent to 200 percent, and a simple switch that toggles added ambience on and off.

The currently selected Preset is highlighted in orange on the lower right of the ARC On-Ear window, and by one of five LEDs along the side of the hardware unit. Any changes you make are automatically written to the active Preset, and there's no undo function. In practice, this is actually a positive, as it saves time, and the whole system is simple enough that there's no possibility of getting lost in a maze of edits. Presets can also be named in the ARC On-Ear software;

I would welcome the option to have the selected headphone model automatically chosen as the Preset name.

ARC Tangents

Of the three buttons on the ARC On-Ear unit itself, two have fixed functions and one is assignable. The function you give it is retained across all Presets. The Cal button engages and bypasses the headphone correction for the active Preset, whilst the Studio button does the same for the virtualisation. It's possible to assign the FN button to turn the speaker emulation on and off; if not, it follows the setting stored in the Preset. The virtualisation can be active regardless of the headphone correction, but you can't have the speaker emulation without the virtualisation, which seems logical.

Other possibilities for the FN button include mute, dim (by an amount configurable in the software settings), mono, phase correction on/off, LEDs on/off and to trigger and pause playback. Since only one of these can be chosen at a time, this does mean that there will always be settings that can't be accessed in standalone mode within a Preset. However, on the assumption that you don't need to store Presets for five different pairs of headphones, you could set up multiple

Presets using the same headphone curve and access them that way. Preset switching from the ARC On-Ear itself requires pressing the Cal button whilst holding down FN; what the manual doesn't tell you is that once in Preset selection mode, you then need to let go of FN and hold it again, whereupon Cal will cycle through the Presets.

Story ARC

In my experience, the correction curves produced by different room calibration packages tend to be pretty similar, assuming the same room and the correct measurement procedures. For example, IK's ARC Studio, Sonarworks' SoundID and Genelec's GLM all deliver much the same end result in my test space. This is very much not the case with headphone correction tools. Each of the major developers here has their own test procedure and their own target curve.

Consequently, there can be surprising differences between the calibration curves that different products apply. The Shure SRH1840 open-backed headphones provide a particularly extreme example. SoundID applies very little correction between 50Hz and 8kHz, whereas dSoniq's RealPhones recommends a broad cut from 1kHz to 5kHz and a pretty obvious

boost above this. ARC On-Ear's curve is different again; it too thinks the midrange is overblown, but the high frequencies even more so. Consequently, whereas SoundID does nothing much at 11kHz, RealPhones boosts by around 5dB and ARC On-Ear cuts by around 5dB! If you're thinking they can't all be right, that's obviously true, and in this case RealPhones is the outlier to my ears, making the 'corrected' SRH1840 sound significantly worse, whereas I quite liked what ARC On-Ear did.

The contrast is much less stark with some of the other headphone models I compared. For example, all three products suggest that the Neumann NDH 30s have a noticeable bump somewhere in the 2kHz region, although they disagree about exactly where it lies and how much correction is needed. If the examples I compared are representative, I'd say that RealPhones tends to apply the most correction and ARC On-Ear the least. However, RealPhones allows the degree of correction to be varied — I usually keep it around 75 percent — whereas ARC On-Ear does not have any sort of Amount control or wet/dry balance.

It's also clear that the target curves of the three products vary significantly at the top end, as do their approaches to calibration. RealPhones' idea of a calibrated monitoring system is significantly brighter than IK's or Sonarworks'. However, whereas SoundID profiles often include detailed notches and peaks in the treble region, ARC On-Ear tends to offer smoother curves.

Which of these curves sounds best will depend on many factors: personal taste, the specific model of headphones you're testing, how closely your own pair of headphones matches the generic curve derived from different samples, and so on. I tried ARC On-Ear with several different pairs. As always, it makes the most difference with headphones that are audibly not flat to start with, but in every case the calibration seemed to me an improvement, and in most cases a substantial one. I struggled to detect much benefit from the phase correction, but it certainly didn't make things worse, and the additional latency is not enough to be a problem at mixdown.

Subtle Space

If the ARC On-Ear calibration is relatively conservative, the virtualisation is even more so. You certainly don't feel as though you've been plunged into a different



One of five small LEDs down the left-hand side of the top panel lights up to show which Preset is active.

acoustic environment, and I never had that uncanny sensation of being fooled into thinking I was listening on speakers. In that respect, it's the polar opposite of something like Sony's 360VME. But in terms of utility, I found the studio simulation surprisingly useful. Whereas engaging bypass on 360VME can be quite disorientating, toggling the Studio button on the ARC On-Ear make just enough difference to offer a valuable alternative perspective. For this reason, I preferred using it with one of the high-end monitor emulations rather than a virtual pair of NS10s or similar.

Finally, these product comparisons ignore the ARC On-Ear's uniqueness. All the other dedicated headphone correction tools I know of are software-only, so unless you're using one of the relatively few audio interfaces that can apply sophisticated EQ to its headphone outputs, the ARC On-Ear is the only hardware option in its field. You can buy bus-powered digital-to-analogue converters that have DSP, but most of them don't have headphone outputs, and they certainly won't come with a library of 200+ calibration curves for different headphone models.

The flip side of this is that IK's headphone correction tech is presently not available outside of the ARC On-Ear box. There's no systemwide software with virtual drivers, or headphone equivalent of the ARC plug-in. How much does that

matter? For mixing and editing on the go, not at all. But in other scenarios, it ties you into using the ARC On-Ear alongside your main audio interface, which could be a nuisance. The analogue input offers an alternative to messing about with aggregating interfaces, but I can't help feeling that a digital input would have been a better option here. The analogue input is intended primarily for connection to the headphone output on another device. That might be the best way of hooking ARC On-Ear up to a mixer, but with an audio interface, it means passing the signal through two headphone amps and two stages of conversion. Most of us have a spare S/PDIF output somewhere on the back of our audio interface, and that would seem a better option to me.

However, that's really the only serious criticism of the ARC On-Ear system I can muster. In every other respect, I like this product a lot. The headphone calibration is effective, the virtualisation is useful, the companion software is intuitive and easy to use, and the huge catalogue of headphone profiles is impressive. In a world where you can pay a lot more for a supposedly audiophile USB DAC that lacks any of the same functionality, the ARC On-Ear also looks very good value for money. ■■■

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Dave Stewart

Producer & Songwriter

As one half of Eurythmics, Dave Stewart created a chart-topping blend of soul music and electronica. Now he's paying tribute to his earliest influences.

JOE MATERA

“When I was a kid, I had two Bob Dylan albums — *The Freewheelin’*

Bob Dylan and *Another Side Of Bob Dylan* — and I was just fascinated with the lyrics,” recalls Dave Stewart, co-founder of influential ’80s synth-pop duo Eurythmics. “And I used to sing these songs when I was about 16 years old in the folk clubs of my home town of Sunderland.”

Almost 60 years later, Stewart has paid tribute to this formative influence by recording an album of covers titled *Dave Does Dylan*, recorded in a minimalist fashion that retains the simplicity and directness of the originals.

“It depended on where I was,” explains Stewart. “Sometimes I’d be in a hotel room, sometimes in my home studio in the Bahamas, and sometimes in Nashville or sometimes, God knows where. We — Jesse, my engineer, and I — always take a couple of mics with us on the road so that we’re able to record into the laptop. So, I’d just sing the song live straight into the iPhone and I would post the recording on Instagram, and people would react to it and go, ‘I love that song and it sounds really good. Why don’t you put it out so we can get it?’ Then management said, ‘Why don’t we make some vinyl for Record Store Day?’ So, we did a vinyl run of about 3000 for Record Store Day.”

Stewart’s go-to microphones for the road weren’t extravagant by any means. “They’re not fancy or special but very reliable Shure mics,” says Stewart. “Basically, two Shure SM7Bs — one for the voice, and one on the guitar — and sometimes we used an Audio-Technica AT5045 on the guitar too, depending on where we were.”

When it came to guitars, Stewart’s choices were more varied. “Again, it depended on whatever guitar I had with me at the time,” he says. “One of them was an Australian-made miniature 12-string Maton which I used on the track ‘Shelter From The Storm’. I also used my blue Takamine that I originally bought back in the ’80s, as I was using it for concerts and we’d have it

— Dave Stewart performing songs from *Dave Does Dylan* on PBS TV.

in my hotel room. I also used this beautiful limited-edition Martin that John McBride, who owns Blackbird Studios in Nashville, gave to me for my birthday. It is made out of the wood of a ship that sank like 200 years ago. Martin made a very small amount of these guitars out of this wood. Also, an old 1948 Gibson LG-2 parlour guitar. I have around eight guitars in my studio in the Bahamas, and about 16 guitars in my little home studio in Nashville.”

Tour Management

Dave Stewart first appeared upon the scene in the late 1970s with pop-rock group the Tourists, fronted by his then girlfriend and later Eurythmics partner Annie Lennox. The group released three albums in quick succession: their self-titled debut in 1979 was followed later that same year by *Reality Effect*, yielding two UK top 10 singles with ‘I Only Want To Be With You’ and ‘So Good To Be Back Home Again’. *Luminous Basement* appeared in 1980 before the group disintegrated while on tour in Australia. It was while they were licking their wounds and contemplating an uncertain future in a hotel room in the Australian country town of Wagga Wagga that Stewart and Lennox stumbled upon an idea that would alter their lives.

“I’d brought in my suitcase this tiny little miniature sort of plastic synthesizer called an EDP Wasp,” recalls Stewart. “You couldn’t press a keyboard down and it was like a Stylophone and was black and yellow like the colours of a wasp. But what was great about it was that you had the basic synthesizer controls where you could change the sound. And at first, I was trying to make it sound like a didgeridoo. Then I started playing just very simple bass lines on it. As Annie and I were in the hotel room, Annie started singing along a bit to what I was playing. We were just messing around but we kind of vaguely had this idea that just the two of us writing songs could be quite good. When we got back to England we decided to pursue the idea. We’d been living together for about four years, in a squat at first, but now were living in this apartment. So, all we ever did was see each other and talk to each other 24 hours. It was just us two. We started to get *folie à deux* so we decided we should live apart — so Annie moved upstairs, and I downstairs.”

In early 1981, both Stewart and Lennox headed to Cologne, Germany to begin the recording process for the Eurythmics’ debut album *In The Garden* with German producer Conny Plank, who had worked



— Dave Stewart and Annie Lennox in the studio, 1982.

Photos: Peter Anderson

with leading Krautrock groups such as Neu!, Cluster and Kraftwerk. By this point Stewart had added further gear to his sonic arsenal, and working with Plank gave Stewart an education in the art of recording.

“I had gone to a music shop and got a little bit more additional gear for the Wasp,” he says. “I got a Caterpillar master keyboard and a Spider sequencer and basically when I wired them together, I could start sequencing the Wasp and it was really powerful-sounding. So, I got really into that and then I got a couple more other things:

a Roland TR-606 Drumatix, which is another little drum machine, and a Roland Space Echo, and I was making all this kind of weird experimental-sounding stuff.”

Dreamscapes

Released in October 1981, *In The Garden* featured a supporting cast of musicians that included members of Krautrock pioneers Can and Blondie drummer Clem Burke. It didn’t meet with much commercial success, but its atmospheric soundscapes and experimental spirit hinted at what was to

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■ The makeshift studio setup that was used to produce *Sweet Dreams (Are Made Of This)*: to the right of the mixing console are a Lexicon Prime Time delay and Roland Space Echo, and Dave Stewart's prototype drum computer.

» come. For the 1983 follow-up *Sweet Dreams (Are Made Of This)* Stewart and Lennox relinquished the musicians, preferring to remain a duo. Holing themselves up in a makeshift studio located in a picture factory attic, they went to work on what would become their signature sound.

"On the *Sweet Dreams* record, you can hear it's me in the bedroom with my gear playing the bass lines on a Roland SH-101 onto a four-track cassette TEAC 144 Portastudio, like on 'I've Got An Angel,'" he explains. "Later on, when we got an eight-track, we bounced that four-track onto the eight-track. *Sweet Dreams* is very unique-sounding because it was almost like soul music meets a very Germanic kind of electronic Kraftwerk-y kind of thing. It wasn't like New Order's 'Blue Monday'. But on the other hand, it had elements of this dystopian kind of emptiness to it that left a lot of space for Annie's vocal. That was one of the things I really focused on when I first began producing. The first thing I ever produced was a Tourists song called 'The Loneliest Man In The World' where I left a lot of space as well, but I was only given two hours in the studio to do it, and I had everything, like tubular bells and all sorts of things going on.

"A lot of it was trying to create as much space as possible so that Annie's voice sounded crystal clear, pure, and not drowned out. But making sure the backing

track still sounded incredibly powerful. A lot of that has to do with literally not having anything in the stereo picture on top of each other, as well as looking at the difference of frequencies and not having the cymbals or the hi-hats from a drum machine be on the same frequency as, say, an electric guitar within the limited tracks we had.

"We only had seven tracks to work with, because the eighth one was running a sync tone into this very strange prototype of a drum computer. And so those tracks, if you played them back, it's really hard to make sense of them because you'll hear on one track, it might have Annie's harmony of a backing vocal before it suddenly turns into a sequencer playing."

Pulling The Strings

The album's celebrated title track owed much to an illicitly borrowed instrument. "The guy who owned the picture-framing factory had this expensive Oberheim OB-X synthesizer with a cover on it, and we were told not to touch it. When I started the rhythm of 'Sweet Dreams', Annie was very depressed at that point, lying on the floor saying, 'Oh, nothing's going to work'. She was waiting for me to get this drum machine working because it was a prototype, but it was very unusual and she was really feeling like giving up. And all of a sudden, I got it working and I put in this downbeat, which was a mixture of

a tuned-down tom-tom on top of a bass run, and then a four-on-the-floor on the bass and she suddenly leapt up and was like, 'What's that?' It was just so infectious.

"And then we said, 'Ooh, let's just take the cover off this synthesizer and switch it on. He won't mind.' And we did, and we switched it on and it had that exact string sound without programming anything. It must've just been left on that sound. Maybe it was some preset sound or something that was in it. So, Annie started playing along with my sequence and it was just so infectious and powerful that we were both jumping around. Annie went from laying on the floor completely depressed to sort of celebrating. As she started singing 'Sweet dreams are made of this,' I quickly got a mic. We only had one beyer stick microphone — the kind of mic you used for hi-hats and things like that. And all of Annie's vocals were done with that mic.

"When we were mixing the album, we had to have little marks on the board so that myself, Annie and Adam Williams — a great bass player from the band the Selector, who helped put that studio together for us, and was also great at engineering — had to write all sorts of marks at various points in the songs. We had to mark where something might quickly move from there to there and that from there to there and that from there to there, like you would with automation today, but there was no

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new



— Dave Does Dylan was recorded very quickly, using two Shure SM7B dynamic microphones.

» automation back then. If you didn't do that, suddenly a sequencer would've come blasting through a harmony track.

"Normally we would get it right in about the third or fourth take. We'd rehearse the moves and then press record and print it onto a Revox two-track reel-to-reel tape recorder that we stole from the record company, because we may have had an eight-track, but we had nothing to then get it off the eight-track. We used the tape already on the machine, and printed the album on top of it because

we didn't really have much money to keep buying tapes, so God knows what we recorded over! We were just recording on top and wiping out stuff."

Private Affair

Sweet Dreams (Are Made Of This) would set the course for the group's upward trajectory to becoming one of the biggest synth-pop acts of the '80s, with a sound that was uniquely their own. "We were determined and desperate to make a new mark, and forge a new identity in music

and for ourselves," admits Stewart. "It was all about the things that we loved, put into one thing. In order to achieve this, we made a board and I would write in a line down the board on one side our likes, and on the other side, our dislikes, just in case we wandered into doing things we didn't really like. On the 'like' side we'd have written things like, 'electronic', 'soul', 'Motown', 'Stax'. We'd look at that and go, 'Well, the vocal can be like some soul thing, and the backing can be electronic,' and we kind of stuck to that format."

The Eurythmics writing process was always a collaborative effort between the pair. "It was just always the two of us in a room. We didn't like anybody else being there while we were doing it. And because we'd lived with each other, we knew each other really well and we could say anything, like, 'Oh, that's not going to work.' I'd usually start these kind of soundscape music tracks, whether it's 'Sweet Dreams' or 'There Must Be An Angel' or 'The Walk' or whatever. And Annie would have a journal that she was writing, though not particularly song lyrics; sometimes it was just rhymes, sometimes just sentences. And then she would always let me have a look at the journal and I would be thinking about the rhythm of the track and what was going on. And then I would pick out a line or maybe four lines and say, 'This is an amazing title for this and that.' And we'd get inspired about it, think about it and then start singing along with the backing track and then it starts to become a song.

"We worked together like that for like eight albums in a row, and it became a very quick process after *Sweet Dreams (Are Made Of This)*. While *Sweet Dreams* was a lot of experimentation, with *Touch [1983]* it took just three weeks of writing and recording. And the same with *Be Yourself Tonight [1985]* and *Revenge [1986]*. We'd sort of created a way of working and a sort of template for ourselves. Even though, on *Revenge* for instance, I realised that we were now playing in arenas and big places, and so made songs that would possibly fit in those arenas, like 'Missionary Man' and things like that, it was still the same kind of process.

"To this day, I don't make any demos, and never have. If I had something on the Portastudio, it was bounced onto the eight-track and that was part of the master. And whoever I work with, I will not make a demo either. I think when you make a demo, often you capture some spark of

brilliance and then you try and recreate it again, but it starts to fade away. I might then add to it, change it a bit, edit it about, but the thing that first came out is always in it. The record label often will say to me, 'Do you have any extra demos that we could put as special tracks?' And I'm like, 'No, we just have other songs!'"

Further Explorations

In 1990, the duo decided it was time take a break from what seemed to have become an endless treadmill of recording and touring. Not one to remain idle, Stewart was soon busy with new projects. "I sort of decided I'd done so much music, I needed a break from music," he says. "Instead, I went to New York and made this solo album called *Greetings From The Gutter* [1994]. And then I went to Jamaica and recorded another solo outing, *Sly-Fi* [1998], in this little house I had there by the river. And it was the first album ever released on the Internet, before David Bowie. They say that Bowie was the first guy to release an album in 1999 [*hours...*] on the Internet, but he wasn't, as that was already released through an online music services company called N2K.

"And then I didn't make any records for about 15 years, until I ended up in Nashville.

And I made *The Blackbird Diaries* in 2011 and then literally a trilogy of albums in two years. Each of those albums was made in about five days; writing the songs, and recording them. And then I didn't make any albums until this Dylan one."

Eurythmics reunited in the later part of the '90s before Lennox retired from the road in the '00s to focus on other projects, though she and Stewart have since performed a handful of one-off special shows, such as Sting's 30th anniversary 'We'll Be Together' benefit concert in December 2019. In 2024, Eurythmics returned to the road without Lennox, her position being filled by Australian singer Vanessa Amorosi. "Vanessa and I had been messing around playing and writing songs just as friends since she first came to Los Angeles about 14 years ago," says Stewart. "She's incredible and one of the greatest voices I've ever recorded."

Rain & Shine

When asked if there's any particular favourite song or album in Eurythmics' body of work, Dave Stewart chooses 'Here Comes The Rain Again' and its parent album *Touch*. "When we came to make that album I thought we couldn't just stay stuck in the same *Sweet Dreams* world; we had

to branch out a bit. So, I got myself this big Gretsch Country Gent guitar, brought in a whole orchestra, got a sequencer and another little high sequencer and a bass sequencer, and it was like this mad mixture of sounds. And it captured again this sort of melancholy atmosphere with a very hopeful chorus, which is one of the signatures of Eurythmics music. In the verses it might sound like, 'Oh God, this is so tragic', but once you get to the chorus, some major progression comes in and the lyrics are slightly altered to make it feel like you're not going to slit your wrists.

"If you analyse just about all the songs, there's only a couple that actually make you feel really up all the way through, like life's great. And one of them, 'Right By Your Side', even the chorus goes, 'cause when depression starts to win / I need to be right by your side'. It's always got that little twisted thing going on, and I think that's what keeps your music current. It's not all fluff and Disney and it's not all completely the end of the world, yet it's got this funny mixture in the middle somehow. To me it is reality. Reality isn't all damn terrible and it isn't all great. It's up and down every hour or every day. You don't know what's going to happen next — but you're hoping for the best." ■■■



■ Eurythmics' sound developed through the '80s.



SAM INGLIS

Very occasionally, a new product creates a real buzz within the industry. Word spreads in trade-show corridors and dealer showrooms, and suddenly everyone is clamouring to hear this new phenomenon. Sometimes it turns out to be a damp squib. Sometimes, as in the case of Celemony's Melodyne all those years ago, it's the start of something very special. It's been a while since anything else had that sort of impact, but the jungle telegraph has definitely been sounding about Sony's 360 Virtual Monitoring Environment.

On the face of it, that might seem odd. After all, isn't Sony 360 Reality the 'other' immersive format in a market dominated by Dolby Atmos? And aren't there plenty of other products on the market already that promise to deliver immersive virtual control room environments on headphones? Well, yes, and yes — but also no.

First of all, 360VME is entirely agnostic about, and independent of, mix and delivery formats. It can be used to mix material in Sony 360 Reality, but it's equally capable of supporting Atmos, 5.1 and indeed stereo. And second, whilst there are other products that virtualise control-room monitoring, 360VME has a unique selling point. To

Sony 360VME

Virtual Immersive Monitoring System

Sony aren't the first company to promise believable surround sound on headphones. But are they the first to deliver it?

explain the significance of this, it's worth a quick delve into how these products work.

Acting On Impulse

No matter how many speakers there are in an immersive setup, a human listener has only two ears. These can be considered single-point receivers, like microphones. Thus, in principle, it should be possible for an impulse response to perfectly describe how the sound from any given speaker is reflected, refracted, absorbed and otherwise coloured on its way to each ear. And if we can generate a set of impulse responses mapping each speaker onto each ear, we can recreate the sense of listening in a 3D space on headphones.

In practice, there's a problem that limits how faithfully this can be done. The most important source of coloration in this context is the physical geometry of our own head, torso and outer ear. Our

perception of direction is derived from the filtering effects introduced by these structures, so in order to 'binauralise' headphone sound, the filters we apply need to perfectly describe the effects of the listener's head geometry. Considered independently of the room and speakers, these are known as the head-related transfer function or HRTF.

The limiting factor is that no two people have the same head, ear or torso shape. So in order to create a viable commercial product, manufacturers need to either work out a practical way of measuring the HRTF of the person who's going to use it, or choose a generic HRTF and hope that it will be close enough. In the past, the latter option was the only feasible one, but in recent years, we've seen several attempts to derive personalised HRTFs from visual sources. Genelec's AuralID, for example, can do so by analysing a video of your head,



whilst Embody's Immerse Virtual Studio uses just a photo of the ear.

Once the HRTF has been calculated, it can be superimposed on the characteristics of a control room to create a virtualised listening experience. In some cases, such as AuralID and APL Virtuoso, these characteristics are derived from first principles, to create an 'ideal' control room. In others, including Immerse Virtual Studio, they are measured in real-world control rooms. Either way, there is no such thing as a completely neutral set of headphones, so it's often desirable to add further equalisation to compensate for the headphone frequency response.

Profile Pictures

What's different about 360VME is that it uses a single set of measurements to capture the characteristics of all three elements: the room and speakers, the headphones, and the geometry of the ears and head. It's not just creating a personalised HRTF and applying that to a generic room simulation, or an impulse response recorded separately. It's measuring the entire system, with the listener in place.

In order to use 360VME, you will need a file that Sony call a Profile. This is based on measurements taken in the space you wish to replicate, and needs to be done

by an authorised representative of Sony. Because your ears and the room are measured together as a system, it's not possible to separate them, so you can't for instance load up an emulation of Abbey Road's control room instead of your own. You'd have to make arrangements with Abbey Road to visit and have your own Profile measured *in situ*.

Back in June 2023, I reviewed Sony's first ever open-backed headphones, the MDR-MV1, and was a little mystified by the marketing, which made a big deal out of their supposed suitability for use in immersive mixing. It seems this may have been part of a wider plan, because 360VME currently requires either the MDR-MV1 or the closed-back MDR-M1; the former are recommended, and were used for this review. Typically, when you adopt 360VME, you'll be buying a package that includes a pair of Sony headphones, a one-off fee to cover the cost of the Profile measurement, and a monthly or yearly subscription licence. We're talking professional rather than home studio pricing here, but the same is true of AuralID and some other competitors.

To get started on this review, I visited the London offices of UK dealers HHB, which houses an immersive demo space with a 7.1.4 Genelec Atmos setup. The measurement process begins by adjusting the headphones to get the most comfortable fit. You then remove the headphones and fit tiny omnidirectional mics in your ears. These are held in place by conventional earhooks and spiral wire mounts, and feed a custom preamp that is connected to the measurement system. Once the mics are in place and you're sat in the sweet spot, you'll hear sine sweeps sent through each speaker in turn. Finally, you put the headphones back on again, and the system sends a sine sweep through each earpiece. The entire process is surprisingly pleasant, quick and easy — much more so, for example, than being measured for custom in-ear moulds.

Plain Panes

The 360VME software is currently available only as a standalone program for macOS. Like most such applications, it installs a virtual soundcard driver, which presents 16 channels to your DAW or chosen streaming application. Its two-channel output can then be routed either to the Mac's headphone out or to an audio interface of your choice. A certain amount of mucking around with permissions in System Settings is necessary to get any

Sony 360VME

PROS

- Convincing recreation of immersive speaker listening on headphones.
- Simple, fast measurement process.
- Easy to set up and use.

CONS

- It's only as good as the room and speaker system it measures.
- No plug-in or Windows versions.

SUMMARY

As long as you have access to a very good-sounding control room, 360VME is a step forward in allowing you to work in immersive audio on headphones.

sound out of it, but once you've overcome that hurdle, it's fairly straightforward.

There's something rather early-2000s about the 360VME interface, which is rather small and can't be resized, but it's clear and straightforward to understand. I encountered a couple of minor bugs, such as a tendency to throw a strop when other audio devices were connected, and it occasionally swapped the headphone channels over for no obvious reason, but for the most part it works as advertised. The upper pane is where you configure audio routing, meter input signals and mute or solo virtual loudspeakers. The lower has three tabs labelled De-reverb, Monitor EQ and Panning, which allow you to customise the sound to suit your taste.

You might ask why this should be necessary, if a 360VME Profile is such a perfect representation of the system that was originally measured. There are, I think, two reasons. One is that there's no such thing as a perfect immersive monitoring system in the first place, and the other is that the experience of listening on headphones still differs in some important respects. In particular, 360VME doesn't incorporate motion tracking, or sample the room at different positions. HHB's demo suite is impressive, but it's not a purpose-built Atmos room, and I preferred the results I got from my Profile by engaging a little De-reverb and applying a gentle high-frequency shelving cut.

Deep Fake

So why is 360VME creating a buzz? In part, I think, because the initial demo that people experience is so impressive. Once your ears have been measured, you'll get a chance to A/B some material on speakers and headphones; and when you're sitting in the space that has been measured, with all the same visual cues, the illusion is absolutely

»



The measurement process involves having small microphones inserted in each ear.

» irresistible. Indeed, multiple people have told me that they've been completely fooled into thinking they were listening to speaker playback when in fact the sound was coming through the headphones. The immersive mixes I heard at HHB's studio really did appear almost identical on speakers and headphones, with the only obvious giveaway being the lack of physical presence from the low bass.

The question, then, is whether 360VME can maintain that believability in other contexts. When you're looking at the back of an airline seat or the wall of your home office, will it still feel like you're listening in a high-end control room? And the answer is yes; or, at least, more so than I've experienced before.

I've tested most of the other immersive virtual studio products on the market at present, and they're all helpful in their way, but inasmuch as they create a believable surround stage, it's only up to a point, and mostly in two dimensions, and sometimes with audible artefacts. To a certain extent,

moving sources can appear to be above or behind you, but this is much less the case with static sources, and I've certainly not heard another system where you can solo, say, the Left Top Rear speaker and really feel as though the sound is coming from over your left shoulder. With 360VME, you absolutely can. Stable, believable imaging around the front and left/right surround speakers is familiar from other products. Stable and believable imaging above and behind you is not.

Moreover, 360VME also works in stereo. If you feed it a two-channel signal, what you'll hear is how that would sound played back over the left and right speakers in the measured room. Once again, the illusion of 'being there' is really very impressive, although I was surprised at how bright my familiar reference tracks sounded with stereo material. The timbral difference when you click the Bypass button is certainly pretty stark, in part because 360VME is no longer compensating for the bump in the MDR-MV1s' low midrange. Stereo listening

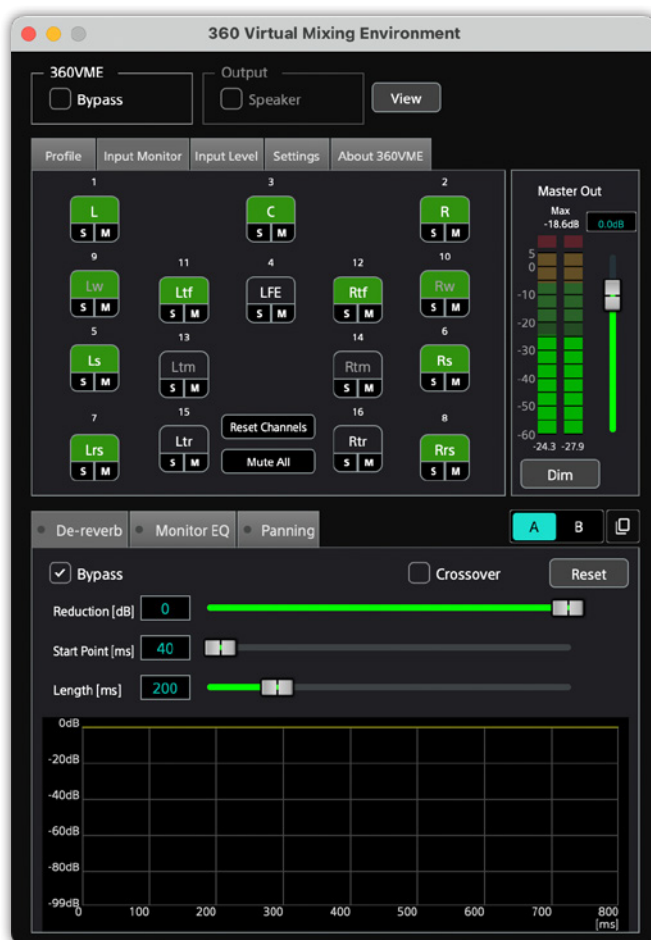
also exposes some grittiness in the upper midrange that wasn't present on the original speakers. Even in surround, I actually found music more enjoyable to listen to in bypass, although that's hardly the point. In stereo, I think 360VME would still be a useful mix referencing tool, but it does more to earn its keep if you work in immersive audio.

360 Deals

Not having an immersive speaker setup myself, I've always felt that virtual monitoring software is probably most useful to those who do. Nothing I've tested has made me confident that I could undertake an Atmos or 360 Reality mix entirely on headphones. Rather, I think the value of products like AuralID, Virtuoso and Immerse Virtual Studio is that you can take away projects started on a speaker-based system and continue to work on them in a somewhat immersive context until you can next get back into the control room. In terms of confidence, Sony's 360VME is a clear step forward from this.



The 360VME software is currently available only as a standalone app.



The Input Monitor pane shows activity in each speaker channel, and allows you to mute or solo individual speakers.

It is, however, designed to piggy-back on an existing speaker-based monitoring arrangement. There wouldn't be much point in buying 360VME unless you have access to a very good immersive control room within which your Profile can be measured. And although its virtualisation is a big step up, I think it's still likely to be most valuable if you can use it in addition to, rather than instead of, that control room. The ideal client for 360VME is perhaps a large post-production house or university, who can build out one really good immersive mixing room and then acquire 360VME Profiles for all its staff or students.

In some ways, 360VME is less versatile than other software-based solutions. You can't swap between multiple virtual environments, or synthesize speaker arrays that weren't present in the space that was measured for your Profile. But the difference in realism, to my mind, easily outweighs these drawbacks. If anything, the main rival for 360VME is not AuralID or Immerse Virtual Studio or

Pricing

The 360VME system comprises three elements: the headphones, the Profile measurement and a yearly subscription. Sony strongly recommend the open-backed MDR-MV1 headphones, which officially retail at £339, though the closed-back MDR-M1 (£209) are also supported. The cost of having your Profile measured and the yearly subscription will vary with circumstances, but will typically be several hundred pounds. All prices include VAT.

Virtuoso, but the long-established Smyth Research Realiser A16. This hardware-based system was reviewed by Mike Senior back in 2013, and has continued to evolve since. It works on the same basic principle as 360VME, whereby measurements are taken with mics inserted into the listener's ears, but is designed for self-operator use, and can be supplied with everything you need to make your own measurements.

Unlike 360VME, the Realiser is targeted at consumers as well as producers. It offers features such as head-tracking and online sharing of measurements, and supports any good-quality headphones. However, it is only available as either a 2U rackmount unit or a fairly large desktop

device, and either of them will set you back several thousand pounds. It's not something you can load up on your laptop when you need to work on the go. In comparison, 360VME is a lean, focused solution that targets a very precise use case, and I wouldn't be surprised if even die-hard Smyth Research fans end up buying this as well. For the rest of us, 360VME is currently as close as you can get to recreating speaker listening on headphones. **///**

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BOB THOMAS

Founded in California in the mid-1960s, Audio Engineering Associates are more familiarly known by their initials AEA, and best known for their range of high-quality, handcrafted ribbon microphones and preamplifiers. Their involvement in this area dates back to 1976, when they took over the repair of RCA's ribbon mic range. 1998 saw the release of the first AEA mic, the R44C, and 27 years on AEA still produce it — but their range has grown considerably since then, to include 10 ribbon mics, five mic preamps, and two phantom-powered, active DI boxes. Now, the company have moved further along the signal chain, with the recent release of the AEA 1029, a stereo VCA compressor.

First Look

The AEA 1029 takes the form of a 1U half-rackmount device, and shares the classic aesthetic of its preamp stablemates: a dark grey chassis and fascia; white lettering and scaling; a circular, retro-styled red and silver company logo; and AEA's signature tall, knurl-tip silver knobs with black position indicators. Despite being small, the knobs' combination of knurling and narrow diameter ensures

AEA 1029

Stereo VCA Compressor

The debut compressor from ribbon-mic legends AEA is a classy and versatile affair.

that the controls feel nicely spaced, even to my large hands. Depressing the Compressor In button activates relays that switch the compressor out of a hard bypass. An associated green LED illuminates to indicate that the compressor circuitry is active.

The compression characteristics are set by a three-position mode and timing toggle switch (Peak Fast / RMS / Peak Slow), and continuously variable rotary threshold and ratio controls. Up to 20dB of make-up gain allows you to compensate for any loss of level due to compression, and a dry/wet control encourages you to explore parallel compression. Below/between the threshold and ratio controls, a red LED illuminates when the set threshold level is exceeded and gain reduction occurs.

Three horizontal LED ladders indicate the level of gain reduction (six segments, 1-12 dB), and of the summed input and summed output signals (seven segments, -20dBu up to clipping). Finally, there's a comparatively large red power on/off button, and a green power indicator LED. The 1029's rear panel carries the 5-pin XLR connector for the external switching power supply, and balanced XLR I/O connectors for each channel.

Lifting the lid of the AEA 1029 reveals a single, beautifully laid-out circuit board that hosts all the componentry and each channel's Fred Forssell-designed discrete op-amp. The majority of components, even on those op-amps,

are surface-mount types, though a number of full-size, electrolytic capacitors are employed in the audio circuits.

The stereo-linked compressors feature THAT Corporation's 4305 Analog Engine single-chip dynamics processors, and the 4305 itself has two main parts, an RMS (Root Mean Square) level detector, and a Blackmer voltage-controlled amplifier (VCA). Both the RMS detector and the VCA are pre-trimmed during chip manufacture to deliver low distortion (typically 0.07% THD).

Current Thinking

Since the 4305's VCA acts on current, not voltage, the L-R voltages of an incoming stereo signal are first converted to currents, which are then routed to the input of their respective channel's 4305. In parallel, a summed tap of those currents is sent to the RMS detector of one 4305. This routing configures the 1029 as a feed-forward, true-stereo compressor. The RMS detector computes the RMS level of the input current signal by rectifying it, converting the result to a logarithmic voltage, and then applying that voltage to a log-domain filter. That filter's output is a DC voltage that's proportional to the RMS level (in dB) of the input signal current. This voltage, modified by that single 4305's side-chain controls (mode/time constant, threshold, ratio and make-up gain), is then sent to each channel's 4305 VCA to control its output current level.

AEA 1029

£1499

PROS

- Superb audio performance.
- Intuitive operation.
- Excellent value for money.

CONS

- Absolutely none!

SUMMARY

A high-quality, handcrafted, true-stereo peak/RMS compressor that delivers a superb level of performance at an attractive price point.

The idea of all that is that, since the VCA gain (in dB) is proportional to the control voltage, and the control voltage is proportional to the RMS level (in dB) of the input signal current, changes in the level of the VCA's output signal current reflect the way in which we humans perceive loudness. The advantage is to create a more natural-sounding gain control, with high levels of precision and stability over a wide dynamic range.

For the RMS mode, the control voltage is derived from the mean average of both the positive peaks and negative troughs of the incoming current's amplitude. In Peak mode, on the other hand, the control voltage is derived only from the peaks. The 4305's RMS detector can be easily reconfigured to act as a peak level detector by changing the values and number of certain capacitors around the chip. In the 1029, a DPDT (Double Pole/ Double Throw) switch and an array of solid-state opto-relays is used to carry out the necessary changes to toggle between RMS and twin Peak modes.

Since the 4305's VCAs output currents, a pair of discrete op-amps then act as current-to-voltage converters, creating the signal voltages necessary to drive the electronically balanced L-R outputs. Those op-amps are designed by Fred Forssell, who has a 50-year-plus career designing, making and selling high-performance audio equipment in the USA, and has played a key role in designing not just the 1029 but the entire range of AEA mic preamps. His proprietary low-noise, low-distortion op-amps for the 1029 feature complementary NOS Toshiba PNP and NPN JFETs, and are designed to deliver current-to-voltage conversion of the highest quality and precision.

In Use

Unusually for a compressor, in my experience at least, no signal is shown on the 1029's input and output meters when it's in bypass. In practical terms, this is no big deal, but it does require you to have the compressor active if troubleshooting

or gain-staging your signal chain. There's no shortage of headroom, as the clip LEDs on the input and output meters come on at +22dBu, 4dBu before the actual clip points of +26dBu. Like me, then, you shouldn't have any issues integrating the 1029 into your workflow.

As you might expect given the headroom, the frequency bandwidth specifications (dry: 10Hz-200kHz +0/-0.05 dB/wet: 10Hz-200kHz +0/-0.1 dB) and its signal-to-noise ratios of -97dBu (dry) and -91dBu (wet), the 1029 delivers a performance that is clean, open and transparent. Whilst the 1029 doesn't offer the sonic character of, say, a vintage-style valve or optical compressor, it has its own distinct qualities. In all three operating modes, the 1029 is impressively precise in the way it handles peaks and transients, particularly at lower frequencies. To my mind, the transparency and precision of this sonic performance may have something to do with the Forssell op-amps that drive the outputs.

The continuously variable threshold (± 20 dB), ratio (1:1 to Limit) and up to 20dB of make-up gain allow you to explore and exploit the 1029's compression modes in detail. The Peak Fast mode (4ms attack, 17ms release) is adept at controlling the peak transients of drums and percussion. At higher ratios and thresholds, this mode can crush these sources into ideal candidates for wet/dry parallel compression. Peak Slow (4ms attack, 200ms release) could well become the perfect partner for a stereo-miked double bass, but I also found it very effective on the spoken word.

For vocals, whether male or female, the 1029's RMS mode (12ms attack, 50ms release) worked superbly, its strong and subtle overall control remaining transparent, smooth and virtually inaudible, until I started pushing past a ratio of 4:1. RMS mode also performed superbly on more complex stereo sources, with the sort of transients you'll find in acoustic pianos, guitars, bowed instruments and even, dare I say



■ The dry/wet mix control makes parallel compression easy — and with the superb transient control of this compressor, it can be really effective.

it, my banjos. For my applications at least, a 1029 would probably spend the majority of its time in RMS mode.

Final Thoughts

At first, given the simplicity of its control layout and the fixed attack and release times of its various modes, I suspected that the 1029's compression options might be a bit limiting (if you'll pardon the pun). But first appearances can be deceptive, and my misgivings proved to be groundless. The time constants have been very carefully chosen for use in the real world, which implies much thought, consultation and experimentation in their selection. Once I'd become familiar with the extent of its capabilities, working with the 1029 on any particular source became a hugely enjoyable and intuitive exploration of the wide range of possibilities offered by the multitude of potential combinations of its mode/time constants, threshold levels, ratios and wet/dry mix levels.

After many years of multitracking, I've recently begun to return to my roots in old-school analogue recording, using a stereo microphone and a stereo mic preamp direct to two-track tape. The AEA 1029 has already proven itself an ideal travelling companion on this journey. If you're looking for a high-quality, true stereo compressor that offers excellent value for money and a superb level of performance, you really should give the AEA 1029 a try. ■■■



■ On the rear, as well as the balanced I/O on XLRs, there's a 5-pin socket for the external PSU.

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Eventide Temperance Pro

Reverb Plug-in

Thought you knew reverb? Eventide's unique new plug-in might just make you think again...

PAUL WHITE

Available in two versions, Lite and Pro, Eventide's Temperance takes reverb plug-in design in a new direction, not only allowing its simulations to emulate realistic spaces as you might expect, but also to function almost as part of an active, musical instrument. Traditional reverbs replicate how sound diffuses in air and bounces off walls, either by capturing IRs, or with synthetic reverb algorithms based on FIR filters (multi-tapped delays), using a mixture of comb and all-pass filters to recirculate the sound. By contrast, Temperance operates in what Eventide call "the modal domain", where each space is represented by a very large number of differently tuned resonators. Eventide describe the process as the resonators behaving rather like tuning forks, each one tuned to a specific frequency. This suggests some parallels with Zynaptiq's *Adaptiverb*, which creates the reverb tail using a large number of oscillators whose respective levels are influenced by the spectrum of the incoming sound.

Temperance Tour

All the controls present in the Lite version of Temperance are also present in the Pro version, so I'll start off by describing the Lite version — at the time of writing this one is available for free — and then move on to the added extras of the Pro version. This plug-in exhibits very low latency and combines the natural sound of modal modelling with intuitive controls that enable the user to easily explore its creative potential. Fifteen factory presets, including five tutorials, are included.

When an audio signal is fed into the plug-in, it excites the resonant modes, just as it would in a real space, and each mode has its own resonant frequency, decay time, amplitude and phase. This is what creates the familiar wash of reverberation. But within Temperance, each resonator



Temperance Lite (left) is free for a limited time, but the Pro version (right) offers much deeper control and more extreme effects should you want them.

can be individually manipulated in real time — and to make possible musical treatments that stray well away from natural reverb, Eventide have tied this in with the 12-note chromatic scale.

The large Temper knob determines how strongly the plug-in emphasises or suppresses resonant modes corresponding to musical notes set on the chromatic scale, and the user can click on notes to turn them on or off. Set midway, the reverb behaves like a conventional space creating a smooth, natural-sounding reverb tail. Turning the knob clockwise increases the strength of resonances corresponding to the selected notes displayed around the control, allowing those frequencies to create a reverb that sounds like it's in harmony with the input signal. Turning it anticlockwise de-emphasises the chosen notes, allowing the others to come through more strongly.

At the bottom of the GUI are range sliders, a pair of which define the high- and low-cut filter corner frequencies, while the other pair set the upper and lower frequencies affected by the Temper control. To help visualise what's happening, the display shows the reverb energy around a one-octave circle showing how the energy level moves in response to the input. There's a further display within the filter section at the bottom of the window that shows spectral details of the reverb.

All the familiar reverb adjustment parameters are present: Mix, Decay, Size, and up to two seconds of Pre-Delay. There's

also a choice of Bright Room, Large Studio or Synthetic Space. Bright Room derives from Eventide's well-known SP2016 algorithm, whereas Large Studio is modelled after an impulse response provided by acoustician Dr Ralph Kessler. Synthetic Space is a purely in-house design. Finally on the control front, there's an output level adjuster.

As Temperance Lite allows the reverb tail to respond musically, it can be tuned to the key of a song, which can reduce or remove dissonant overtones, and there's a tuning adjustment to adapt to pitches other than the standard A=440Hz. At more extreme settings, the reverb tail can sound like an evolving pad or an ambient haze, in which only specific intervals 'hang on'.

Getting Heavy

Temperance Pro, of course, does all of the above but more besides. It comes with 29 different reverb types, encompassing a range of sizes plus some synthetic options, and it has an input level control with an associated Dry Kill switch, as well as an output level control with a Dual Mono mode button below. Dual Mono mode applies the same processing separately to both channels, and can help maintain a strong stereo image, especially at lower Density settings. The maximum pre-delay time is two seconds, which is the same as the Lite version. Note selection can be handled manually (as with the Lite version), but there's also the option to enter by Interval, Scale, Chord or Mode.

Eventide Temperance Pro \$179

PROS

- A genuinely original take on reverb.
- Really musical character.
- Capable of subtle and extreme effects.
- Built-in sequencer.

CONS

- None.

SUMMARY

A rare case of a plug-in effect that genuinely breaks new ground: Temperance is capable of subtle or extreme effects, and delivers a particularly musical take on reverb.

Interestingly, there's a built-in sequencer of up to 64 bars, and the user can enter up to eight different chords to the sequencer steps. Furthermore, notes can be actioned from a MIDI input, and each MIDI octave can be repeated (using the Octave Repeat button) so the Temper resonators will respond correctly, regardless of the octave in which you're playing. If you like the sound of something less 'correct', the Pro version also has a frequency offset

control that affects all the resonator pitches, and this has a range of $\pm 500\text{Hz}$ — you can use this to make things very angsty, in an almost ring-modulator kind of way, if you want to.

Whereas the Lite version has no option for making separate adjustments to the early and late reflections, Temperance Pro allows the user to balance the early and late reflections, and gives you the option of applying the Tempering process only to the early reflections, just the late reflections, or to both. The Density control adjusts the number of resonant modes, allowing for less natural-sounding effects when lowering the density, and the filter slopes can be switched between 12 and 36 dB/octave. Note Width allows the note resonances to be narrowed, much in the same way as increasing the Q of a filter.

Sound Judgement

So Temperance works differently from other reverbs, and there's an incredible amount of control if you want it. But what does this actually sound like? The straightforward reverbs exhibit a smooth,

natural character, so you shouldn't write this off as a plug-in that's only to be used for new effects. In fact, adding just a hint of Temperance set to the key of your song can really enhance the reverb in a very subtle and natural-sounding way. But when you push it further, the reverb tail rings out with a really distinctive chordal structure, and only the selected notes really pop out. The effect becomes even more evident if you utilise the sequencer to step through different chord note combinations. Narrow the Note Width to tighten up the resonators, and the reverb can take on an almost steel drum or wind chime sort of character. Most of the effects are nicely subtle, and seem to work in a very musical way, but with the Pro version there's lots more scope for pushing the boundaries if that's what you prefer. Temperance is a great example of character meeting creativity, and I'm always happy to see a product that breaks new ground like this. **////**

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How I Got THAT SOUND

Jeff Balding • Shania Twain 'Man! I Feel Like A Woman'

JOE MATERA

Nashville-based producer and engineer Jeff Balding has worked with huge artists such as the Eagles, Faith Hill, Morgan Wallen, Blake Shelton, Taylor Swift, Shania Twain and Megadeth, to name but a few. Asked to nominate a favourite sound from his vast back catalogue, he zeroes in on the drum recording for Shania Twain's 1997 hit 'Man! I Feel Like A Woman'.

The Luxury Of Time

"On most tracking sessions in Nashville, you might spend an hour dialling in drum

sounds. 'Man! I Feel Like A Woman!' got the benefit of three full days of auditioning drums, trying different placements, and fine-tuning sounds for the album. We went through dozens of snares, plus multiple kicks, toms and cymbals until we found the few with exactly the right character for the record.

"When Mutt [Lange, producer] was looking for a studio to track Shania's *Come On Over*, he stopped by the Tracking Room in Nashville to check it out. I happened to be working on a metal album at the time, with the drums set on a riser in the studio's stone room. Mutt mentioned he'd always loved the sound of drums on a riser, so we kept that concept for Shania's tracks. The combination of the riser and the stone room was the perfect recipe for the big, lively tone we were after.

"Once we locked in the drums and cymbals, I auditioned several mic choices for each position, including different room mic placements. Even though I have my go-tos for drum mics, I wanted to make sure every choice worked specifically for the tone we were after on this record —

Hear The Sound

W <https://open.spotify.com/track/6sxptembJVty4sNtcPMAVz>
W www.youtube.com/watch?v=ZJL4UGSbeFg&list=RDZJL4UGSbeFg



mics that complemented the drums, EQ'ed easily, and blended naturally. The end result was a drum sound as much about the room and mic placement as it was about the performances themselves."

Lots Of 67s

"For the kick, we used a clear [Ludwig] Vistalite bass drum for its punchy, fat sound, miked just inside the front head with an Audio-Technica ATM25. I ran it through a 31-band EQ for subtle tweaks to even out the tone. All the drums went through Neve 1073 preamps and a Neve BCM-10 sidecar, using the 1073s for EQ and occasionally bringing a signal into the SSL console for extra shaping. The snare was miked with an ATM23 instead of the usual SM57, and ATM25s went on the toms. Overheads were vintage Neumann U67s. Room mics were also U67s — two in the back corners and two in the front corners — with the front pair compressed through a UREI 1178 and the rear pair through a Fairchild 670. We also set up additional mics to capture alternate room effects.

"The final drum sound was worth the work — clear, punchy, and leaving lots of space for the rest of the production." **///**

Jeff Balding: "Mutt mentioned he'd always loved the sound of drums on a riser, so we kept that concept for Shania's tracks."



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GIOVANNI CORDOVA

If you're mixing in multiple studios, working on location, or dealing with an untreated room, monitoring consistency becomes your biggest challenge. Most engineers reach for headphones as the go-to alternative, but there's another solution that's been hiding in plain sight: the same in-ear monitors that revolutionised live sound. IEMs offer something that neither headphones nor most rooms can deliver — absolute consistency. While they might seem like stage gear, they're solving monitoring problems that working engineers face daily, from mobile recording setups to critical listening in less-than-ideal spaces.

A Shifting Landscape

The way we monitor has changed dramatically over the past decade. Where once you might have worked primarily in a single, well-treated control room, today's reality often involves bouncing between project studios, hotel rooms during touring work, and home setups that range from purpose-built to makeshift. Each environment brings its own acoustic compromises, and adapting your ears to these varying conditions can be exhausting and time-consuming. Even worse, it introduces uncertainty into your decision-making process at exactly the moments when you need confidence most.

My own journey into IEM mixing began almost by accident after seeing a compelling review on a trusted YouTube channel. The relatively low initial investment allowed me to experiment alongside my existing monitoring system, and the results were immediately positive enough to spark deeper research into what turned out to be an entire world of IEM technology. What started as curiosity about an alternative monitoring approach quickly revealed an ecosystem with decades of development behind it, driven primarily by the demanding requirements of live sound engineers and performing musicians.

Technical Advantages

The key difference with IEMs lies in how they couple to your ear. Unlike headphones, which create a variable air gap between driver and ear, IEMs sit directly in your ear canal. This eliminates the complex interactions between driver response and your individual ear geometry, which means you get reliable frequency response regardless of positioning or head movement.



If there was a monitoring system that was reliable, affordable and sounded the same everywhere, would you use it for mixing? Well, there is — and you probably already own it.

This physical coupling also bypasses much of the HRTF (head-related transfer function) that colours headphone reproduction. While this might initially sound like a disadvantage — after all, we're used to hearing the world through our individual HRTF — it actually means the response you're hearing is closer to what the drivers are producing. There's less individual variation to account for. Which is one reason why IEMs translate well between different listeners once you understand their characteristics.

I've found this predictability transformative in practice. The learning curve with IEMs was surprisingly gentle compared to adapting to new rooms or headphones, likely because they're inherently stable. After I'd established familiarity through reference playlist listening, and undergone careful experimentation before making any significant mix changes, the translation quality became reliably repeatable. Unlike headphones, where slight shifts in positioning can alter the sound, or rooms,

where your listening position creates dramatic differences in what you hear, IEMs deliver the same response every time.

Beyond consistent response, IEMs offer another practical advantage that standard headphones struggle to match: sound isolation is superior to most headphones. Quality universal-fit IEMs typically achieve 15-25 dB of passive isolation, while custom-moulded versions can reach 30dB or more. This enhanced acoustic sealing lets you listen critically at lower SPLs, reducing fatigue while revealing detail that external noise would otherwise mask. In practical terms, this means you can work effectively in environments that would be challenging with conventional monitors.

The portability factor is worth mentioning too. After years of transporting full-sized headphones in their cases between sessions, having reference-quality monitoring that fits in a pocket represents a meaningful improvement. The protective cases most IEMs ship with are typically no larger than a glasses case, making them trivial to pack for any session or

trip. This convenience factor shouldn't be underestimated — monitoring you'll actually bring with you is infinitely more useful than monitors left at home.

Practical Limitations

The one thing IEMs can't replicate is that tactile bass response you get from large monitors. Some IEMs offer excellent low-end performance and linear extension, but the physical sensation of low-frequency energy 'in your chest' — which many producers and engineers rely on for judging how kicks and subs will translate to club systems or large speakers — simply isn't there. Full-range speaker verification in a well-treated room remains essential for this reason.

This becomes obvious when working on bass-heavy genres or when making critical judgments about sub-bass extension and impact. While you can hear both the frequency content and the dynamics accurately through quality IEMs, you won't feel it the same way. It's worth thinking about when planning your monitoring approach: IEMs work brilliantly for detailed balance work, automation and general mixing decisions, but a final check on speakers is always helpful.

Communication during in-person sessions becomes more challenging too. Unlike open-back headphones, which maintain some ambient awareness, most IEMs create complete acoustic sealing that affects both client interaction and the ability to follow conversations between collaborators in the control room. This requires conscious adjustment to working practices. Some engineers develop a habit of keeping one IEM out during collaborative sections, though this obviously compromises the monitoring stability. Others establish clear communication protocols, using visual cues or talkback systems to maintain workflow.

However, the enforced lower listening levels actually benefit long-term hearing health. While those accustomed to high-SPL monitoring will need adaptation time, this shift towards quieter working practices helps protect against the industry's widespread hearing damage problems. It's a change worth embracing.

Implementation Considerations

In terms of fit, in-ear monitors are available in two forms. So-called universal models come in generic moulded shells, to which the user fits soft 'tips'. Custom moulds,

by contrast, are based on personalised measurements and shaped to precisely fit the user's ear canals.

Universal-fit models can work well, but depend heavily on proper tip selection. Silicone, foam and hybrid materials each offer different noise rejection and comfort characteristics. Budget additional time and cost for experimentation. Foam tips typically provide the best ambient exclusion and bass response due to their expansive nature, but they compress over time and need regular replacement. Silicone tips are more durable and easier to clean, though they may provide slightly less sound rejection and require more precise sizing. Some manufacturers offer hybrid designs with foam cores and silicone exteriors, attempting to combine the benefits of both approaches.

Finding the right tip size is crucial. Too small and you'll lose both acoustic sealing and bass response; too large and you'll experience discomfort during extended sessions. Most IEMs ship with multiple tip sizes, and it's worth systematically trying each one during extended listening to find the optimal fit. Your ear canals may differ in size, so don't assume you need the same size for both ears.

Custom moulds are where things get interesting. Yes, you'll need to visit an audiologist for ear impressions or 3D scans, but the improvements in comfort and isolation are substantial. It's not cheap, but when you compare to the cost of high-end monitors plus proper acoustic treatment, custom IEMs start looking fairly cost-competitive. The impression process involves having silicone material injected

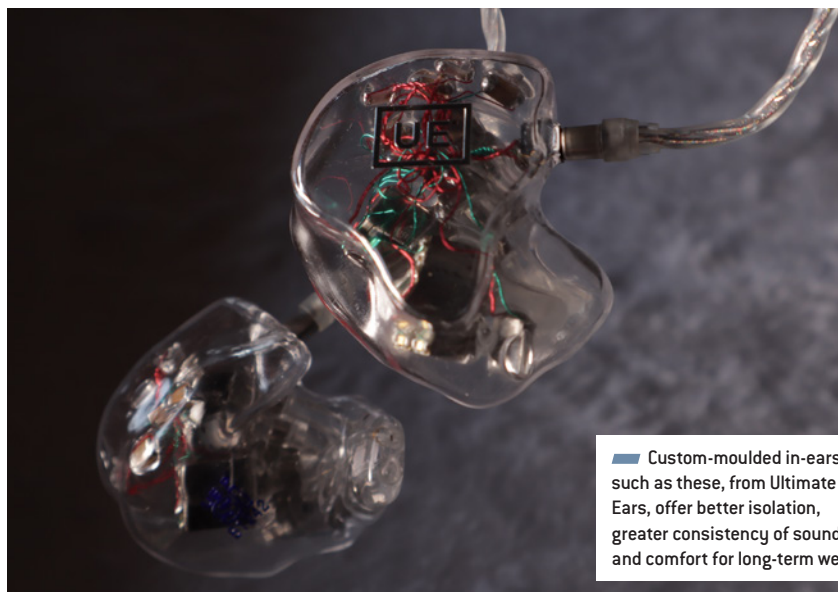
into your ear canal, which sounds more uncomfortable than it actually is. Modern impression materials are soft and warm, and the whole process takes around 20 minutes.

The resulting custom shells fit your ear anatomy precisely, distributing pressure across a larger surface area rather than relying on a few contact points as universal tips do. This makes extended wearing far more comfortable. The improved seal also provides superior ambient exclusion and more dependable bass response, as there's no risk of the seal breaking with jaw movement or slight head turns.

The selection process really benefits from hands-on testing. If you can get to a specialised IEM centre — there are some excellent ones in London — you'll get invaluable access to expert knowledge and the ability to audition everything from sub-£100 models to £3000+ references. These consultations give you insights into the practical trade-offs between different driver technologies and tuning philosophies that you'd never get from specifications alone. A knowledgeable consultant can also help you understand whether your monitoring needs align better with a neutral reference tuning, or something with slight contouring for extended listening comfort.

Don't overlook amplification, either. Many IEMs benefit enormously from dedicated amplification rather than your standard headphone output. A good desktop headphone amp or even a portable DAC/amp combination can dramatically improve both dynamic range and detail resolution — it's often the difference between "these sound decent" and "now I understand what all the fuss is about." The difference stems

»



Custom-moulded in-ears such as these, from Ultimate Ears, offer better isolation, greater consistency of sound, and comfort for long-term wear.

» from several factors: better power delivery for transient response, proper output impedance matching, and often superior DAC implementation.

Personal Experience & Adaptation Strategies

Making the switch to IEM mixing turned out to be easier than I'd expected, and I think it makes a difference if you're already comfortable with earbuds or AirPods. Sure, consumer devices don't come close to the linear frequency response and resolution you'll get from dedicated monitoring IEMs, but the basic concept of listening in-ear isn't foreign to most of us.

In my case, the adaptation period took roughly two weeks before I felt comfortable making significant mix decisions. During this time, I continued using my main monitors as primary references but increasingly verified decisions through the IEMs. This parallel approach let me develop trust in the IEM translation without risking mix decisions during the learning phase.

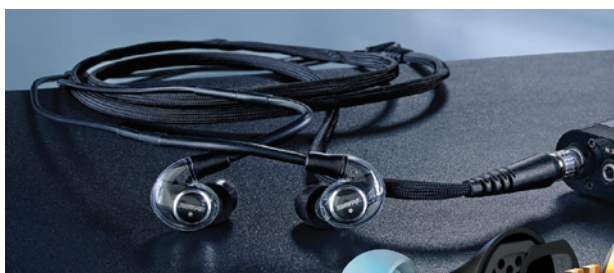
After the initial fortnight, I found myself naturally reaching for the IEMs for detailed work, using speakers primarily for bass verification and final checks.

I've found that developing custom EQ profiles has become essential to my workflow. Understanding psychoacoustics and the Equal Loudness Curve lets you compensate for more than just device frequency response — you can create EQ curves that maintain a perceptually flat response at different SPLs. This means you'll make accurate decisions regardless of your monitoring level, which is particularly valuable if you want to avoid ear fatigue.

The key thing to remember about the Equal Loudness Curve is how our perception of bass frequencies diminishes at reduced volumes. Once you're conscious of this, you can compensate when monitoring at safer levels. If you want to go more in depth on the matter, I recommend having a look at ISO standards 226 and 532.

The high level of isolation you get with IEMs actually became one of their biggest advantages for me. The ability to focus entirely on the mix without external distractions gives you a level of detail awareness that's difficult to achieve in most environments. This benefit extends beyond just noise rejection — there's something psychologically different about the complete immersion IEMs provide. You're not sharing the sonic space with room reflections or external sounds

Understanding Driver Technologies



Alternative driver technologies such as Shure's electrostatic IEMs and Audeze's planar magnetic models are beginning to make inroads into the market.

Several different driver technologies are used in in-ear monitors. Understanding the differences can help you choose what'll work for your situation.

- Balanced-armature drivers excel at midrange clarity and high-frequency extension, offering remarkable detail in a tiny package. They're the technology that powered early hearing aids, and remain dominant in the IEM space due to their efficiency and precision. That said, they typically struggle to deliver great low-frequency performance without precise tuning.
- Dynamic drivers bring a more conventional speaker-like response, with natural bottom-end extension and a sense of ease that many engineers find immediately familiar. Modern micro dynamic drivers have improved dramatically, offering surprisingly linear response down to 20Hz in ear-canal-optimised designs. They tend to deliver better slam and impact in the lower octaves, though they're physically larger than balanced armatures and less efficient overall.
- Hybrid designs combine multiple technologies, typically using dynamic drivers for bass response and balanced armatures for mids and highs. This approach aims to

capture the best of both worlds, though the crossover implementation becomes critical. A well-executed hybrid can deliver both visceral low-end impact and crystalline high-frequency detail, but a poorly integrated one can sound disjointed, with obvious transitions between driver regions.

I'm currently using a hybrid 2DD 4BA configuration, which has proven ideal for my mixing and mastering work. You get the punch and extension of dynamic drivers in the low end combined with the detail and precision of dedicated balanced armatures in the mids and highs. When you pair this with a quality DAC/amplifier combination, you get both the tactile engagement and analytical precision that our work demands.

More exotic technologies like planar magnetic and electrostatic drivers are appearing in flagship models, offering potentially superior transient response and even finer detail retrieval. These typically come at significant cost and complexity, requiring even more thoughtful amplification matching.

the way you are with speakers. It's just you and the mix, which can accelerate decision-making and help you focus on what actually matters in the production.

Final Assessment

IEMs won't necessarily replace your main monitors, but they might complement them brilliantly. They excel when you need reliable performance across multiple environments — which is particularly valuable if you're working between different studios or on location. The trade-offs are clear: you gain remarkable consistency, superior sound rejection, and unmatched portability, but you sacrifice that tactile speaker feel and easy collaborative listening. Whether this exchange works for your workflow depends entirely on how and where you work.

If you're managing multiple venues and dealing with variable acoustic environments, having monitoring that delivers identical performance anywhere has real value. The compact form factor simply makes this uniformity practical to implement. I've found the learning curve manageable, the translation quality dependable, and the workflow benefits extend far beyond simple portability.

The technology keeps evolving rapidly, with driver innovations and tunings reaching new levels. As more engineers discover these benefits, IEMs are likely to become as common in studio workflows as they already are in live performance contexts. It's an interesting time to get involved. ■■■



True Diversity
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True Wideband
True DPA Clarity

N-Series

Digital Wireless System

N-Series is a fully digital, true diversity, wideband system that features DPA-quality sound.

- Renowned DPA clarity – known from our mics
- True Radio Diversity – strongest, most reliable stability
- Fully digital – all the way through, no quality loss points
- True Wideband – 470 to 870 MHz in same system
- True dynamic range of 126 dB



RORY DOW

Australia's Innerclock Systems have been around for 20 years, but you may never have heard of them. They specialise in a niche corner of the music technology market: MIDI timing and synchronisation. They obsess over jitter, port alignment and sample accuracy, and create tools that work to eliminate the timing problems inherent in music production.

Until now, they have mainly focused on MIDI clock generators. The GridLock II promises a lot more: it is their new flagship MIDI interface. In fact, Innerclock Systems are billing it as the first ever "professional" MIDI interface. Why?

Well, there are certainly a lot of ports, both USB and 5-pin. It supports 5-pin MIDI, USB MIDI, Din-Sync and CV/Gate. There are also some very impressive specs on timing, which promise event, clock and port-to-port accuracy to within a few microseconds. The customisable firmware allows you to create virtual OS system ports for each instrument in your studio. Then there's the clock generation, which also promises frightening accuracy, and, crucially, the ability to shift clocks independently across all available ports so that all your externally clocked gear is perfectly in sync. In short, it might be the most over-engineered MIDI interface ever made.

Hardware

Let's look at the physicals first. The GridLock II is a desktop USB 3 MIDI interface with ports on the front and back. It can also be rackmounted using a custom 19-inch

Innerclock Systems GridLock II MIDI Interface

Can a MIDI interface really make a difference? Innerclock Systems think so.

rackmounting kit. It's a fairly chunky unit measuring 190 x 235 x 70mm. On the rear are eight 5-pin DIN ports. These can carry any MIDI or Din-Sync data, switchable per port. There is also a pair of DB25 connectors for the optional VTX-1 Eurorack CV breakout box, which provides 16 CV outputs. Innerclock tell me that additional breakout box formats are coming.

Around the front, you'll find eight more 5-pin DIN ports. Four are MIDI outputs, two are MIDI inputs, and two are dedicated Din-Sync outputs. You'll also find two USB-A 2.0 ports and two USB-B 2.0 ports. The two USB-A ports will support powered hubs, allowing you to connect up to four USB devices each. The USB-B ports are host ports, enabling you to connect to another computer or hardware device that supports host connections, such as an iPad or a modern Akai MPC. Each host connection will provide four new virtual ports on the host. There is also a dedicated headphone output that provides an

accurate click track. Perfect for drummers or anyone needing a reference clock.

Between the physical and USB MIDI ports, there are a potential 28 MIDI outputs (eight of which are switchable to Din-Sync), 10 MIDI inputs, and two dedicated Din-Sync outputs. Plus 16 CV outputs. That should be enough to drive any small to medium-sized studio, but if you need more, you can connect up to four units, for a total of 112 MIDI outputs, 40 MIDI inputs and 64 CV outputs. Innerclock claim the port-to-port alignment across the entire system is accurate to a few microseconds, which is seriously impressive.

To connect the GridLock II to your computer, you need a USB 3 connection and a pair of spare audio outputs on your audio interface. The audio outputs provide a sample-accurate analogue clock via the included GridLock II plug-in (available in VST3, AU and AAX formats). The GridLock II converts this audio clock to latency-free MIDI clock, Din-Sync and voltage clock

signals. You assign a track containing the plug-in to your spare audio outputs, and GridLock II will take care of the rest.

The point of having GridLock II generate the clock, instead of the DAW, is that it eliminates DAW-specific timing and latency issues. Also, every port can have different clock divisions, swing and time shifting, which means you can compensate for any instrument's inherent lag by shifting the clock so that each one is perfectly aligned to the grid. It's a very impressive system. There is nothing quite like having multiple drum machines and sequencers running in perfect lock-step.

Software

The main configuration of GridLock II is done through the GridLock II Control application. Sorry, Windows users, but for now, it's Mac-only. Installation is quick and easy, and if you have your GridLock II connected, the Control app allows you to customise all manner of things by completely rewriting the GridLock II's internal firmware.

Let me give you an example. Let's say you have a Yamaha DX7 that you use as a master keyboard, but you also want to sequence it with your DAW. In this case, you would connect the MIDI output of the DX7 to a MIDI RX port on the front of the GridLock II, and the MIDI input of the DX7 to one of the MIDI TX ports on the rear. In the GridLock II Control application, you create a new instrument with a custom name, 'Yamaha DX7', and assign those two ports to the instrument. Then, you configure the ports, specifying whether they send MIDI clock, or filter MIDI data such as aftertouch or SysEx.

Once you save this new configuration, the GridLock II firmware is rewritten, and suddenly your DAW will detect a new

MIDI input and output port, beautifully named 'Yamaha DX7'. A specific port for every instrument!

Let's now add a MIDI sequencer, such as an Elektron Digitakt. The Elektron boxes have USB MIDI, so we'll use the USB-A connector on the GridLock II (you can use a four-port hub here to connect up to four MIDI devices to the same USB port). In the GridLock II Control application, we create another instrument, name it 'Elektron Digitakt', assign it to the first USB-A port, enable MIDI clock (because we want the sequencer to play with the DAW), and save the session again.

Now, your DAW will display two additional MIDI ports with the custom 'Elektron Digitakt' name. When you use the GridLock plug-in in your DAW, the Digitakt will lock perfectly in time. If, for any reason, the sync isn't perfectly in time, you can use the GridLock II's clock shift function to move the clock relative to the DAW grid, just for that instrument. You can shift in 10-microsecond increments, for total accuracy, and in divisions of the DAW grid (16ths) for more rhythmically creative offsets. While you're there, why not experiment with the swing and clock rate PPQ divisions, which can dramatically change the feel and speed of the Digitakt clocking.

Let's add a CV-based instrument, such as a Roland SH-101 or a Eurorack modular. With the VTX-1 Eurorack breakout module connected to the DB25 connector on the GridLock II, you can configure each of the 16 mini-jack outputs as Gate, CV, Trigger, Clock or Transport. As before, we create a 'Roland SH-101' instrument in the Control application and add some Voltage ports: one each for gate, pitch and clock (to have the SH-101 sequencer play along when needed). Gates can be V-Trig or S-Trig. Amplitude can be 5, 10, 12 or 15 Volts (try getting

Innerclock Systems GridLock II

\$2500

PROS

- By far the best MIDI timing of any MIDI interface, ever.
- A generous number of ports, and it's expandable.
- Built-in MIDI to CV conversion.
- Support for USB MIDI devices and hosts.
- Custom systemwide MIDI ports per instrument is a revelation.
- Perfect for medium to large studios.

CONS

- It's Mac-only.

SUMMARY

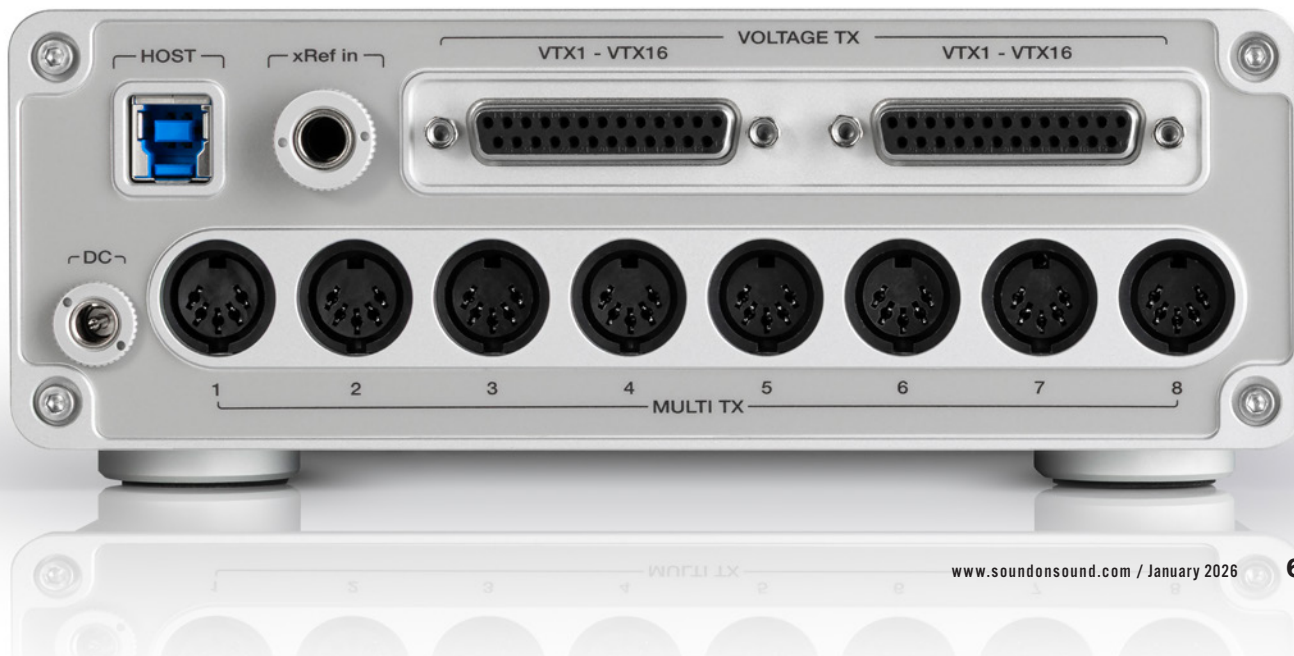
The GridLock II is a new flagship MIDI Interface from the timing-obsessed Innclock Systems. It has a ton of MIDI ports (USB and 5-pin DIN), and also supports Din-Sync and CV/Gate/Voltage Clocks (via the VTX-1 breakout). The real innovation here is the frankly astonishing timing. MIDI clock, MIDI notes, CV, Din-Sync and crucially, port-to-port alignment are all within microseconds. That near-perfect timing is independent of which DAW you use. And, the ability to rewrite the firmware to create per-instrument MIDI ports is superb. A new standard has been set.

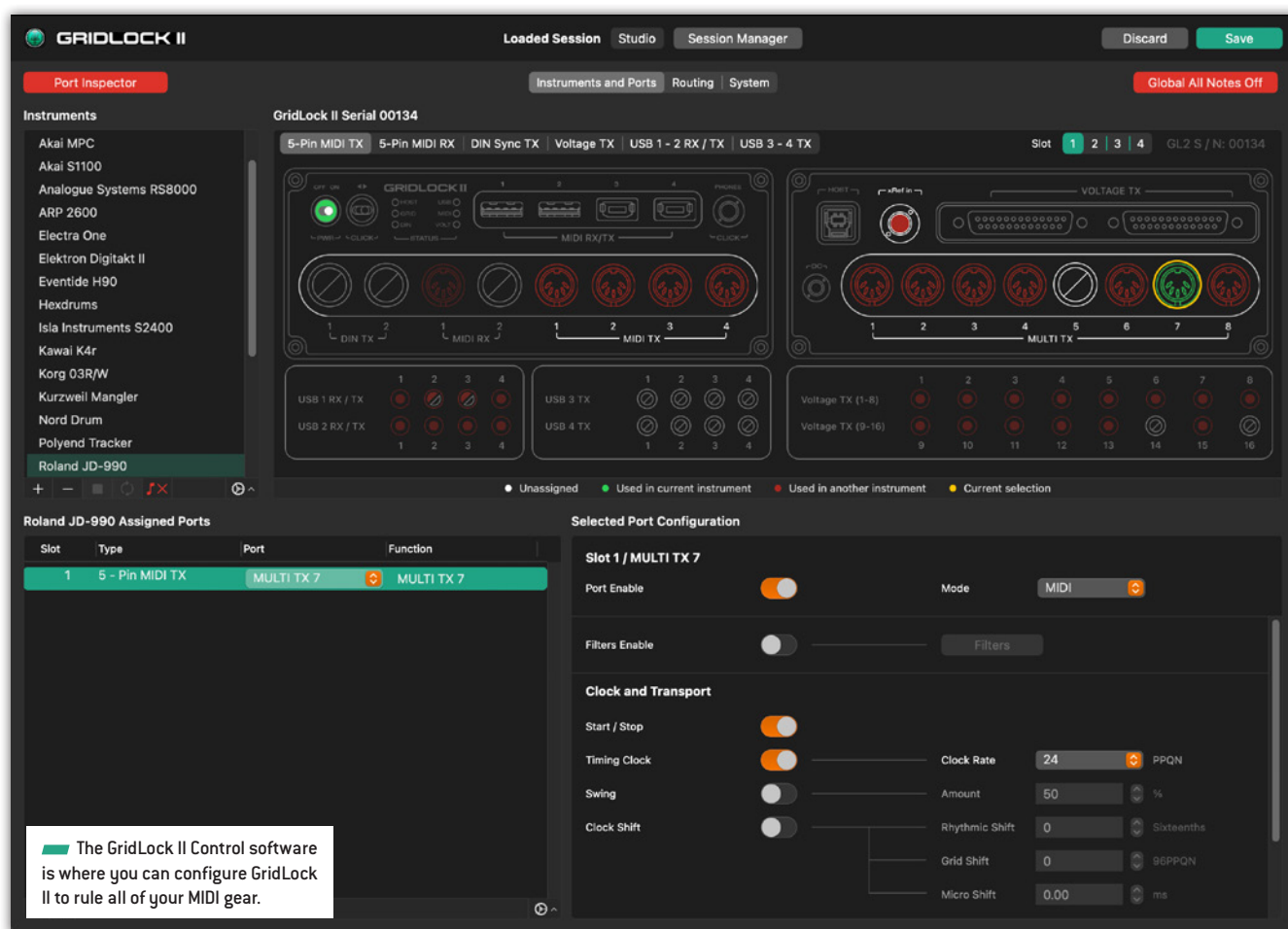
a 15V signal from a typical DC-coupled audio interface!). CV signals can be V/Oct or Hz/V, with custom pitch scaling, pitch-bend support, velocity support or MIDI CC binding. Analogue clocks are fully configurable too.

Once the configuration is saved again, the SH-101 appears as another MIDI port in the DAW. The GridLock II hardware is doing all the heavy lifting of converting the MIDI to CV signals needed for the SH-101. This handling of MIDI and CV instruments together, all with the super-precision and

»

On the GridLock II's back panel we find a USB-B port, another eight 5-pin DIN sockets, and two DB25 connectors for connecting optional breakout boxes.





» zero-latency timing, is a minor miracle. And it's all DAW-agnostic. The ports are systemwide, so every piece of MIDI software benefits.

One last example. Let's say you have another computer, or a mobile device such as an iPad. As long as the device supports class-compliant MIDI, you can connect it to the USB-B ports on the GridLock II and create another instrument. Now you can sequence your iPad, or second laptop, from the main DAW just like the rest of the studio, even clocking DAWs to each other.

Once you have these instruments fully configured, each with its own dedicated MIDI port(s) in your DAW, you can merrily use the DX7 to play any of the other instruments, whilst anything that is set to output MIDI clock (or Din-Sync / Voltage Clocks) will sync effortlessly every time you start and stop the DAW transport.

The GridLock II has a built-in routing page and a virtual patchbay for all instruments. Create a routing path from the Digitakt to the DX7 and the SH-101, save the firmware again, and you can happily sequence those instruments from the Digitakt without needing a DAW. And because all these configurations (including the routing) are saved to the GridLock II firmware, they are persistent through reboots. That means you can take the GridLock II onto the stage and use it as a central MIDI hub for a live show.

The GridLock II's ability to rewrite its firmware while performing all the complex calculations within the unit is seriously impressive. It means the computer is relieved of almost all duties, and you can rely upon the kind of MIDI timing that, until now, has not been seen before. Having each instrument on its own well-named port

means less head-scratching with routing and MIDI channels. It feels like someone has made you a custom MIDI interface for your studio. And as soon as you buy some new gear, it's easily added with its own virtual MIDI port in your DAW (even the CV-based instruments). It's refreshing to spend more time making music and less dealing with the vagaries and whims of MIDI.

Conclusion

There's a great deal of information, and misinformation, around timing, 'groove' and MIDI capabilities. People will tell you that under 10ms of lag or jitter doesn't matter because the human ear can't detect it, or that it's natural 'humanisation'. Poppcock.

Musicians (especially drummers) and studio engineers will tell you different. Sub-millisecond timing can make a huge difference in 'feel'. Imagine a kick drum and a tom, hitting at the same time. A few milliseconds in the trigger timing can mean the difference between two signals being in and out of phase, which can affect the mixed audio by removing critical bass frequencies and smearing transients. Now

"I cannot overstate the difference that good MIDI timing makes. Having several drum machines and synthesizers all locked perfectly into the groove is a revelation."

compound that problem across a whole drum kit, or even an entire studio.

The same is true with clocks and synchronisation. The more accurate the clock, the more reliable and predictable the results will be, with the ultimate prize being that everything sounds 'in the pocket'. It is informative to understand how instruments can relate to each other in the time domain. To use MIDI to replicate how good players create 'feel' by naturally adjusting their timing, you need absolute control over timing to begin with.

Many will argue that MIDI is a serial protocol (meaning it cannot send two pieces of information simultaneously; they must be sequential), so bad timing is inherent in the system. Innerclock have taken every possible step to mitigate the timing issues that may arise as a result of this. Their custom Mac OS driver and a dedicated scheduler release events exactly on time using the GridLock's own master clock. The result is sub-10-microsecond jitter and skew across all 44 ports, even under heavy load — a feat that no other MIDI interface can achieve.

I am seriously impressed by GridLock II. You don't need to measure the timing to feel the results. It feels solid in the way that a good audio interface does. In fact, Innerclock are so confident in their engineering that they publish detailed specifications on their website, just as audio interface or mixer manufacturers might publish specifications for harmonic distortion, signal-to-noise ratio or dynamic range.

The GridLock II is perfect for medium to large-sized studios. And also live rigs. It feels faster, more reliable, and more flexible than anything I've used before.

There is the occasional niggle, and a few signs that this is still a product in its formative stages. For example, in the Routing page of the Control application, you cannot filter by MIDI channel, which means that if you route a MIDI sequencer to a CV instrument, that CV instrument will respond to all MIDI channels. Innerclock tell me this should be easy enough to fix, so hopefully it will be sorted by the time you read this. The GridLock II Control application isn't the prettiest thing in the world either, but that's purely cosmetic. Also, using the MIDI clock

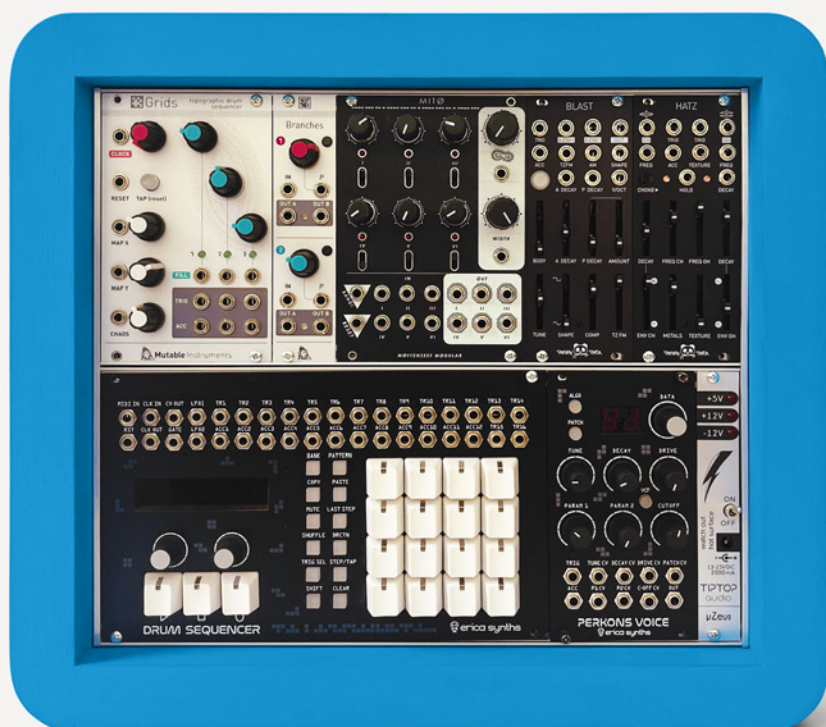
time shift function can be time-consuming because every time you adjust, you must save the firmware to hear the difference. It's a shame that the shift functionality doesn't work in real time, which would make it really easy to shift things by ear.

These are minor gripes, however. The GridLock II's core functionality is seriously impressive. I cannot overstate the difference that good MIDI timing makes. Having several drum machines and synthesizers all locked perfectly into the groove is a revelation. You don't necessarily hear it, but you feel it.

GridLock II is the most innovative MIDI interface on the market. At \$2499, the price is significantly higher than any of the competition, but it's not unreasonable. You get a ton of ports, near-perfect timing, MIDI-to-CV, Din-Sync and genius DAW-agnostic instrument-based configuration. The GridLock II is a big step in the evolution of the MIDI interface, and it might just be the last MIDI interface you ever buy. **///**

£ \$2500, VTX-1 Euro Analogue Expander \$240. Prices include VAT.

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— The reversible Type-C connector (above) is newer than the original Type-A and B connectors (left) — yet most interfaces that use it are actually USB 2.0 devices.

The ABC Of USB

Is USB 2 Good Enough?

Why are almost all audio interfaces still USB 2 devices? And does it matter?

SAM INGLIS

Audio interfaces mostly connect to computers over Universal Serial Bus, or USB. Back in the day, computers used the rectangular Type-A USB socket, while peripherals used the house-shaped Type-B socket. Today, both computers and interfaces use the smaller Type-C socket.

Depending on how recent your computer is, you might find your Type C sockets labelled USB 3.1, USB 3.2, USB 4 or Thunderbolt. But delve into the specifications of your audio interface, and the chances are you'll find it's actually a USB 2.0 device. How can this work? And why are interface manufacturers apparently stuck in the past?

What's The Difference Between Type-C And USB 3?

The lettered connector type and the numbered USB standard refer

to related but different things. The connector type determines how electrical links are made between the host and destination devices. The USB version determines how those paths can be used to transfer data, and hence how audio and MIDI flow from interface to computer and back again. It's even possible to use Type-A, B or C connectors purely for charging, without supporting any form of data transfer.

One of the central pillars of the USB standard has always been backwards compatibility. The idea is that older USB devices should never be left behind as computers move to newer versions. When you plug a USB 2 device into a computer's USB 3 port, it'll be recognised for what it is, and the computer won't attempt to do anything that isn't supported under USB 2. What type of physical connector it has makes no difference.

In this sense, then, the fact that most audio interfaces are USB 2.0 devices isn't a problem. There is no risk of them becoming orphaned by newer connectors or computers. But even so, it's reasonable to ask why that's the case. Since the rest of the world has moved on to newer versions of the standard, why haven't interface manufacturers followed suit? And wouldn't Thunderbolt be even better?

Is USB 2 A Bottleneck?

Interface manufacturers will tell you the reason they haven't moved to USB 3 or later is because there's simply no need. The USB 2 standard is more than adequate for audio interfaces, and thanks to USB's backwards compatibility, a USB 2 interface can be connected to any modern computer. Because USB 2 ensures maximum compatibility without compromising performance, there is no reason to use anything else.

There's a lot of truth in this answer, but it isn't quite the full story.

It's certainly true that USB 2 offers plenty of bandwidth, at least on paper. If that bandwidth could be fully exploited by an audio device, it would be able to handle more than 400 streams of mono 24-bit audio at 48kHz. In practice, though, this isn't possible, because the ability to control the data flow is exceeded long before the data flow itself becomes overloaded. In terms of channel count, one of the highest-performing USB 2 interfaces on the market at the time of writing is RME's MADiface USB, which provides 64 inputs and 64 outputs at 48kHz. That's about at the limits of what

So, in that sense, USB 2 is certainly ‘good enough’ — except that if you look around, you’ll find that very few other interfaces get anywhere near this. The large majority of USB 2 interfaces offer only 24 ins and outs, or less. Why?

One reason is that most audio interfaces are constrained by other factors not directly related to the USB 2 specification.

Suppose you are in the market for a vintage sports car from the early 2000s. You might test drive a Lotus Elise and a Vauxhall VX220 and find

At the heart of an interface is a hardware processor that has a lot of complex jobs to do. It handles communication with the computer over USB. It takes audio from all the various inputs, manipulates it in various ways and routes it to different outputs, whilst keeping everything in perfect sync.

“Many of the limitations of current interfaces are actually down to the implementation rather than to the USB 2 standard itself.”

Nearly all manufacturers take the second route, and as a consequence,

However, this widely used platform is designed to be affordable and reliable, not to exploit the USB 2 standard to its very limits. So, if you're wondering why there are hardly any 32-channel USB interfaces on the market, or why the internal mixer on your interface loses some features at high sample rates, that's not because USB 2 can't support it in principle. It's because

Latency is a word you'll hear a lot in discussions of audio interfaces. In this context, it refers to a delay between the signal entering the interface and being heard again at its output. This can make it hard for singers and musicians to properly hear themselves whilst performing, so one of the most important features of any professional audio interface is the ability to keep the delay as short as possible.

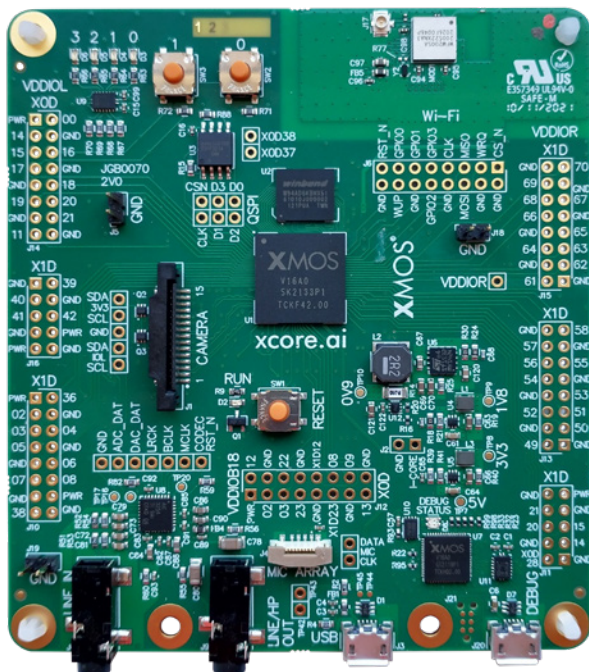
an analogue-to-digital converter. The processor described above then pipes it down the USB cable to the computer, where our DAW software takes care of writing it to a drive, before spitting it back down the USB cable. Finally, the hardware processor in the interface routes it to our headphones or monitors.

Does the choice of USB 2 make it hard to achieve low latency? Not necessarily. How low a buffer size you can effectively work with depends on several factors, but one of the most important is the audio interface driver. This is a piece of low-level code that runs on your computer and handles the transfer of data over USB.

It's not only the connection protocol that determines how good or bad the low-latency performance is, it also depends on how efficient the driver code is. And interfaces that are built around a generic processing platform often rely on generic driver code. On Apple machines, most manufacturers opt »



Thanks to its use of custom hardware and drivers, RME's MADiface USB can provide 64 inputs and outputs at base sample rates over USB 2.



■ The large majority of USB audio interfaces are based around Xmos's xcore chipsets.

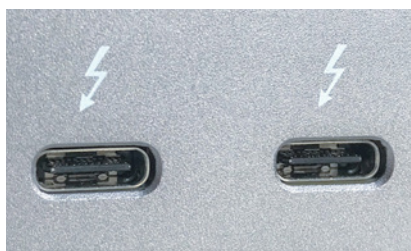
although larger rackmount models still usually need a dedicated mains supply.

Is Thunderbolt Better?

You'll sometimes hear that Thunderbolt is superior to USB for connecting audio interfaces. Both Thunderbolt and later variants of USB offer far, far higher data bandwidth than could ever be required for audio, and eliminate the USB 2 control restrictions that might prevent this

bandwidth from being exploited fully by an audio interface. When it was first launched, Thunderbolt also delivered more electrical power than USB, but since both now use the Type-C port, this is no longer an issue. The remaining important difference is that Thunderbolt integrates into the computer architecture at a lower level than USB. Technically, in fact, Thunderbolt is an 'externalised' version of the PCI Express protocol.

Put very simply, what this means is that there are fewer hoops to jump through to transfer data over Thunderbolt. Unlike USB 2, there hasn't been a generic platform and driver code available off the shelf, but transferring data without delay or interruption is intrinsically more straightforward. Hence Thunderbolt interfaces usually perform better than USB 2 interfaces based around generic processors and drivers, though they can be matched by a well-written custom USB driver.



■ Thunderbolt now uses the same Type-C connector as recent USB variants. It's generally more expensive to implement, but does offer some performance advantages.

However, there are also down sides. First of all, although all modern Apple computers support Thunderbolt, the same has not always been true of Windows computers (though this looks likely to change with the latest generation of USB 4 chips, which support Thunderbolt 3). And second, developing and manufacturing a Thunderbolt interface is more expensive than developing a USB 2 interface, in part because the hardware components are more costly and in part because there are no generic platforms available.

It should also be pointed out that crucial aspects of audio interface performance such as dynamic range and frequency response are not dictated by the choice of platform. It is up to the manufacturer to choose converters and design the best possible circuitry around them. Two USB interfaces based on the same processing platform won't necessarily have identical audio specifications, and nor will a Thunderbolt interface necessarily outperform a USB interface in this respect.

Is 'Good Enough' Good Enough?

Ultimately, then, USB 2 does have intrinsic limitations, and these can make it unsuitable for professional applications where large numbers of channels need to be handled. But many of the limitations of current interfaces are actually down to the implementation rather than to the USB 2 standard itself. Manufacturers who choose not to use an off-the-shelf solution, and instead design the entire system from end to end, can achieve higher channel counts and lower latency. They are also able to realise more diverse or unorthodox designs.

However, the existence of a widely used generic platform has advantages too. Most of all, it has brought down the cost of multi-channel audio interfaces. Interfaces based on this platform more than meet the needs of most users, and they do so at a highly competitive price for the features on offer. For most of us, what this platform offers under USB 2 is indeed good enough. If you're one of the minority who needs something more or different, you have the option of buying an interface based on a different platform, whether that be Thunderbolt or a custom USB controller — but be prepared to pay significantly more. ■■■

» to run the interface in Class Compliant mode, which means they can use Apple's own Core Audio drivers. On Windows computers, the generic ASIO driver is supplied by a third-party provider to work with the processing platform in the interface. Both work reasonably well, but manufacturers who go to the trouble of writing their own custom drivers usually achieve better low-latency performance.

Does Type-C Make Any Difference?

As we explained earlier, the choice of connector doesn't determine what USB standard is used. Thus it's perfectly possible for an audio interface to be a USB 2.0 device, but have a Type-C port for connecting to the host computer. And that's exactly the way it works with most current interfaces.

This switch to Type-C ports brings with it some concrete advantages. One is that the connector and cable are both reversible, so you need never again fumble around trying to get it the right way round. A more substantial plus is that the Type-C standard can deliver much more power from the computer to an attached device. Twenty years ago, bus-powered USB interfaces were very limited as to channel count, processing power and headphone volume, simply because they couldn't draw enough current from the computer. Today's bus-powered interfaces are a lot more capable and better-sounding,

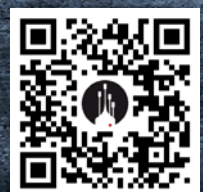
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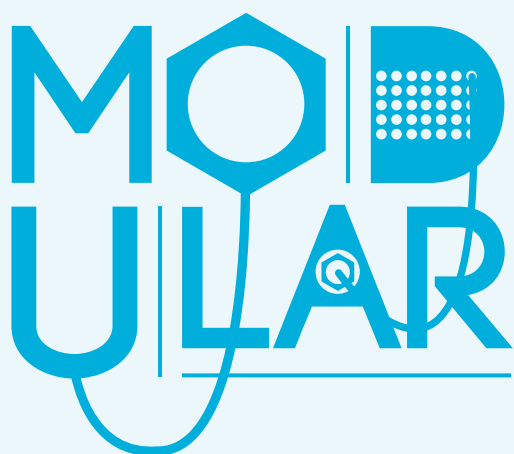
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SOS EDITORS' PICK
GEAR OF THE YEAR 2025

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4MS MetaModule

Eurorack Module

The literature accompanying Expert Sleepers' Disting NT multifunctional module includes a section titled "What is it not?", which reads, "The Disting NT is not a 'virtual modular'. There are no virtual cables flying around." It continues: "It is not a 'patch player'. It is not the intention that you do all the complex setup on a computer and then copy it over to the module."

What could Expert Sleepers possibly be referring to here, I wonder? Surely not the MetaModule from Portland's 4MS, with its computer-programmable virtual modular environment and potential scores of virtual patch cables? Needless to say, as a dotting user of the Disting NT, I was intrigued to find out. The MetaModule does indeed present a virtual modular environment that can accept patches ported over from the free VCV Rack, and which can also be used to create virtual patches internally. It has a central colour screen accompanied by a single encoder and button, with which virtual modules ('plug-ins') are selected from an internal library and compiled into patches. These



can then interface with the rest of a physical system via a set of 12 assignable knobs, as well as eight assignable inputs and outputs. There's a microSD card slot to accommodate memory, a USB-C port and — at least on the MetaModule proper — that's about it.

As for the library of modules on offer, suffice to say it's extensive: 1500 emulations are currently available. Whew! Mutable Instruments designs are available as clones through Cloned Instruments, as is a wide selection of 4MS modules, including classics like the VCA Matrix and Ensemble Oscillator. There are offerings from Befaco and Moffenzeef (whose founder Ross Fish, I notice, is now working with 4MS, excitingly), and many, many more. It's not a stretch to say I'd be getting my money's worth with 'just' a module that could be more or less any Mutable Instruments module I pleased, let alone the rest. The option to incorporate MIDI via a virtual MIDI-CV interface module also expands many familiar favourites' functionality considerably.

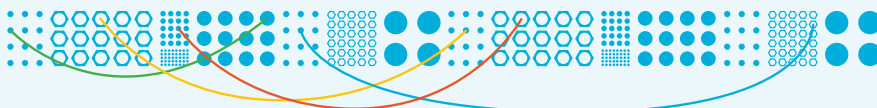
Virtual patches can't expand forever, since the CPU can only handle so many modules (depending on their complexity). As such it will not co-operate with more than a modest handful if you're operating at the more CPU-intensive end of the spectrum. I was not disappointed by this: things can quickly get visually complicated on that little screen, and it can become laborious tracing reams of virtual patch cables and navigating through layers of different 'knob Sets' (since the virtual controls are likely to outnumber the physical ones, it's possible to create more than one mapping per patch). In short, the more there is going on, the less playable the MetaModule becomes, and the less creative it feels. Even a simple patch consisting of a Mutable Instruments Rings clone, a Music Thing Modular Turing Machine

emulation from Stellare Modular and a 4MS Listen Four mixer, for example, took time to construct and configure in the first instance, but once patched and mapped I was away, tweaking in real time and patching it into the rest of my system.

Indeed, I found much of the beauty of the MetaModule to lie in the simple — but expandable — things. Using it solely to emulate a simple mixer, for example, is very useful. But using it to emulate a mixer with a different insert effect on each channel is fantastic. It can be a 909-style drum array ready to pair with an external sequencer, but if desired, that array can also take care of its own bus compression or effects. When it comes to my own workflow, these sorts of use cases are where the MetaModule works at its best, but that's just me: happily I can say it very much retains all the open-ended adaptability we would hope for in modular.

Beyond the MetaModule itself are some expansion options: there's a Wi-Fi expander for wireless patch transfer from VCV, a button expander with eight extra buttons, and an I/O expander with six extra inputs and eight extra outputs. So not only is the MetaModule customisable virtually, but different combinations of hardware can also adapt to your workflow or particular patch needs. You can, for instance, chain MetaButtons units to give a whole grid of buttons if that's your bag. The full set at the time of writing will set you back over \$1000, which is no snip, but needless to say, their combined functionality might span a clutch of modules worth three times that amount.

It's fair to say there are some areas that feel ripe for expansion and improvement here: the size and resolution of the screen struggles at times with more detailed module panels, and in the absence of a computer, the process of patching several



jacks together can be very slow going with just one encoder and button. In practice, the latency between spotting a specific parameter to edit and actually editing it feels longer than is comfortable — at least, in the ordinarily tactile and immediate world of Eurorack. When it comes to VCV, not all VCV modules work on the MetaModule, so be ready to save the Module Finder page on the 4MS website to your ‘favourites’ (VCV itself — very much an unofficial partner of the MetaModule — has no dedicated directory of compatible modules).

This said, you’re unlikely to be left frustrated with the compendium on offer, and it would be prudish not to assume that this catalogue is only going to expand, as is that of hardware expansion offerings, from what I gather. It’s in light of thoughts like this that the MetaModule, while hugely functional, has the air of a relatively early step down a rich avenue of potential. I wouldn’t be hugely surprised — certainly not disappointed — if other iterations of the platform were to emerge (one wonders if the Wi-Fi expander doesn’t open up the potential

to pair with an iPad app), or if 4MS came to release their own software modular environment specifically for pairing with the MetaModule.

In any case, the key question is: can I imagine the MetaModule living in my system for a long time to come? The answer is yes. It’s not about to replace a raft of modules with virtual equivalents, but I don’t think that’s its intention. Atop that, it’s fast becoming the best of friends with my Disting NT. *William Stokes*

£ £599

W www.4mscompany.com

Tobinski Sequencer

Eurorack Module

Sequencing in Eurorack has always felt like a bit of a problem. Perhaps it’s the standard of sequencing we expect these days versus the amount of physical space available in Eurorack, but it feels like sequencing is a question a lot of people are trying to find the answer to, and one which a lot of manufacturers are possibly making more complicated in their ever-more inventive attempts to solve the problem.

Traditionally, modular sequencing means a step sequencer into which you

program a pattern and then sit back as it loops endlessly. Nothing wrong with that — entire sub-genres of electronic music have been built around it — but if you want to move away from that repeating pattern, a whole lot of options suddenly open up, and anyone looking to sequence a modular in 2026 is going to have to do a lot of thinking about how they want to do it.

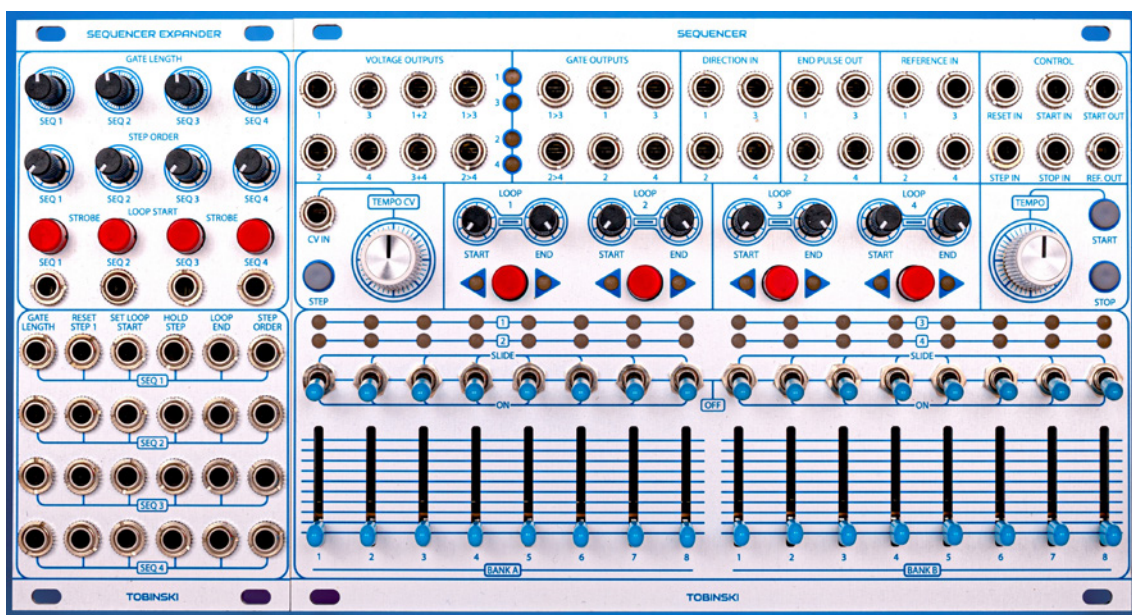
One approach, as demonstrated by the Tobinski Sequencer, is to go back to basic principles — analogue, unquantised, eight steps — and then mess with those principles. Yes, the Sequencer cycles eight-step patterns, but it also gives you all sorts of opportunities to muck around with them and create variation.

At the heart of it are two banks of eight sliders and eight switches. The sliders adjust the pitch, the switches flip between gate on, gate off and slide. Each bank of sliders and switches, though, can be output separately — there are four CV outs and four gate outs. What’s the use of that, if it’s the same gates and pitches being output? Well, the first way to mess with your sequence is with the four sets of loop start and end controls. With sequences one and two patched to different voices, you just have to reduce the length of sequence one to seven

steps and you’re off to the polyrhythmic races.

Using the different outputs at the top of the module, sequences one and three or two and four can be combined to form two 16-step sequences, again sharing the same pitch and gate info but with individual lengths. There are also inputs and buttons to change each sequence’s direction as well as individual clock inputs (labelled Reference In) and End Pulse Outs for each channel. A final section at the top gives you Reset in, Step In (ie. global external clock), Start and Stop Ins, Start Out and Ref Out (ie. clock out). That’s a lot to take in at first, but what it gives you is a lot of flexibility in how you control the Sequencer and how the Sequencer controls what it’s connected to.

So, there are plenty of ways to mess with your eight- (or two-, three-, four...) step sequence here, but Tobinski also make an expander module called, ingeniously, the Sequencer Expander. This connects to the Sequencer via a ribbon cable and gives you per-channel control over Gate Length,



» Step Order (which switches between six preset step patterns), Strobe, Loop Start, Loop End, step Hold and Reset. As you look into the wider Tobinski range, though, you find that this is very much the tip of an interconnected iceberg, designed to work with modules with such fascinating names as the Harmonic Timing Generator and the Interpolating Scanner, all of which have been cleverly conceived to work together as a complete system. It's an ambitious and exciting project and clearly a lot of thought and work has gone into designing it, but it also represents one of the only problems with the Tobinski Sequencer: once you see the rest of the system, you'll very likely want the whole thing! This in turn brings us to the other problem, which is that these modules are not cheap. Cost aside, though, this is an incredibly creative and comprehensive approach to modular, and proof that there's plenty of life in analogue yet.

The Tobinski Sequencer is a very modular way of looking at sequencing. Instead of trying to imitate the sort of functions you'd find in a DAW or a modern groovebox, it takes us back to basic voltages and gates and gives you the tools to get creative with these. Analogue purism means that you may find yourself reaching for a quantiser, and the manual recommends a giving the Sequencer time to warm up before use, but there's still a lot to be said for the back-to-basics approach; it's about working out how to build something with the blocks you've got, rather than simply buying more blocks. And, naturally, this is a menu-free domain. If you want an authentic, hands-on analogue sequencing experience without any danger of boring yourself through repetition, the Tobinski Sequencer is highly recommended. *David Gasper*

£ £719, Sequencer Expander £294.

W www.tobinski-audio.com

Exay One Astrax Eurorack Module

Exay One have entered the Eurorack fray with a remarkable-looking scanner and panner. Astrax is full of futuristic vibes, and projects its sci-fi-inspired design in dramatic front-panel cuts that give you a view into the guts of the machine. The layout is spacious, splitting the module nicely in two, locating the patch sockets tidily at the bottom and offering up some confidently sized controls. It gives a great first impression. That said, the module description, in a rather dull font, sitting in a box on the front, feels really out of place.

On the left, you have Scanning and on the right you have Panning, which doesn't entirely explain what's going on but it's a good place to start. The Scanning side gives you four inputs and a single output. As you turn the knob, the inputs are routed, in turn, from 1 to 4, to the output. Between each pair, we have some smooth interpolation to carefully blend two inputs as you scan between them, so you get a mix of 1 and 2, 2 and 3, and then 3 and 4. Stick in four different waveforms from an oscillator, and you can blend between them. Add some modulation in the CV input and you have a constantly varying waveform at the output. The Scan knob stops at 1 and 4 and doesn't cycle around like some similar scanner modules can. There are four LEDs peeking through the front panel window that follow the input scanning and also visualise the movement of one to another. I confess I expected the LEDs to be more dramatic, considering the design of the front panel and the gash through which you view them, but they are very gentle and reserved.

There are plenty of other uses for the scanning, especially if you plug a voltage sequence into the CV input. You could mix through modulations, leap between trigger patterns or remix four channels of audio, to name a few, and it will take CV or audio.

The Panning side has the opposite configuration, with a single input routed to four outputs. As you turn the knob, the source is sent to each output in turn, mixing from one to the other. The website suggests that this is quadrophonic panning,



which it is in that I can send a signal to four different speakers, but it isn't because I can only move from one speaker to the next in a straight line and not place it in space using a mix of all four speakers. It could be useful for moving a sound on a defined path, but as output 4 doesn't mix into output 1 it couldn't do a continuous loop; it can only go there and back again.

However, there are plenty of other things you can do with a single source and multiple destinations. You could route through different filters, or different effects modules, you could send triggers to different destinations or use it to route modulations through different parts of your rack.

Going in either direction, Astrax offers a wide range of modulation and mixing possibilities. It's the sort of module that grabs your attention and gives you ideas. It's comfortable to use, moves and mixes smoothly, and its striking physicality means you won't lose track of it amongst the chaos of your modular. It could have had a more dramatic lighting system though.

Robin Vincent

£ €289

W www.exay-one.com

“Astrax offers a wide range of modulation and mixing possibilities. It's the sort of module that grabs your attention and gives you ideas.”

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INSIDE TRACK

Carter Lang

Songwriter, producer and musician Carter Lang has built a stellar career on collaboration and a love of obscure hardware.

Carter Lang's laid-back, collaborative approach to writing and production has made him extremely successful. Growing up in Chicago, Lang started his musical career playing classical piano, guitar and bass before being introduced to synths, and to the Akai MPC 1000, in the late '00s. The beats he was making at home resulted in him becoming part of the Chicago hip-hop scene. "A friend sold me a Roland JX-8P, and the sounds were so mesmerising and warm! I then bought a Roland SH-2000, which is such a charming, interesting synth. That was my introduction to early '70s monosynths. It has a very classic sound.

"I was never super good in the box, and the MPC allowed me to start making my own compositions and arrangements. It was my introduction to rhythm, and sampling, especially of records. I worked with a rapper called Zack Wicks who really inspired me, but who sadly passed away while I was still

in school. He showed me Sun Ra and Frank Zappa and tons of esoteric music that the great producers were sampling. That got me curious about the sounds that were used on these records. Later when I got into pedals, I was able to combine them with synthesis, sampling and drum machines in a way that sounded good."

Going South

Lang went to New Orleans to attend the Loyola University music business programme, which allowed him to explore many different sounds and genres as it was "a laboratory of playing in different bands". On returning to Chicago, Lang started to work with the hip-hop collective Savemoney, particularly their members Chance the Rapper and Vic Mensa, as well as rapper Mick Jenkins. The MPC 1000 was still a crucial tool for him, one that allowed him to "differentiate" himself and "add value

PAUL TINGEN

"I'm categorised as a songwriter and a producer. But I also feel an artist in how I create sounds, and help bring an artist's vision to life. What I do is not about me. It's not about making a name for myself. It's about the music that comes out of collaboration. I don't care if it takes one person or five or 10, the question is always: how can we make music that gives us and other people the chills?

"Out of all the things I've worked on, I don't think there was one time that we thought, 'We need to make a hit today.' Instead, you land on an idea, it becomes a thing, and eventually hits sometimes happen. The question in that situation is always, 'How can I get everybody to be excited about what we're doing, without forcing anything?'"

to the writing process with interesting sonic flavours and rhythms". Not long afterwards, Lang's career took a step up when he began working with SZA in 2015, playing as a guitarist in her live band and co-writing and co-producing half of her debut and breakthrough album *Ctrl* (2017). Lang went on to co-write and co-produce Post Malone and Swae Lee's monster hit 'Sunflower' (2018), which is the seventh most streamed song of all time on Spotify.

"*Ctrl* was a dream and super defining for my career. SZA and I really crafted a sound together. Then 'Sunflower' came out of nowhere. It was based on a beat made by Louis Bell and I, and then our management pitched it to the *Spider-Man: Into The Spider-Verse* [2018] movie, and suddenly it's the main part of the movie. That song became larger than life, and a great calling card for me. It allowed me to move forwards in publishing and to acquire a lot of gear! It also prompted my move to LA in 2019, so I could be closer to many of the friends I was working with, and led to me establishing my current studio."

Since then, Lang has gone from strength to strength, being at the heart of the making of SZA's second album, *SOS* (2022), as well as the extended and reissued version, *Lana* (2024), and of Justin Bieber's albums *SWAG* and *SWAG II*. In addition, Lang has worked on tracks by Camila Cabello, Chance the Rapper, Doja Cat, Lil Nas X, Alicia Keys, Kali Uchis, Lola Young, Rosé and more. Lang has had a hand in more than 30 *Billboard*-charting songs, six of which went to number one. He has received 10 Grammy nominations and two Grammy Awards.

Comrades In Arms

Reflecting Lang's emphasis on working with friends, the credits of songs he's been involved in typically draw from a small group of co-writers and co-producers: Rob Bisel (see *SOS* March 2023), Blake Slatkin (*SOS* February 2023), Jared Solomon, Omer Fedi, Dylan Wiggins, Mk.gee, Dijon Duenas, Tobias Jess Jr and Eddie Benjamin.

"Of course, being a producer sometimes means being behind the computer and recording and editing my friends, instead of jumping in and always playing. I've found a lot of joy in watching the flow and discovery of people playing, and not feeling so eager to apply my own sound or cast my opinion on a song idea as it develops. Instead of saying, 'OK, we need to do this,' I encourage others to do what they do, while not losing sight of how everyone in the room is feeling."



Although he also uses Ableton Live, Carter Lang credits the Akai MPC with helping him carve out a unique identity.



Carter Lang's collection of rare vintage gear includes (clockwise from top left): Dynacord EC280 delay, Wersi Bass-Synthesizer, Korg Poly-Ensemble string machine and Olson Rhythm Beat drum machine.

» “As a producer, I read the room, and focus on everyone feeling stoked about the idea we’re working on, and not pressured to meet certain expectations. I watch the flow happen, and make sure that the artist is feeling it too, asking questions about what they like and want. It’s best to be vocal about it, instead of dealing with implicit expectations of needing to do this or that. I don’t want people to sacrifice their soul to make a hit. The aim is to make something that fits the artist’s world and that’s also fresh and new. I don’t like to contain myself or others in the creative process. When I hear something I like, I do positive affirmations. I don’t do a lot of negative affirmations. I let people play, and when it’s hot, let’s keep going.”

Beautiful Things

By way of example, Lang recalls the two songs he was involved in on Reneé Rapp's album *Bite Me*, a UK number one a few months ago. "Both songs began with a guitar riff, and my friends Jared [*Solomon*] and Omer [*Fedji*] embellished it with more guitars and tones. I also played guitar on 'I Can't Have You Around Anymore', and on 'Mad' I helped create the structure and arrangement of the song, and then Omer took it back to his place and dialled it in with Reneé, and made it into something beautiful."

SZA's latest hit 'BMF', from *Lana*, was made, recalls Lang, "with Omer and Blake [*Slatkin*]. It's based on an Omer guitar lick, and Blake played drums and I played bass. We made the initial beat a long time ago, and when we played it to SZA she did something amazing to it. I formatted the song in Ableton, and programmed some stuff and added a synth. This is a song that I really had a hand in. It's pretty minimal, and turned into something crazy with a hypnotic, poppy vibe."

On the Rosé song ‘Drinks Or Coffee’ from her hit album *Rosie*, “Dylan [Wiggins] and I manipulated samples in Ableton and then we played them on a keyboard. But in the end, there’s a mystery behind the sounds and behind how these songs were done, and I think that’s fun. People wonder, ‘What was that thing?’ I think it’s good to let people’s imaginations run wild. They can then hopefully do something that creates a mystery as well. Not everyone needs to be doing or using the exact same things.”

On the making of Bieber's *SWAG*, Lang notes, "There were some sessions we did with him in New York, in a friend's living room, where we walked away with



Lang's studio is home to a rare EML Synkey synth (top) and Freeman String Sumphonizer.

incredible emotions and a real sound. Everyone relinquished their expectations, we had a great rapport with each other, and something really magical happened during that week. I was playing guitar on some tracks, and bass, and we just let the room do what it does. Mk.gee, Dijon, Eddie [Benjamin] and I played the guitars and in fact all instruments on 'Daisies'. That track is pretty much 100 percent analogue.

"I had a really amazing opportunity to record a lot of Justin's vocals and to be able to arrange a lot of them. In no way was I controlling the process. I just happened to be able to record him super-fluidly in Ableton. The microphone that we were using most was the Shure SM7, and occasionally I'd bring my Telefunken ELA M 251. That was primarily the vibe, but it changed every time. There wasn't really one specific mic or signal chain that we used. It varied depending on the environment, and what gear was available. It was nice to see how his vocal sound changed on different mics and just inhabiting a space and seeing what each piece of gear had to offer. I later got the chance to bring a lot

of the material for the album back to my studio and do my Carter thing.”

Breathing Space

Lang's "Carter thing" is a creative and post-production process, again very laid-back, that consists of "layering and overdubbing at my studio, making sense of what's there, trying different tones, and so on. I might get on my drum kit and play and feel like time has stopped. No one has a gun to my head telling me, 'Do this in this amount of time.' I come back to things over the course of many days and months. It's not a rushed process. Sometimes something happens really quickly, and then I let it sit for a while and come back to it later. Mostly, I'm manipulating the audio that we have in Ableton, and I'm getting things to sound and feel like they are in their most natural place and not covered up by a bunch of other stuff. It's about letting things breathe."

All this is done in Lang's "madhouse" home studio in LA, most of it in a huge room full of daylight that's so choc-a-bloc with gear, that, he says, "I hardly have any space »

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» left to actually live here. Every table is full of stuff! The beautiful daylight reminds me of the openness of Rick Rubin's Shangri-La studio. That studio made a real impression on me in the years leading up to living here. This room here looks like a miniature Shangri-La. Or rather, a Strangely-La."

The latter is a reference to the abundance of weird and wonderful gear, which all ends up being sent through some more common gear, notably Lang's DAW of choice, Ableton, and "Universal Audio Apollo x8p and x16 soundcards, the older versions — they suit my way of working. I also work with a gnarly pair of ADAM A7X monitors. I like having small nearfields close to me.

"In the back of the house I have a couple of PMC 8-2 monitors. I'm having a new studio built at the moment, with a 48-channel API desk, a big live room, iso booths and so on, and the PMC's will sound great there, with the big blaring Augspurgers I'm having fitted. I'll be enveloped in sound, it'll be amazing! But for now, with the ADAMs I have no distractions. They just show me what I need to hear, really close to me."

The Good Stuff

Lang's more esoteric gear includes "a set of 10 mic pres from the late '70s, designed by the late and legendary Oliver Archut, who founded TAB Funkenwerk. He used to remake a lot of Telefunken and Neumann stuff. I bought these 10 mic pres from Riverside Studios in Cologne, and they told me that they used to be part of a one-off console Archut made for the West German Philharmonic Orchestra. They have EQs and really high gain. They're great.

"I have bought tons of mic pres over the years, that are waiting to be used in the new studio. I'm really happy with my remakes of the Neumann mic pres that are Danner cassette modules. There was a board of 40 channels, and I have 12 of them. The same with the Sennheiser mixer with KAR-15 preamps that was made for the Nagra portable recorders: high gain, Class A, with great distortion. They have a really nice drive to them, and I tend to use them for synths and drum machines. I also have tons of obscure DI boxes, that I use to send signals to different effects and split things out. I might send a signal to an effect box that has multiple outputs, which I may send to an amp and my pedalboard. I may have three or four different signals plus several room mics, just for one instrument."

The instruments in Strangely-La include "a Serge modular [synth], with all these banana plugs and so on. At one point I was thinking of using it to create a solo album that's very minimal and about textures that I find soothing. My Yamaha CS60 has been a real pleasure to work with the last few years. I love string machines, and have units like the Farfisa String Machine, the Freeman String Symphonizer, the Korg Polyphonic Ensemble PE-2000, and so on. Another favourite unit is the Steiner Crumar Master's Touch, which is a wind controller that I can use with my Moog, ARP and Oberheim synths. The Yamaha FX500 also is a really interesting

box, but it always breaks down, just like the Milano 3R-S Echo Chamber. I also have a Bel delay and sampler from the early '80s, and a Hawk tape delay.

"It was and remains exciting to find pieces of gear that can help make your sounds even better and richer, and raise those hairs. I started to uncover that by being influenced by people that have been in it for a while, like Mike D from the Beastie Boys and his sons, [producer/mixer] Norm Block, and Eric from Uniform Music. Digging into what different preamps and delays and other things sound like, and feeling I had to catch them all, has been really affirming my journey. Though I bought most of my gear



■ Vintage and modern gear side by side in Lang's rack. From top: Studer tape machine, Furman power conditioner, UA Apollo x8p and x16 audio interfaces, rack of 10 input channels made by Oliver Archut, and Eventide H3500 Harmonizer, with a pair of rackmounted Neumann input modules at the bottom.



■ Carter Lang's work with SZA and Justin Bieber has made him one of the hottest producers around.

person to person, I was also looking at eBay and Reverb, and at a certain point getting more gear felt like an arms race!

Smile Curves

"Soundscaping is all-encompassing: designing tones and chaining things together in ways that might seem unorthodox or backwards, but create a unique sound. The studio space I'm in now has afforded me to do that, because I prefer working with hardware for the initial point of creation. For that, analogue chains are just irreplaceable. They add an initial smile to the music. And I don't feel I have to replace anything later on, as opposed to when I work with plug-ins, and might have to replace the sound or layer it a few more times, just to get enough thickness and warmth or take away the aliasing.

"However, I get into the box as well, where I will use plug-ins to give me a vibe, whether it's a slap delay from a random UAD plug-in or utilising all the random plug-ins in Ableton, just to create different chains. I don't normally pull up a lot of software instruments, though I did so a little bit more this year, when I was in situations where I didn't have much of my gear with me and just got creative. I'd find a patch that I really liked and kept using that, and it became a sound. When I have to, I will work in the box as well for the initial creation point."

Lang also has an impressive collection of guitars and basses, and an extensive pedalboard that he created when part of SZA's band. "When working with SZA I wanted to be able to create different textures, and I built the pedalboard in a briefcase that I bought at a thrift store, so I could easily travel with it. I got for example a Strymon Volante, and some interesting delay looper pedals and got into looping

as well. These pedals also lit up the writing and recording sessions we were doing, with all sorts of textures that would trigger new ideas. At that point I realised that a single piece of gear can be a great starting point when writing with others, and can even become the crux of a creative process."

One obvious disadvantage of hardware is that it's not terribly portable, so for sessions away from home, he has to be selective. "Friends and artists definitely come to my place to write and record, and work with and experience all these instruments. But I also travel around a lot, and like to bring a random funky drum

Carter Lang: "What I bring is primarily about dedication to the room and to genuine friendships, and the capacity to see an idea through."

machine that's fun to manipulate, like a Roland TR-909 or the TR-808 or a Korg, or something weird that is going to help inspire a sound, like an Ace Tone. I'll also carry a guitar if there isn't a guitar there, a bass if there isn't a bass, and a small synth module or something, just to have some sounds immediately available. And I may bring my Denon DIC EM-1000 half-inch tape delay, which is super cool and almost impossible to find. Or my Quad Eight digital reverb, which is rare as well. It's really fun to mess with people's sounds using these units.

"Having said that, I often don't use any of them and may end up going with whatever the flow is. They are there as a safety net. If there's very little gear in a place, it may inspire me just to see what I can work with. I may just bring my laptop and rummage through my Ableton Finder window and try some keywords to see what random drum sound or random loop they bring

up. As I said, I'm not into soft synths much, but I sometimes use Arturia stuff, like their Yamaha DX7 or Minimoog Model D, if I don't have real ones around. The native Ableton plug-ins can be really inspiring to effect a sound that's already good before it goes into the box, but I tend to be very minimal with them."

Ideas Man

"I'm not known as the toys guy," concludes Lang. "What I bring is primarily about dedication to the room and to genuine friendships, and the capacity to see an idea through. I just want to have a good time with my friends and meet new people and bring relationships together. I'm really like a chameleon, adapting to the situations as they emerge, whether we're in my place, someone's home or a commercial studio. Wherever we are, it's about the attitude and expectations in the room. If

people are sitting with their arms crossed waiting for someone else to dance, it isn't fun. Everyone should have the feeling that they can contribute equally.

"The most common situation is that all music and lyrics are made in the room. Sometimes someone brings in a basic idea they have done before as a starting point, but it's rare that someone comes in with something fully prepared. Ninety percent of the time, we make it together. And then one of us will take it home to develop or finish the idea. I may get on the computer and work on the idea, or I may relinquish control and let someone else take it into their realm, and trust that they will respect the vibe. We all want to see each other win. When I work on it, I want everyone to be surprised by what I have done, but also to feel that they have been part of something and that it's a win for everyone. It brings me a lot of joy to spread that love." ■■■



Clavia Nord Organ 3

Digital Combo & Tonewheel Organ

The latest generation of Clavia's organ series introduces some pretty big changes.

GORDON REID

Some years ago, Hammond tonewheel organ emulators — often called 'clonewheels' — achieved such a degree of realism that (by and large) only those who played them could hear any significant difference between the original instruments and the emulations. Nowadays, you can stick the likes of a Nord C2D or Hammond SKX Pro in a fake wooden case, and nobody in the audience will be able to tell whether you're playing a fistful of zeros and ones or a menagerie of rotating thru'penny bits. But as players, we always want something new, something improved, something that woos us to hand over large wedges of cash for the promise of sounding better than ever before, and it's this as much as anything that forces manufacturers to carry

on developing existing product lines. If they didn't, they would fall by the wayside and some other bunch of Herberts would then get their hands on your money. Consequently, and notwithstanding component obsolescence, I have no doubt Clavia would have loved to keep selling us C2s and C2Ds *ad infinitum* and avoid the costs of further development and tooling up for manufacturing a new instrument. But today we have the Organ 3, which replaces the previous models and promises numerous updates and improvements over them.

The Organ Models & Effects

The underlying Hammond emulation in the Organ 3 sounds to me to be very similar to, if not the same as, that in the latest C2 organs but with some useful enhancements to the

surrounding technology. For example, the panel promises three overdrive types compared with the single type of the C2s (although the third isn't yet implemented so we'll have to wait to see what that offers). The existing two are less aggressive than before and, for me, are an improvement, offering a more organic growl when dialled in appropriately. There's also a new 'Deep' mode within the Hammond chorus effect. This sounds similar to the chorus on my C2, so perhaps it's the shallow mode that Clavia added. Either way, there are now more chorus/vibrato options, which can't be a bad thing. The next enhancement is a Soft B3 option, which, at the touch of a button, removes the percussion, key click and leakage to produce the unsullied sound of the virtual tonewheels. If you're looking for the cleanest representation of a tonewheel



generator, this is a quick and easy way to obtain it.

Unlike its Hammond model, Clavia's Vox and Farfisa emulations have not yet reached the point of convincing me that I'm listening to the original instruments. Indeed, I long ago stopped liberating my large, heavy and fragile Vox Continental II and Farfisa Compact Duo from storage to make detailed comparisons with modern instruments because the result was always that the emulations were Vox-ish or Farfisa-ish. But Clavia have promised an "updated Vox emulation" in the Organ 3, and a "fully revised Farfisa emulation, with improvements to the tone generators, filters and dynamic behaviours such as compression and key click".

When testing the Vox model, I was surprised to find that the method for mixing the sine and triangle tones (which, traditionally, is achieved by pulling out two drawbars by the desired amounts)

has been changed on the Organ 3. This offers a single drawbar with the pure sine wave at one extreme and the triangle at the other, which means that you can't control the volume by pulling the waveform drawbars out by the desired amount; you can only control the waveform mix. I'm not sure that I like this. When compared with the C2D, there are also changes in the way that octaves 'fold' at the upper and lower extremes of the keyboard. (Mind you, the Continental II only had four-octave keybeds and a more limited range of footages, so folding wasn't an issue.) Ultimately, the Nord's drawbars don't reflect those of the original precisely but provide additional voicing options, which is good. Happily, the V3 chorus/vibrato setting does a pretty good job of emulating the original vibrato so, if you add some overdrive and use suitable EQ, you can approach the sound of the Conti. Nevertheless, I doubt that you'll ever feel that you've perfectly imitated 'I'm A Believer' or 'The House Of The Rising Sun'.

While the control mechanism remains the same as before — using the halfway point of a drawbar as a simulacrum of flicking a tab up or down — there are differences in the Farfisa model when comparing the C2D with the Organ 3. Most noticeably, the 16' sounds fold on the C2 but don't on the Organ 3, which means that you can now obtain greater bass extension. Unfortunately, as with previous Nord's, the Organ 3 lacks the Multi Tone Booster tabs that generated the Farfisa's most aggressive and instantly recognisable sounds. To be fair, the underlying tone can be subtly brighter on the new model, which is a good thing, but the Organ 3 remains too polite to be completely convincing. Sixty-year-old Farfisa Compacts tend to be brash lumps of noisy, distort-y gnarliness, whereas emulations always sound to me as if they've been to a Swiss finishing school, joined the local yacht club, and would never be seen serving biscuits without a doily.

Now we come to something that Clavia don't boast about in their marketing materials. The lovely sample-based Pipe Organ of the C2 and C2D has been reduced to a single, modelled Principal (diapason) sound. You can add ranks using the drawbars to create a range of registrations, and the onboard EQ can help you to obtain a brassier sound, but the Flutes, Trumpets, Vox Celeste, Nazards and all of the others nestling in the C2

Clavia Nord Organ 3

£3206

PROS

- There's a lot to be said for concentrating on doing one thing well.
- It's cool to be seen playing a Nord organ on stage.
- Numerous incremental improvements in the Hammond, Vox and Farfisa emulations make it a better product than the C2 series...

CONS

- ... except that the pipe organ section has been completely gutted (as am I).
- Neither the organ itself nor its accessories are cheap.

SUMMARY

The Organ 3 is a first-class instrument that will appeal to many players but I fear that, by allowing other manufacturers to catch up and in some ways overtake it, Clavia are at risk of losing their position as the leader of the clonewheel pack.

series have been discarded. I was hugely disappointed when I discovered this, so I asked the Product Specialists at Clavia why they had done it. They told me that the company had dispensed with the sampled pipe organ sounds to focus the Organ 3 on the Hammond, Vox and Farfisa organ models, and to simplify the control panel. Hmm... you could simplify my car's dashboard by removing the multi-speaker sound system and the air conditioning, and I wouldn't like that either. Unfortunately, even the surviving sound isn't perfect because it exhibits aliasing at the top end. All in all, you can treat the Organ 3 as a pipe-organ-free zone. You might wonder if this has any relevance to rock & roll but, while writing this review, I watched a performance of Bonobo performing 'Otomo' live at the Royal Albert Hall with Anna Lapwood playing the mighty 'Voice Of Jupiter' organ. The same evening, I also watched a recording of Yes' classic track 'Awaken', in which the Nord C2D pipe organ worked as a perfect replacement for the genuine pipe organ played on the album. Whether it's dance or prog, there's a place for pipe organs in modern music.

Happily, the flexibility of the bass pedal sounds has gone in the other (ie. good) direction. The C2 generated appropriate pedal sounds depending upon the organ model chosen but, on the Organ 3, you can choose which model you would like to play at any given moment. I suspect that the Trek II and Moog Taurus options will find many uses. Happily, there's also

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» an FX Bypass button in this section, so you can apply the Leslie effect to the organ manuals while your bass pedals remain unmolested.

The next bunch of changes lie in the various effects. Clavia say that the Organ 3 includes a new rotary speaker emulation that promises “further increased resolution for the Rotary Speaker horn and bass rotor model”. Since I had no idea what that meant, I asked the Product Specialists who responded that this relates to the amount of detail in the frequency domain (I’m not sure that that’s much clearer) and how closely the model matches the sound and behaviour of the original speaker. Whatever the explanation, it sounds nice so I’m not complaining. The other significant change here is the replacement of the previous three rotary speaker models — two Leslie 122s and a Leslie 145 — with a single model that offers five microphone positioning options.

The delay effect has also been updated, and you can apply this to the upper manual or the whole instrument, which makes it possible to play psychedelic Pink Floyd solos on the upper manual while obtaining clean accompaniments on the lower. Perhaps even more significant are the improvements to the reverb section. There’s now a Chorale mode that choruses the ‘wet’ signal, and a spring model that, when wanted, you can place before the rotary speaker effect. I’ve been crying out for the second of these for as long as I’ve been reviewing digital organs because the sound of a reverberated signal passing through a rotary speaker is quite different from the sound of the same speaker in a reverberant space.

Round the back we find a quarter-inch input for Rotary Control (either via a pedal or the optional Half Moon Switch), a USB-B port, inputs for Swell and Sustain pedals, two 5-pin DIN ports for MIDI I/O and an extra one for the optional Pedal Keys, a selection of audio outs, a 3.5mm monitor input and, at the farthest right, an 11-pin Leslie connector and accompanying High Level output. Phew.

The final upgrade in the existing effects sections is an additional amp model. As always, the name alludes to its inspiration without being specific. Which British tube amp and 4x12 speaker do you think that they mean?

Clavia have also added a modulation effects section derived from its other product families, and this allows you to apply a phaser, flanger, chorus, tremolo or ring modulation to the upper manual or the whole instrument. The amount of control is sparse (just rate and amount) so these effects would be more useful if they offered at least the same degree of control as a stompbox, but they sound fine within their limits.

In Use

I’m not going to discuss the minutiae of things such as inputs and outputs (other than to say that the new auxiliary output, which can carry the bass sound separately from the rest, is a notable improvement), the patch memories or the updated method for controlling registrations using the LED-equipped drawbars, mostly because I don’t have the space. Nonetheless, there are a few other aspects of the Organ 3 that deserve mention.

The first is its shape; while it’s of similar size and weight to its predecessors, it’s an inch wider. This may not sound like much, but it means that it won’t fit into a hard C1 or C2 flightcase unless you terrify the foam with a Stanley knife. Similarly, while the panel is similar to that of the C2D, you

won’t be able to jump instantly from the older model to the newer because many of the common controls have moved. I don’t want to overstate this; if you can use a C2D, you should be comfortable with the Organ 3 in a day or two, but don’t forget which model you’re playing or you could find yourself grabbing the wrong wotsit.

Inexplicably, Clavia describe the Organ 3’s screen as a “large OLED display”. Don’t be silly. The screen on my OASYS is (by keyboard standards) large; the one on the Organ 3 is tiny but, nonetheless, a step up from previous Nord organs, and the new menus are much quicker and simpler to navigate than before, so let’s be grateful and overlook the hyperbole.

As on earlier models, the manuals have a High trigger mode that imitates the response of a vintage Hammond organ, but you can’t select the High mode for local playing and simultaneously use Normal mode for playing external MIDI devices as you could on the Nord C1. What’s more, there’s no emulation of the slightly staggered activation of the drawbar pitches that occurs when you depress the keys slowly on the likes of an A100, B3 or C3. Some alternatives to the Nord are now offering this and, while it’s a minor detail, it would be nice to see it here. It’s also worth noting that, as on previous models, MIDI velocity appears to be sent and received, but the manual isn’t clear about this. It says, “Note On and Note Off messages are transmitted and received. Velocity is included if the related MIDI menu setting is set accordingly,” but there’s no relevant menu in the current OS. Having said that, the Organ 3 operating system has only reached version 0.98 at the time of writing, which suggests that it’s still a work in progress that hasn’t even achieved the v1.00 status normally afforded to initial releases. Hopefully, this will have been sorted out by the time that you read this.

Finally, I feel obliged to note that pressure sensitivity (aftertouch) is not

Extras

Although you can use the Organ 3 ‘out of the box’, there are numerous extras that enhance its abilities. Perhaps the most important of these is the Half Moon Switch that determines the speed of the rotary speaker emulation and can also control a genuine Leslie speaker connected via the 11-pin connection on the rear panel. Others include a stand, a soft case, and two expression pedals. But if you want the full organ experience, you’ll also need the Nord Pedal Keys 25 floor pedal that’s powered by the Organ 3 itself via the MIDI cable that connects the two. (An external PSU is provided for those times that you may wish to use it as a standalone MIDI pedal.) Unfortunately, all of these devices come at an extra cost, with the pedals adding a brain-melting £2150 to the price of the Organ 3 itself.

provided. You may wonder why I would want aftertouch on an organ, but I find it useful for controlling things such as the rotary speaker speed when I have neither hands nor feet free for pushing buttons, switching switches or stomping on pedals.

Conclusions

The Nord C1 proved to be a great success in the '00s, overshadowing serious competition including the Korg BX3, the Roland VK-88 and the Viscount/Oberheim OB5. The series then retained its eminence with the C2 and then the C2D. But the competition is tougher today, and includes the similarly-priced Hammond SKX Pro (SOS August 2022) and Viscount Legend Soul 261, as well as the top-of-the-range dual-manual version of the Hammond XK5, all of which have fans who find them superior to Clavia's offerings. So I fear that this was the wrong time for Clavia to be amputating bits. I would have been tempted to retain all of the facilities of the C2D and then added other organ models — say, the Hammond L100 series — and maybe some Nord Sample Synth capabilities that would have made it a far more flexible instrument without diluting the original concept. The ability to assign a different organ model to each manual would also have been a significant improvement, but I've asked Clavia's Product Specialists about this

before and, perhaps because of technical limitations, it has never reached their to-do list.

So, will I be upgrading to the Organ 3? No, I value the pipe organ in my C2 too highly. But would I sell a C2 or C2D and upgrade to the Organ 3 if I didn't use the pipe organ? If I'm honest, probably not. While there's no doubt that the Organ 3 looks slightly more attractive, has slightly improved tonewheel and transistor organ sounds and is a slightly nicer instrument to use, I don't find that the differences would justify selling what was, until

“These iconic red slabs of Swedish engineering have become deeply trusted, and there's a great deal to be said for being confident that your instrument will do exactly what you ask, day in and day out for years to come.”

recently, the 'ultimate' clonewheel organ and then having to find another couple of grand to buy the latest model. So perhaps a better question to ask is whether I would choose the Organ 3 if I was buying for the first time. That's a trickier one. I've not played the Viscounts, so I'm not in a position to comment on them, but I know the Hammonds well and the SKX Pro, while perhaps less elegant than the Organ 3, is a superb stage instrument with hugely configurable organ engines plus additional synths and an array of advanced features that the Nord lacks.

Then there's the issue of price. When the C2 appeared it cost around £2200 and boasted four organs that therefore cost around £550 each. Today, the Organ 3 costs around £3200 and offers just three organs (I don't include the gutted pipe organ) for nearly £1100 each. To be fair, inflation accounts for much of this increase but, on this basis, the Organ 3 is still more expensive in real terms than its predecessors. You might therefore wonder whether it can carve a niche for itself in today's marketplace. Despite everything,

I believe that it can and will. Over the past 30 years, these iconic red slabs of Swedish engineering have become deeply trusted, and there's a great deal to be said for being confident that your instrument will

do exactly what you ask, day in and day out for years to come. Nevertheless, I recommend that you try the alternatives before splashing any cash. You currently have a choice of at least four excellent instruments, and that can't be a bad thing. If you stick to the Hammond emulation then, whichever you choose, the audience need never know that you're not playing the 'real thing'. ■■■

£ £3206 including VAT.
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How strong are you feeling? The Nord Organ 3 measures 985 x 165 x 446mm and weighs a reassuringly hefty 17kg.

We dive deeper into the theory and practice of hybrid arrays, and consider the benefits of using more than one array.

Part 7: Advanced Hybrid & Combination Stereo Arrays

This is Stereo!



HUGH ROBJOHNS

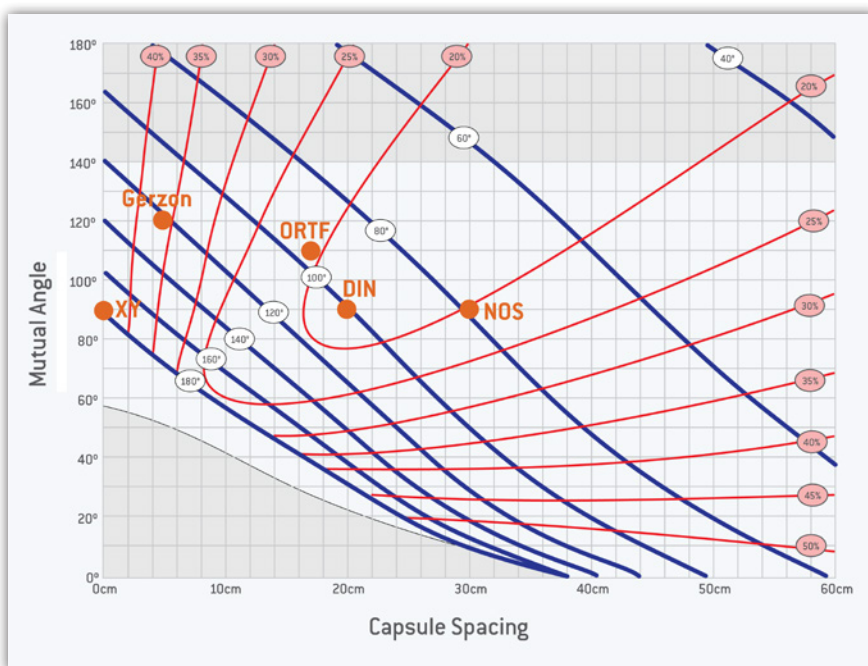
In the previous part of this series, I introduced the concept of hybrid stereo mic arrays, and explored some of the initial practical processes you can undertake to choose either an established named array — all of which are defined precisely — or to adapt the idea to tailor your own to the recording task in hand. This time, I want to dive a little deeper into the subject, and we'll start with some useful tools that can inform your choice.

Visualisation

In Part 4, I mentioned the advantages of visualisation tools for optimising the SRA and capsule configurations, and the increased parameter interaction when tweaking hybrid arrays means good visualisation tools become even more helpful. However, there's another way of looking at these things that I've found very educational and informative: The Stereophonic Zoom. This was the title of an academic paper presented by Michael Williams to the AES Convention in London in 1984, and it has been enhanced by its author ever since, culminating in the excellent *Multichannel Microphone Array*

Design book that I reviewed a couple of years ago: www.soundonsound.com/reviews/multichannel-microphone-array-design. That book covers a far wider range of mic configurations and applications

than we've been exploring here, though, so perhaps an easier introduction to the concepts can be found in an article on Grace Design's website: www.gracedesign.com/support/StereoZoom10.pdf.



■ The capsule spacing and mutual angle of various 'name' hybrid arrays. The shaded areas (top and bottom) indicate regions where the direct/reverb ratio varies across the stereo image.

Michael has, very generously, also published the associated graphs for a range of stereo arrays using different polar patterns on his website (www.williamsmmad.com/MAD/2ch/2ch.htm). I've redrawn one of his graphs for cardioid capsules configured in a stereo array (opposite). The capsule spacing is shown along the bottom axis and the mutual angle along the vertical axis. The blue lines indicate the SRAs provided by different combinations of capsule spacing and mutual angle, while the orange blobs identify the parameters employed by the named hybrid arrays discussed previously.

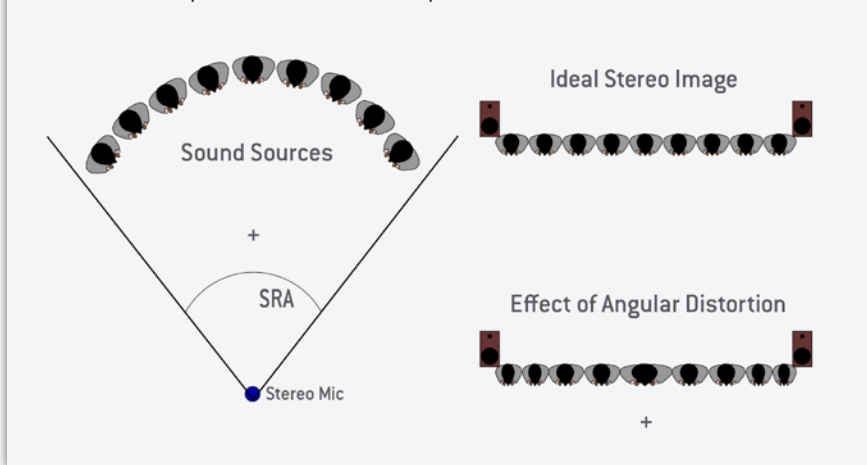
As you can see, the X-Y coincident cardioid array, with zero capsule spacing and a mutual angle of 90 degrees, lies close to the blue 180-degree SRA curve. The DIN array, with its 20cm capsule spacing and 90-degree mutual angle, is offset to the right of the X-Y array, sitting close to the blue 100-degree SRA curve. Hopefully, examining this graph will help to make a little clearer how reducing the capsule spacing moves horizontally leftwards across the blue lines of increasing SRA and, similarly, how reducing the mutual angle moves vertically downwards across the increasing SRA lines. The shaded areas at the top and bottom indicate the regions where the direct/reverb ratio varies across the stereo image. In the upper area there tends to be too much reverberation in the central zone of the stereo image, and the direct sound tends towards a 'hole-in-the-middle' effect. In the lower shaded area, the reverberation tends to increase and 'pool' towards the outer extremities of the stereo image. So, choosing parameters in these areas is not recommended.

Angular Distortion

The red lines, roughly perpendicular to the blue SRA curves, represent the 'angular distortion' created by the array, indicated here as a percentage error from the true position. Angular distortion, in this context, is the variance between the physical angles between sources in front of the microphones, and their perceived relative angles when replayed over loudspeakers. It's analogous to the optical distortion created by a wide-angle camera lens when it's placed close to a subject.

As a general rule, most stereo mic arrays tend to expand the soundstage outwards — they push sources located away from the centre of the stereo image even further out towards the speakers, thus stretching the centre and 'crushing' the extremities. It's usually a pretty subtle effect, but it's

■ An illustration of angular distortion: sources further from the centre tend to be pushed outward towards the speakers.



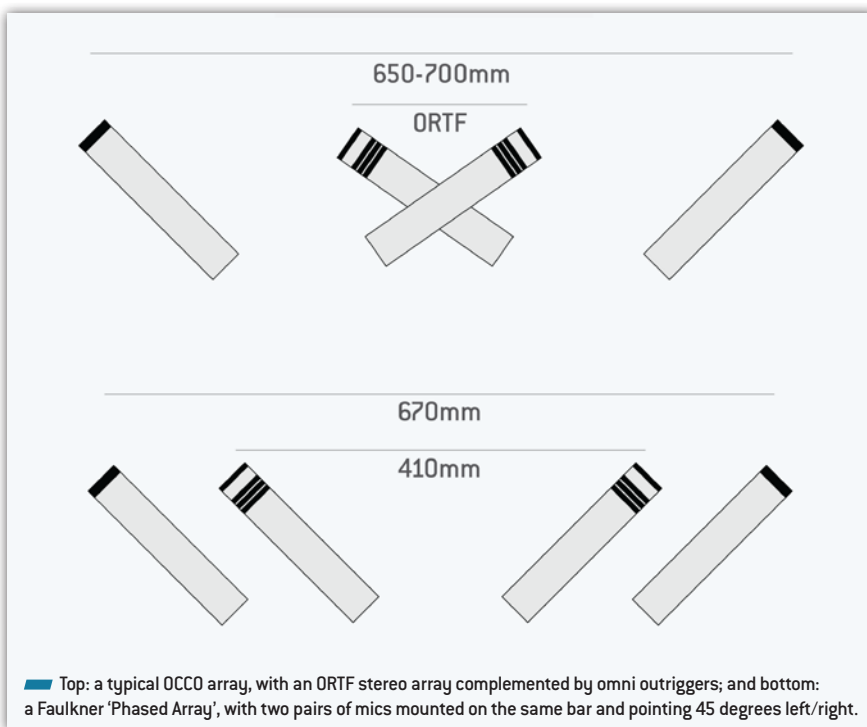
certainly audible when comparing different array configurations in good monitoring conditions. It can also become noticeable when close-miking a soloist who plays in an animated way, as any small physical movements can sound rather exaggerated when auditioned on the speakers.

The optimum conditions to minimise angular distortion vary with different polar patterns in the array, but for cardioid microphones the angular distortion is minimised by using moderate mutual angles (80-110 degrees) and capsule spacings (20-30 cm). In other words, hybrid arrays produce slightly less angular distortion than coincident arrays, and almost half as much

angular distortion as spaced arrays. So it seems that capturing both time and intensity differences between channels is better than just time or intensity alone.

As can be seen on the graph, the DIN array has the least angular distortion of all the hybrid arrays I've discussed so far, with NOS and ORTF both having a little more angular distortion. Not surprisingly, the coincident cardioids and Gerzon array have considerably more. In practice, this angular distortion is rarely even noticed, let alone problematic, but it's still interesting to note how changing parameters affects this aspect of stereo-imaging accuracy. This angular distortion is also illustrated

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» in the Sengpiel Visualisation website mentioned in previous articles (<https://sengpielaudio.com/HejaE.htm>). The five equally spaced coloured radial marks within the shaded SRA represent equidistant sound sources, and are replicated on a line between the two loudspeakers above, their spacing indicating the angular distortion created by the selected mic array configuration.

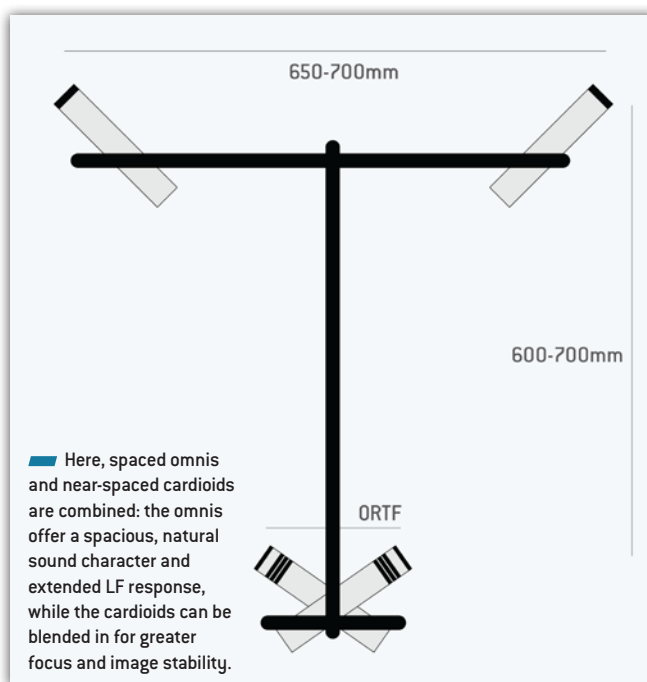
Combination Arrays

Spaciousness in a stereo recording is always a very attractive and desirable feature, but as I hope has become clear, achieving a pleasing level of spaciousness will often compromise the stability and precision of the stereo imaging. Wouldn't it be nice if there was a way to achieve the best of both worlds? Well, there is... but it requires additional microphones and might make the stereo purists go weak at the knees!

The main risk, when adding extra microphones to an array, is the introduction of large phase differences between channels, which can then cause unwanted colorations (comb-filtering effects) when summed to mono. But if the level of the extra microphones is kept low (around 8dB below the main mics) those phase differences are unlikely to cause major problems.

So, why not rig a main pair using a near-spaced hybrid stereo array known for reasonably accurate stereo imaging — ORTF, for example — and supplement the spaciousness by adding more widely-spaced omnidirectional mics a little lower in the mix? The use of omnidirectional capsules also has the potential advantage of improving the low-frequency extension, making up for the known weakness in that area of cardioid microphones. Once again, there are many variations on this core theme, but a particularly popular configuration has become known as the OCCO array. The letters indicate the layout of two pairs each of omni and cardioid capsules. I've occasionally seen it called the 'Optimal Cardioid Capsule Configuration', although that seems rather lacking given the array requires omni capsules too!

I've never seen a precise definition of the OCCO array, but most configurations



employ the cardioids in a near-spaced format, usually ORTF (but occasionally DIN or NOS) at the centre, with the outer omni capsules usually pointing outwards at around 45 degrees and spaced as wide as the shared mounting bar allows (I typically use 660-690 mm). APE spheres on the omnidirectional mics can be useful,

“The main risk, when adding extra microphones to an array, is the introduction of large phase differences between channels.”

if available, to enhance their stereo imaging definition (as in the Decca Tree arrangement — see the separate box).

Typically, the ORTF (or similar) cardioid pair would be considered the main stereo array, and its position and settings would be optimised and balanced accordingly — but ‘erring’ towards a slightly dry balance, as the omni outriggers will make the mix more reverberant. The outer omnis are then blended in to the mix, usually 8-10 dB below the ORTF pair, to add the required spaciousness, ambience and LF extension. An alternative but equally valid approach is to consider the spaced omnis as the main stereo array, optimised for perspective and imaging in the usual way. The central

cardioids can then be blended in just to add a little focus and positional definition to the mix. Again, to avoid the risk of coloration in a summed mono mix, keep the second pair 8-10 dB lower in the mix than the main omnis.

Tony Faulkner, who I've mentioned previously in this series, has taken this idea further with an arrangement he calls a 'Phased Array' — a term borrowed from sophisticated techniques employed with multiple radio antennae, as well as in military radar and sonar systems. The idea is that by controlling the relative phase relationships between transducers, it's possible to 'steer' the directionality or focus of the entire array, although I'm not convinced that's really what is going on in this application.

The particular configuration Tony suggests, which he has used for several orchestral recordings, places all four mics on the same bar, angled 45 degrees left and right as shown in the diagram on the first page of this article, with the outer omnis spaced at 670mm and the inner cardioids at 410mm. With such a wide spacing, the SRA of the cardioids is only around 67 degrees, while the omnis have an SRA of around 100 degrees. Essentially the omnis act as the main stereo array, with the ability to blend the cardioids in to add focus and clarity to the centre of the orchestra as a kind of audio zoom control. Like the OCCO arrangement, this kind of multi-mic configuration is neat and quick to rig — it requires only one mic stand — but it affords a useful degree of flexibility in the output sound character, in terms of perspective, spaciousness and imaging definition.

Separated Arrays

Of course, there's no rule that insists the omnis and cardioids must be mounted on the same bar; that's just a convenience. Often the near-spaced array and the spaced omnis are deliberately separated, with one pair placed in front of the other (usually at the same height). This requires a more substantial and sophisticated mounting rig, but it's a worthwhile investment because it opens up a whole new range of options.

There are multiple solutions to the mounting mechanics, but my preference is to use a pair of Manfrotto 154b mic bars. The first bar is mounted facing fore-aft on a tall Manfrotto light stand (1004BAC), allowing a stable 3.6 metres of height (when weighted adequately with sandbags!). The second bar is suspended from one end of the first, facing left-right with the omni mics near each end. At the other end of the fore-aft bar, I hang a short stereo bar for the near-spaced cardioids. The fore-aft bar is then adjusted to optimise the balance, keeping the centre of gravity of the whole assembly directly over the mic-stand centre. The Manfrotto mic bar is 650mm long, which is enough for most spaced-omni setups, but the supplied tubes can be easily swapped out for longer 25mm aluminium or carbon-fibre tubes should greater mic spacing be required. The same idea can be used to

build cost-effective and secure Decca Tree configurations (see the separate box).

An example of an array using separated spaced-omni and cardioid arrays is with the omnis spaced 600-700 mm apart and positioned around 600mm in front of the near-spaced pair (which could be in ORTF or DIN configuration). The idea here is to match the perspectives captured by each mic array — the near-spaced cardioid pair inherently captures less reverb than the spaced omnis, so needs to be mounted at a greater distance from the source to have a similar amount of reverberation. Like the 'Phased Array', the omnis would typically be employed as the primary pair for their spacious and natural sound character with extended low-frequency reach, and the near-spaced cardioids blended in to supplement focus and imaging stability.

An equally valid yet reversed approach is to mount the near-spaced

cardioids at the front, closer to the source, with the spaced omnis around 600mm (or more) behind. In this case the near-spaced cardioids provide the main stereo sound and the omnis are blended in to give more room ambience. This configuration has the advantage that the increased separation of the omnis behind the cardioids reduces coherence between the two stereo pairs, allowing a higher level of omnis in the mix before coloration occurs in a mono sum.

Next Month

In the next part of this series I'll look at some of the post-production processing options that can be used to further enhance or correct the stereo imaging obtained by the myriad coincident, spaced, hybrid and separated stereo arrays I've described throughout this series. ■■■

The Decca Tree

When discussing the concepts and applications of spaced omnidirectional microphones in this and the previous article in this series, I have deliberately avoided all mention of the Decca Tree array — even though it employs spaced omnidirectional microphones. The reason is that the true Decca Tree arrangement is actually a lot more complex than many realise, and requires specialist omnidirectional microphones fitted with Acoustic Pressure Equalisation spheres to work properly. The technique really deserves an entire article all of its own, but if you are interested in learning more about it I can highly recommend the book *Classical Recording: A Practical Guide In The Decca Tradition* by Haigh, Dunkerly and Rogers: www.soundonsound.com/reviews/classical-recording-practical-guide-decca-tradition

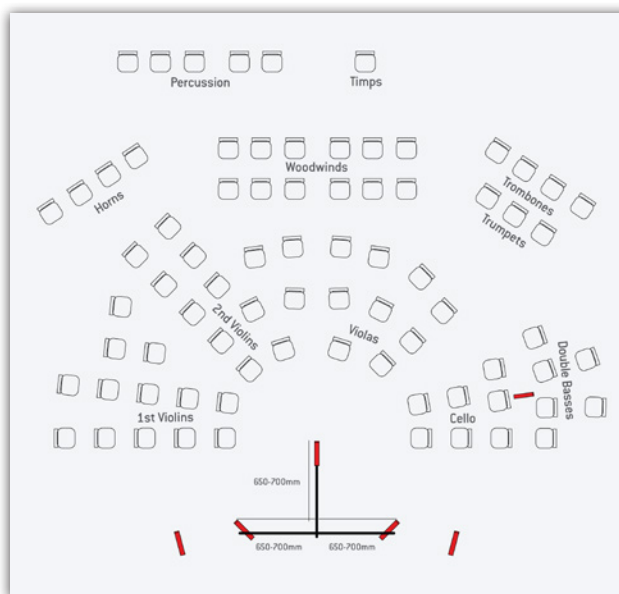
Most think of the Decca Tree as a simple three-mic array but, when employed for large orchestral recordings it's really a complicated six-mic array. The familiar three-mic tree is suspended above the conductor, but it also requires two additional omni outriggers positioned left and right (between the second and third desks of violins and cellos), plus a sixth mic positioned above the double basses. That combination, along with the specific microphones used, delivers the famous Decca Tree's sound character.

The dimensions for the central 'tree' vary depending on the situation, but for a typical orchestral setup the left and right mics are generally spaced around 1300-1400 mm apart (although I have seen wider spacing in some commercial recordings), with the centre mic placed forward by half of the left-right mic spacing.

For a small ensemble, like a string quartet, a simpler three-mic tree can be employed, with the left-right mic spacing reduced to around 600mm, and the centre mic around 150-200 mm forward. However, I've often found this mini-tree approach less pleasing than other hybrid arrays for recording small ensembles.

Mounting the centre mic forward of the left and right pair is a very clever design feature of the Decca Tree. Spaced omni microphones can easily form a hole in the centre of the stereo image if spaced too widely, and the centre mic neatly fills that potential hole in the sound coverage. But more than that, by being slightly forward of the other two it captures sound fractionally earlier by a millisecond or two, greatly enhancing the stability of the stereo image because the reproduction over loudspeakers builds the stereo image from the centre outwards.

Traditionally, the omni mics employed in a Decca Tree are Neumann M50s, which house the omni capsule within a 40mm sphere, and this is critical to the performance of the array. The sphere



■ A full Decca Tree setup, with a central three-mic tree joined by a pair of omni outriggers on either side, and a sixth mic placed above the double basses.

not only boosts the microphone's high-frequency response but also narrows the directivity at high frequencies. So, these omni's are directional, and this is the reason why the omni mics are angled left/right at roughly 45 degrees from the centre. In the six-mic array, the outriggers are also angled slightly outwards by about 15 degrees, and this capsule angling is critically important in generating a coherent stereo image.

Most small-diaphragm omni microphones can be fitted with APEs — either as manufacturer accessories or as 3D-printed options — and I strongly recommend this approach if you don't have access to M50s or equivalents. At a push, subcardioids or even cardioids can be used instead, maintaining the capsule spacing and HF directionality, but at the cost of the LF extension and phase linearity associated with omnis.



DPA 4060 & 4099 Core+

Microphones

DPA's new Core+ technology represents a significant upgrade for their already class-leading miniature microphones.

BOB THOMAS

Released in 1995, the DPA 4060 and 4061 not only propelled Danish Pro Audio (as DPA Microphones were known back then) onto the world stage, but also redefined expectations of what miniature microphones were capable of. By combining measurement microphone and hearing aid technology, DPA created a capsule for those microphones that became the foundation of their miniature microphones for many years. The 4099 supercardioid instrument microphone, released in 2009, married a 4060-type capsule with an interference tube to produce the world's first miniature shotgun microphone.

In 2018, DPA introduced their Core technology, with the aim of reducing harmonic distortion at higher sound pressure levels. That was incorporated in

the d:screet 4060 and the d:vote 4099, which were released later that year. Most recently, in January 2025, DPA announced their new Core+ by DPA technology, which they describe as "a groundbreaking, patent-pending innovation that sets a new standard for distortion-free microphone sound". At the same time, DPA also introduced the new MicroLock miniature locking connectors, which are backwards- and forwards-compatible with DPA's own existing MicroDot connectors, and an extensive and comprehensive range of new and revised stand and instrument mounts for 4060 and 4099 series microphones. The first microphone to benefit from DPA's latest technological development is the DPA 4099 Core+, which made its debut at InfoComm in June 2025, and was followed, shortly thereafter, by the DPA 4060 Core+.

Evolving Technology

The driving force behind the Core and Core+ technologies was and is DPA Microphones' desire to obtain the clearest possible sound from their microphones by minimising distortion and increasing dynamic range. The original Core development focused on the impedance converter that sits inside the electret

■ The 4099 Core+ with gooseneck and piano mount.

microphone capsule. DPA engineers implemented a fully compensated circuit topology that linearised the performance of the impedance converter. This resulted in a significant reduction in distortion, increasing the SPL that the capsule could handle before its THD reaches 1% — the point at which a loss of sound quality will be perceived by most listeners.

DPA 4060 & 4099 Core+ £444/£450

PROS

- Impressive absence of harmonic distortion across virtually the entirety of their dynamic ranges.
- Wide and versatile range of mounting options.
- IP58 rating means that dust and liquids are unlikely ever to be a problem.

CONS

- None.

SUMMARY

DPA's 4060 Core+ and 4099 Core+ miniature microphones deliver an impressive performance that is clean and distortion-free at any realistic SPL.

DPA's engineers then turned their attention to the analysis and correction of the distortion that arises from non-linear displacement of the microphone capsule diaphragm. With a very small volume of air such as that located behind the diaphragm of a capacitor microphone, the forces acting on the diaphragm are different when the diaphragm is forced inwards compared to when it is being sucked outwards. The patent-pending Core+ design corrects these non-linearities in diaphragm displacement, right up to SPLs that approach the 1% THD level of the electronics. The combination of Core impedance converter linearisation and Core+ diaphragm linearisation is said to result in essentially unmeasurable distortion, from the level of the microphone capsule's self-noise up to a few dB below clipping.

4060

The frequency response ranges, sensitivity and typical A-weighted equivalent noise levels of the original 4060, the Core Normal SPL and Core+ Normal SPL variants are identical. (Ever since the Core range, the 4060 is available in Normal, Loud and Extreme SPL versions; the 4099 comes in either Loud or Extreme SPL.) All three generations of 4060 have specified frequency responses of 20Hz-20kHz (± 2 dB), with their removable soft boost grids adding a 3dB boost between 8-20 kHz and their high boost grids adding a 10dB boost at 12kHz. Their sensitivities and self-noise

figures are also identical, at 20mV/Pa and 23dB A-weighted, respectively.

Where differences do appear are in the total harmonic distortion (THD) and typical dynamic range specifications. The original 4060 had a THD of under 1% up to 123dB SPL peak and a typical dynamic range of 100dB. The d:screet Core Normal SPL version improved both figures by 6dB, and the Core+ Normal SPL improves things yet further, with a THD of under 1% up to 132dB SPL, and a typical dynamic range of 109dB.

4099

The original 4099 had a specified frequency range at 20cm of 80Hz-15kHz (± 3 dB) with a +2dB soft boost at 10-12 kHz and a nominal sensitivity of 6mV/Pa. This sensitivity is the same as those of the Loud SPL versions of both the 4099 Core and Core+. Both the Loud SPL Core and Core+ versions of the 4099 have specified effective frequency ranges at 20cm of 80Hz-15kHz (Core) and 80Hz-17kHz (Core+), both with a +2dB soft boost at 10-12 kHz. These figures, together with the three variants' typical A-weighted self-noise levels of 23dBA, highlight these microphones' similarities, rather than illustrating their differences — despite the 2kHz frequency range extension in the 4099 Core+.

As a footnote to the above, these mics are obviously designed for very close placement to the sound source, where the supercardioid pattern's proximity

Making Connections

DPA's MicroDot connectors feel as though they have been around forever, which means that there is a massive installed base of cables and adaptors out there. Rather than simply succumb to that inertia, DPA's new spring-loaded MicroLock locking connector system is fully backwards-compatible with MicroDot connectors in both male and female directions. The only caveat is that the locking function is available only between two MicroLock connectors.

effect inherently boosts low frequencies. Consequently, DPA have tailored the frequency responses of these mics to be nominally flat at a distance of 20cm.

As in the 4060, the differences between the three versions of the 4099 are seen in their THD, typical dynamic range and SPL specifications. The original 4099 had a THD of under 1% up to 123dB SPL and a typical dynamic range of 100dB. The Core Loud SPL variant had equivalent figures of 131dB and 108dB, while for the 4099 Core+ Loud SPL they are 141dB and 118dB. So, the Core Loud SPL 4099 has a dynamic range 8dB greater than that of the original 4099, while the Core+ Loud SPL 4099's dynamic range is 10dB greater still.

Mount & Go

The arrival of the 4099 Core+ was also marked by a revised and expanded set of clips and mounts. This range, which would require its own review if I were to investigate it in any depth, has a sufficient number of components and mounting possibilities and variations that I can't think of any conventional (or unconventional) instrument that I couldn't mount any 4099 or a 4060 microphone on — and the new, one-handed gooseneck G-Mount is a work of genius in my book!

Although it isn't technically a mount or a clip, I have to mention DPA's AIR1 acoustic fur windscreen, which will fit most miniature microphones. It's a little £42 work of art that will keep your audio usable in winds up to around 15mph.

One feature of the Core series that carries over into Core+ is that the microphones have been hermetically sealed and protected with a hydrophobic coating. These actions, together with dual gold-plating of the diaphragm, means that the Core/Core+ 4099 (and all Core/Core+ microphones) are now IP58 certified. This Ingress Protection rating means that Core/Core+ microphones are protected from dust and, more importantly in

»



■ A close-up of the 4060 Core+ capsule. Like all Core+ mics, the capsule has been double gold-plated, hermetically sealed and protected with a hydrophobic coating, giving the mics an IP58 rating.

» my view, from beer, water, sweat and TV, film and theatre make-up.

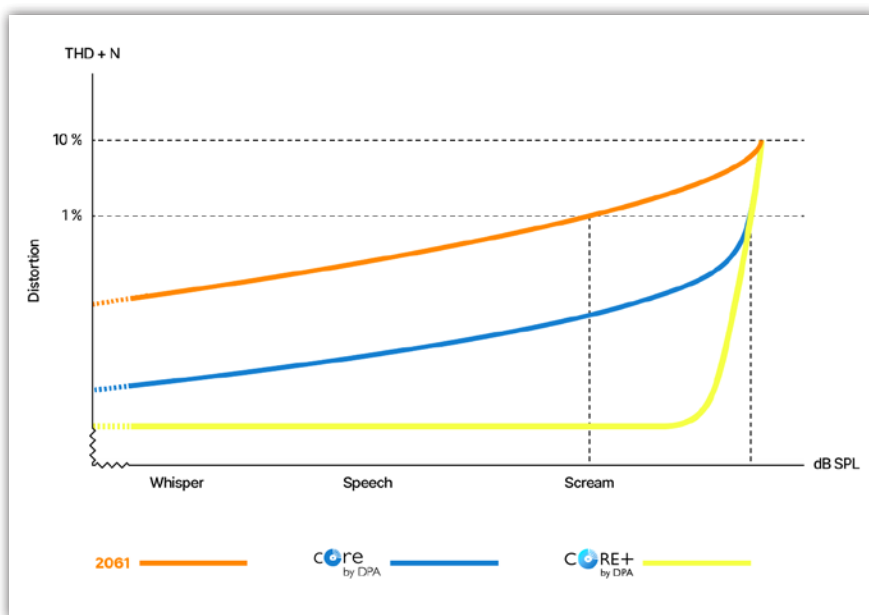
In order to use any 4060 or 4099 you'll need either the new DAD9099 balanced phantom powered preamplifier (included in the individual instrument kits along with their instrument-specific mounting clips), its earlier DAD4009 equivalent, or an adaptor (of which DPA have a wide selection) that allows connection to the transmitter of your chosen wireless system. Like the DAD4009 before it, the DAD9099 has an integral fixed 80Hz high-pass filter. Should you want to exploit the 4060 and 4099's effective frequency responses to their full extent, you'll need the new DAD9001 preamplifier, which doesn't have the 80Hz filter.

Core+ Audition

In order to explore the effect of DPA's Core+ technology, I decided to rig my original 4060 and its new Core+ counterpart side by side underneath the end of a violin fingerboard and pointed towards the bridge. A violin can generate surprisingly high sound pressure levels in the area of its soundboard in front of the bridge. A 4060 placed in this neighbourhood will always produce noticeable harmonic distortion if the violin is played at volume — which, incidentally, is why I habitually use the less sensitive 4061 as an unobtrusive close mic on violins.

Once I had the mics positioned together at a distance from the bridge where I could sense distortion from my 4060 when the violin was played, I switched between it and the 4060 Core+. To say that the difference was astonishing is an understatement, as there was absolutely no distortion in the signal from the 4060 Core+.

I then moved the mics off the violin and taped them to a boom stand so that I could easily change the SPL hitting the mics by varying the distance between them and the violin. What this revealed was that, to my



■ A DPA illustration highlighting the relative THD performance of DPA's 2061, Core and Core+ mics at different SPLs. Rather than increasing gradually, the Core+ distortion remains consistently low.

ears, the 4060 Core+ delivers a more open and natural sound across its dynamic range than my original 4060 is capable of doing at any point within its dynamic range. Since the two variants have identical max SPL values (134dB SPL for 10% THD) this result confirms to me that the Core+ technology really has achieved the goal delivering a performance that is as distortion-free close to its limits as it is at lower levels.

To check out the performance of the 4099 Core+, I mounted the supplied pair on my Feurich upright piano, where they delivered another open, natural and distortion-free performance across the widest dynamic range that I could achieve. In isolation, I did miss the weight of the lowest two octaves of the piano keyboard that were rolled off by the DAD9099 preamps' 80Hz high-pass filter, but I wouldn't have it any other way for live use on piano or any other instrument.

Last Words

In the higher reaches of their dynamic ranges, the review models — the 4060 Core+ Normal SPL and the 4099 Core+ Loud SPL — are such significant improvements over my original 4060 and 4099 that I'm going to have great difficulty in resisting the urge to purchase them both. Not only do the Core+ variants exhibit reduced distortion and increased dynamic range, but these also delivered an enhanced version of everything that I like about the original versions: accuracy, an effortlessly natural sense of reality, and

the sparkle from the soft boosts that adds a crispness to their character.

DPA's 4060 Core+ and 4099 Core+, in my view, set new performance standards for professional miniature microphones. The Normal, Loud and Extreme SPL, and presence-boosted variants of the 4060 Core+, and the Loud and Extreme SPL versions of the 4099 Core+, add significantly to the range of their potential applications. Add in the wide range of mounting options and I've little doubt that these microphones will be dominating my spending for the foreseeable future.

Although all DPA microphones are premium products with a commensurate price tag, I've always considered my two to be investments that offer real value for money over the long run — which is currently 23 years for my 4060 and 16 years for my 4099.

If you're searching for best-in-class miniature microphones for speech or instrumental use, then the DPA 4060 Core+ and 4099 Core+ (and their variants) should be at the top of your list of potential investments. That's where they sit on mine! ■■■

On Your Marks, Headset, Go

For years, theatre and film actors have had 4060s concealed about their person, but recently I've noticed a trend towards headset microphones. DPA included a 4466 Core+ omnidirectional headset mic with the review DPA shipment, which I also tried out. I sang a few songs and declaimed a few half-remembered Shakespeare sonnets, and concluded that I'd be more than happy to go on stage using a 4466 Core+ for my voice, although I'd also want to audition the 4488 Core+ cardioid variant.

£ 4060 Core+ Normal SPL with MicroLock £444. 4099 Core+ Loud SPL with MicroLock £450. 4060 & 4099 Instrument Kits (including DAD9099 preamps) £570 each. Prices include VAT.

T DPA Microphones UK
+44 (0)20 3008 7530

E info-uk@dpamicrophones.com
W www.dpamicrophones.com

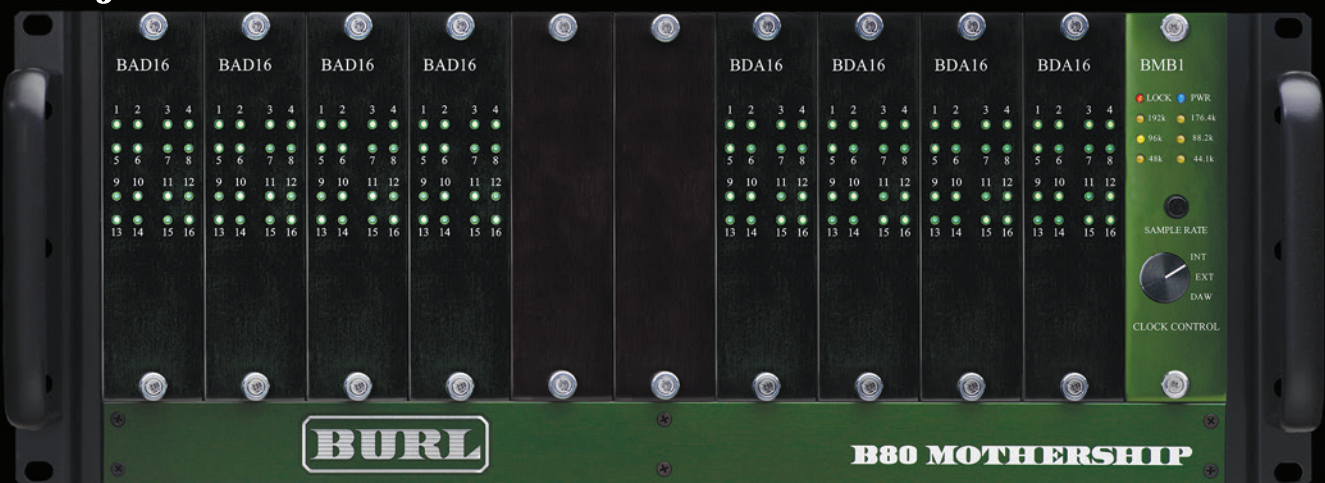
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It's coming, and you will hear it coming...

/* Vibe */ /* Coding */

- 1 How Anyone Can Create Plug-ins
- 2 Got an idea for a plug-in, but no idea how to code? Read on...

BEN MCCULLOCH

If you make music with computers then you surely use plug-ins. They're everywhere: sequencers, samplers, instruments, inserts, effects, utilities such as spectrograms and MIDI filters, and more. But have you ever thought about making your own plug-in? Perhaps you have an idea for a plug-in that doesn't yet exist, but you don't know how to make it yourself?

Until now, your chances of making your own VST, AU or AAX plug-in were limited unless you could write code, and had the time to invest in building your idea, or you could afford to pay someone else to do it, or you were famous enough to have companies competing to put your name on their plug-in. Today, there's another option: a new approach called vibe coding. Vibe coding is a process where you use natural language to describe what software you want to make, and have a large language model write the code for you. You don't even need to read, or understand, the code.

Large language models, often abbreviated as LLMs, are so-called 'artificial intelligences' that have been trained on a vast corpus of written material across all fields. This allows them to generate different types of material, from essays and forum posts to computer code. If you've already used LLMs such as ChatGPT, you'll find vibe coding is similar,

in that it feels like a conversation with someone, except you're talking about the software you want to make. All you need is the ability to put your ideas into words, and the diligence to see it through, even when things don't go to plan. These are skills you already use when writing songs. In this article I'll outline the process you'd need to follow make your own plug-ins, and I'll illustrate my points with an example project.

Starting Points

Before you start messing around with AI code generation, you need to know what you want to build. It all starts with one question: what do you want or need? Perhaps you want to work with MIDI data: filter it, generate melodies or invent outlandish new jazz chords. Or you could work directly with audio, and modulate sample playback so that you can have a snare backbeat that constantly evolves over the course of

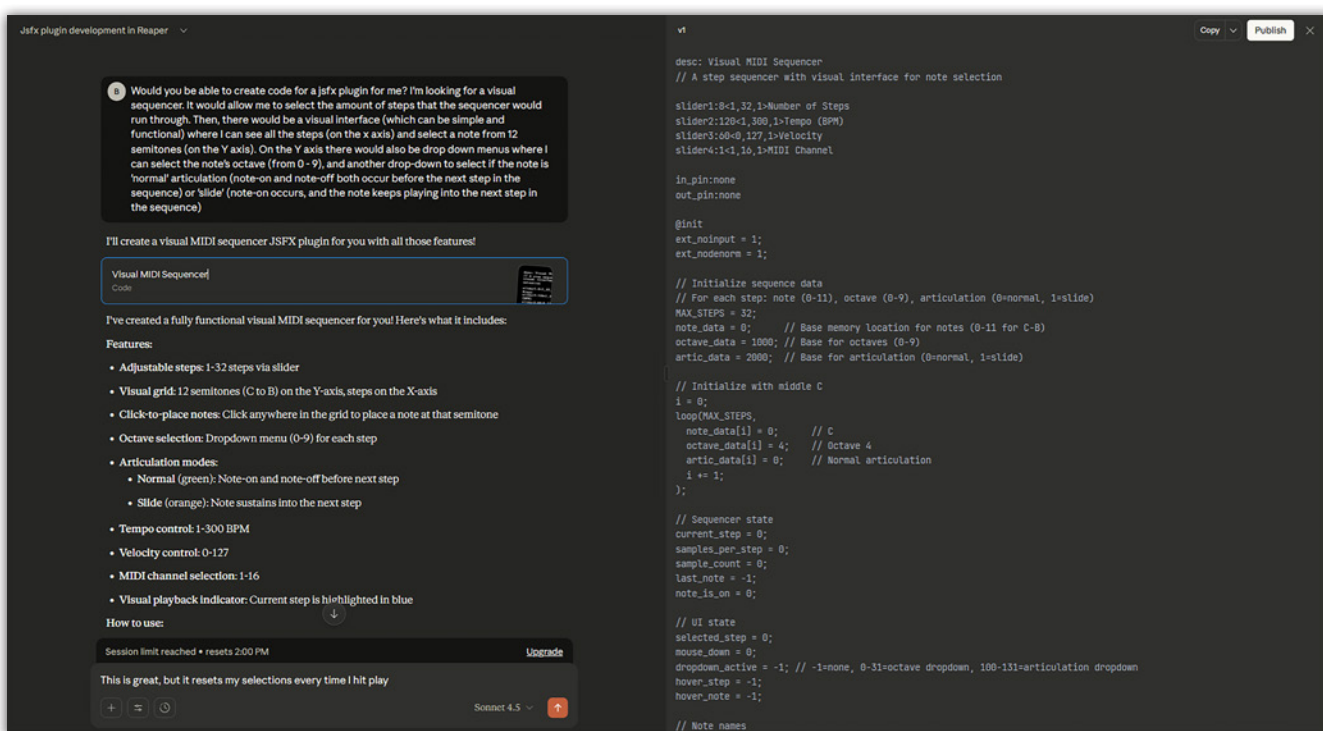
a song. These are possible, as are many other ideas. My only recommendation is that your first attempt is the simplest thing you can imagine. You want to learn the process before attempting to vibe code a complex plug-in.

You will need to rein in your expectations, too. Some things you try simply won't work. You can't expect to create a competitor to plug-ins that precisely model orchestras, or allow you to have granular control over every parameter of sample playback. Commercial plug-ins with those features are complex, and need experts to build them. It's best to think of vibe coding as a means to build something that's uniquely yours, that may not be available to buy. Don't underestimate it, though! I've frequently been surprised how far I could push my vibe-coded efforts. The results usually worked better than I expected.

It's a creative process, so I recommend that you approach it with an open mind.

B Would you be able to create code for a jsfx plugin for me? I'm looking for a visual sequencer. It would allow me to select the amount of steps that the sequencer would run through. Then, there would be a visual interface (which can be simple and functional) where I can see all the steps (on the x axis) and select a note from 12 semitones (on the Y axis). On the Y axis there would also be drop down menus where I can select the note's octave (from 0 - 9), and another drop-down to select if the note is 'normal' articulation (note-on and note-off both occur before the next step in the sequence) or 'slide' (note-on occurs, and the note keeps playing into the next step in the sequence)

Screen 1: The initial prompt used to create the first version of my sequencer plug-in code.



Your plug-in is going to be created during a conversation with a machine, where you give it your ideas and the machine sends new ideas back at you.

In this example, we'll use Cockos Reaper to host our generated plug-in, for two reasons: Reaper's JSFX plug-in format has its own compiler built in, and Cockos generously offer a month's free trial to new users. For the code generation we'll use Anthropic Claude, as they also offer a generous free tier and it's consistently given the best code-generation results in my trials compared to OpenAI's ChatGPT and Google Gemini. Claude's free tier

does limit the amount of code you can generate in any six-hour period, so you may have to wait for it to reset, or consider paying for the licence.

Prompt Action

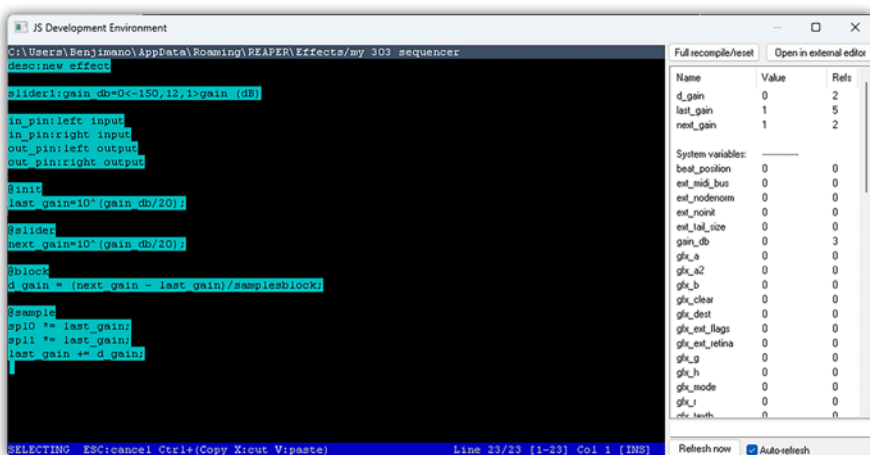
The vibe-coding process I'll outline isn't very risky. As with all software, bugs and crashing are an inherent risk, and there is the possibility that Reaper will fall over if you give it buggy code, but this has never happened to me. Another concern about LLMs is giving them too much personal data, but you've got control. The main thing to remember is to never tell the LLM anything that could identify you.

Worth noting is that Anthropic allow you to decide if they can use your conversations to train their models in the Privacy section of Claude's settings menu.

Let's walk through the process I follow to create my own plug-ins. While your goals may be different, the process you follow should be the same, and it always begins with considering what the final result should be:

1. Define your desired plug-in.
2. Describe it in natural language.
3. Use Claude to create code.
4. Test the code in Reaper.
5. Repeat steps 3 and 4 to optimise the features and fix bugs, if needed.
6. Stop when it works (rather than waiting until it's perfect).

About my own plug-in idea: I love acid house synths. Occasionally, I use sequencers to create repeatable melodies and rhythmic patterns. I want to vibe code a sequencer tool that's similar to a Roland TB-303's built-in pattern creator, but with some notable differences. I want it to offer more than 16 steps in a single pattern, and allow the user to select the exact number of steps; I want a visual interface that allows me to see all the notes in the sequence, not just the note assigned to the current step; I want the option to select any



» octave for each note, rather than just one octave above or below the default; and in place of the 303's 'accent' feature, I'd like to be able to choose from a wider scale of velocities.

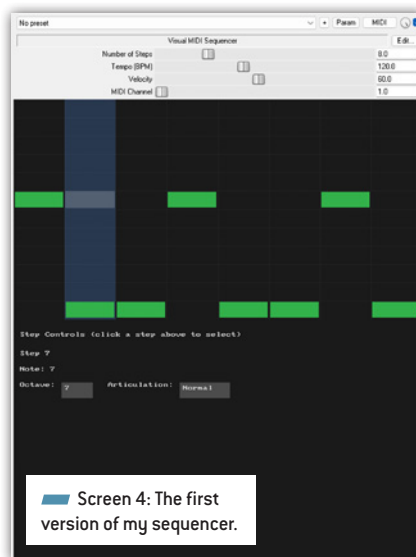
Before you begin, you need to have your software ready. First, register for Claude if you don't use it already. Once you're registered, it should open a new chat window. Keep that open. Then, start up Reaper, make a new project, add a track to it, and then click on the FX button on the track. The last two steps are to click on the FX menu text in the window that opens, then click on 'Create new JS FX...' from the drop-down menu. A new window will appear. Keep it open, and return to Claude.

Now we're ready to get vibey. I know what I want to build. Now I need to be able to put that into words. My description will be the 'prompt' that I type into Claude's chat window, giving the LLM the information it needs to build my plug-in. There's a bit of an art to creating effective prompts (a skill that is sometimes called 'prompt engineering'), and it gets easier the more often you work with LLMs. Many Internet forums are dedicated to it, but it's really just about giving instructions efficiently. Good instructions are specific and clear, include all the vital information, and often break big bits of information down into smaller chunks. Thankfully, LLMs are constantly getting better at understanding what we want, so don't worry about trying to make a 'perfect' prompt.

From The Top

My initial Claude prompt is shown in Screen 1. As you can see, I haven't given all the details at once. I want to see what Claude creates, and then shape it in rounds of improvements. It's a bit like writing a song: you try things, move stuff around, add to it and take things away, and over time, something good hopefully starts to emerge! As soon as I hit Enter to send the prompt, Claude starts to generate code straight away. The first time you see this happen, it's quite startling. Programmers take a long time to think through a coding problem before they begin. AI just dives in!

Incidentally, if you were to give Claude this exact same prompt, you'll probably get a different answer and a different set of code. That's because LLMs like Claude generate fresh answers each time they're asked a question.



“The code Claude generated gave me an idea I hadn't had before.”

The next step is to test the code that Claude has produced. In Claude, there will be a new separate window on the right side showing the generated code. At the top right of that window there is a Copy button. Click on Copy, keep Claude open in your browser, and then head to Reaper.

You should still have the New JS Effect window open in Reaper. You'll be asked to name your creation, so just give it a short descriptive name for now, such as My 303 Sequencer. Click on OK and you'll be presented with

a new and basic JSFX plug-in. Clicking Edit will open up the plug-in's code window, which will already contain some basic code. You want to select all of that and delete it, so that you start with a blank slate. Now you can paste in your Claude-generated code and save it.

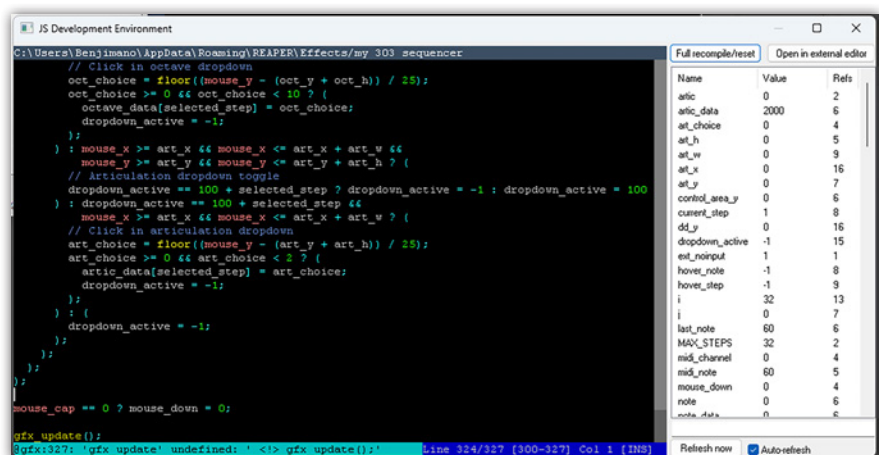
At this point, one of two things is going to happen. One is that you'll see a 'Recompiled OK!' message at the bottom of the code window, meaning that your plug-in is ready for testing. Alternatively, you'll see an error message. Both of these are normal occurrences and are positive steps forward! Sooner or later you'll hit a roadblock, but don't worry. They can be easily overcome.

If your plug-in is ready for testing, go ahead and try it. Just remember to try it 'in context', as if you were in a music-making session. For example, a sequencer won't make sound by itself, so to test that it's working properly, you would need to put another plug-in after it that generates sounds from MIDI data. So play around! Try your first version and see what happens. It might be the start of something incredible! Alternatively, perhaps your code didn't work; or, although the code compiles, the resulting plug-in doesn't behave as you'd hoped. In a moment I'll show you the process for overcoming these issues.

New Directions

The first version of my sequencer is shown in Screen 4. Considering I've only spent a few minutes thinking about what I wanted, and then the code was generated on the spot, that is pretty good! I didn't tell Claude how many steps I wanted, but it gave me a plug-in with a range of 1-32 steps, which suits me

»



Screen 5: This plug-in code has generated an error, shown in blue at the bottom of Reaper's JSFX window.

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» fine. The visual interface is very easy to follow, with steps running from left to right and musical notes from bottom to top — although I would benefit from knowing which musical notes they are. There are drop-down lists for octave and articulation (normal or slide). Claude has also added something I hadn't thought of: a MIDI channel selector. I hadn't asked for that, but that's useful, and makes me think an advanced version would allow me to set MIDI channel per step.

The main issue with the initial version was that as soon as I added the plug-in to a track, it just started running, independent of Reaper's transport controls. Strangely, it also let me set the tempo independently of Reaper's tempo, too. This was less a bug than a breakthrough; once again, the code Claude generated gave me an idea I hadn't had before. I love polyrhythms and strange time signatures. Hearing the sequencer run independently of Reaper's transport instantly made me think of having more control than the usual subdivisions such as quarter-note, eighth-note and 16th-note, plus triplet options. This made me wonder whether I could have the plug-in spread any number of steps evenly over a bar of music, such that if I set it to four steps, it would generate one step per crotchet, but if I set it to something odd like 13, it would evenly spread 13 steps across a bar. This is the kind of stuff that thrills me, with unexpected ideas and possibilities emerging in the creative process. I decided to try to get

B I'd like to adjust the maths used for setting timing on the notes. Here's my idea, and I need you to tell me first if it can be done - don't generate code yet. When I set the steps in this plugin, I want those steps to be 'per bar'. In practise this would mean that the amount of steps I select are spread evenly across the bar, so 4 steps in a 4/4 bar would be 1 step per beat. If I have 7 steps, then they would be spread evenly across the bar (resulting in some unusual polyrhythms when played over a 4/4 beat). Is this possible?

Yes, this is absolutely possible! Here's how it would work:

The Concept: Instead of having fixed 16th note timing, the sequencer would divide one bar into however many steps you've selected. So:

- 4 steps in 4/4 = quarter notes (1 step per beat)
- 8 steps in 4/4 = eighth notes
- 16 steps in 4/4 = sixteenth notes
- 7 steps in 4/4 = each step is 4/7 of a bar long (creating a polyrhythm)
- 3 steps in 4/4 = triplet feel across the whole bar

The Math:

```
samples_per_step = (samples_per_bar) / num_steps
```

Screen 6: You can have a conversation with Claude about what's possible, in much the same way as you might with a human developer.

Claude to make this new idea work, as an improvement to the code.

Incidentally, you might find that your plug-in's interface is ugly or bland. I recommend that you fine-tune the functionality before addressing this. This is because every change to the plug-in's functionality will have a knock-on effect on how it looks, so it's just more efficient to improve how it works before you tweak its appearance.

Power Up

What do you do if your code doesn't work at all, or doesn't work how you intended? You get Claude to fix it or improve it! If

there was an error when you tried to save the code, copy the exact message that describes it, for pasting into Claude. Screen 5 shows an error message I received when working on one plug-in. Annoyingly, Reaper doesn't allow you to select the text of the error message for copying. I work around this by taking a screenshot of the error message and uploading it to a free website that extracts text from images.

With the error message copied, next write a new message to Claude. Start by saying "I got an error -", and then paste the error message in. Claude will attempt to iron out the bugs in the code for you, and then give you an updated version.

Alternatively, you might want to make improvements to a plug-in that compiles and runs OK. If that's the case, you should describe what should be changed in clear and specific language, and let Claude improve the code. The best approach is to prioritise the most important changes first, and do them one at a time. Test each update to the code. You might end up playing 'whack-a-mole', where a new fix creates a new bug, but that's just part of the process. When you've got updated code, return to your plug-in's Edit window, clear all the code that's there, paste in the new version, and save it. Then test the new version.

This is basically the process you will follow from here onwards until you're satisfied the plug-in is done: test the latest version in Reaper, gather feedback, give Claude one piece of

Other Plug-in Formats

In this article, I've used Reaper's JSFX plug-in format, because it's the most straightforward to work with. It's possible to vibe code other types of plug-in, too, such as VST and Audio Units, but things will get a bit more complex. That's because you'll need to work with an extra development tool alongside Claude to create code, and your DAW to test the plug-ins.

There are various plug-in development tools out there. If you're already comfortable working with code and want to use vibe coding to develop plug-ins faster, then you probably want to work with JUCE (which has free and paid plans). The great thing about JUCE is that there are published code examples on the web, and many developers discussing it, so you can be sure the information is out there to guide Claude when it's generating your code! Rather than attempt to summarise the process you'd go through to learn JUCE, let's just say that you can expect to invest weeks or months to learn it. For most people who want to quickly vibe-code their own plug-ins, that perhaps defeats the purpose. On the other hand, if you're a Pro Tools user, JUCE is your only option for creating AAX plug-ins, and Avid have strict requirements and approval processes.

Thankfully, for everyone else there's a tool called Cabbage Audio, which is donationware that is available for free under GNU licence. Don't let the silly name put you off: Cabbage Audio is capable of building great plug-ins. You will need to explicitly tell Claude that you're building for that platform, so the code generated will be Csound rather than JSFX, and you would use their visual designer to create the plug-in's interface. You would need to export the plug-in from Cabbage Audio as a VST or AU for testing in your DAW. You can still expect to spend a few hours or days learning to work with Cabbage Audio (they have tutorials online), but the results should be usable by all DAWs that can accept VST and AU plug-ins.

feedback after another to improve the code. The great thing about working this way is that you can always go back a version if you add too many ideas and it gets unstable.

With my own plug-in, I was still extremely excited about having odd timings with rhythmic subdivisions of the bar. In Screen 6, you'll see a slightly different prompting approach. I didn't want to waste time trying to build something that won't work, so I asked Claude to tell me whether it was possible, without trying to generate any code yet. As you can see, its explanation contains a mistake — the division of seven steps in 4/4 should yield 1/7 of a bar per step — but when I read Claude's full response, I felt confident it had grasped the idea. It's a conversation, and sometimes you can ask for an opinion rather than code.

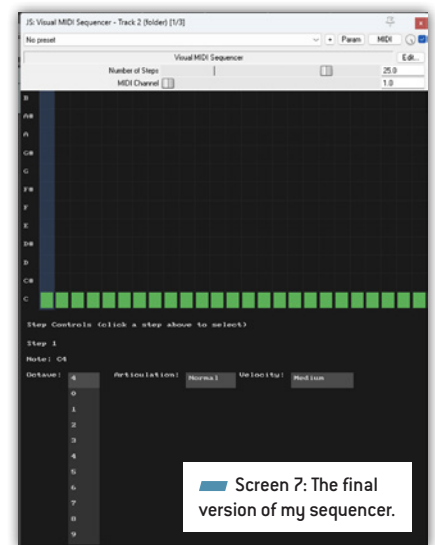
Working fix by fix, I also asked Claude to add the note names up the Y axis, follow Reaper's main transport for Play and Stop, allow 'silent' steps, and add velocity control per step. The last of these didn't work, even after Claude

tried many times, and another issue is that my settings reset every time I hit Play. I'm confident these can be fixed, but Claude hasn't found the fix yet. This means I have to hit Record and add my notes and settings afterwards. It feels like jamming, but of course I'd prefer to be able to save my work!

Good Enough

Screen 7 shows the 'final' version of my sequencer plug-in. I decided to stop tweaking it, because it was good enough for my use. The subdivisions work perfectly, so I want to make music with that feature straight away, and I can always find other ways to alter the velocity, such as recording MIDI and then humanising it, or using automation.

As with anything to do with creativity, 'final' is a relative term! It's often very hard to stop tweaking when you're vibe coding a plug-in. If you find you've invested too much time, my recommendation is to re-read your initial description of what you wanted. How close are the results to what you



wanted? If they're way off, perhaps the idea was too ambitious. If they're close enough to be usable, I recommend you just start using it rather than making any more tweaks. It's best to finish things, and that frees you up to make your next great plug-in! ///

SOUND ON SOUND

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Erica Synths x Hexinverter HexDrums

Drum Machine

RORY DOW

HexDrums is a collaboration between Hexinverter Électronique, a Canadian Eurorack modular company that closed its doors in 2022 due to post-pandemic supply chain issues, and Erica Synths, who subsequently acquired the rights and continued manufacturing, selling, and supporting some of Hexinverter's most popular modules.

Amongst those modules were the Mutant Drum series, a collection of xOx-inspired drum voices that have become the core of this analogue drum machine. There are 10 drum voices comprising two bass drums, a flexible percussion engine called Machine, snare, clap, rimshot, hi-hat (open and closed), and two sample-based cymbals. The most obvious equivalence is with the mighty TR-909. The HexDrums wears the influence proudly, but is not shy about deviating from established norms and treading its own path.

Hex Appeal

The HexDrums is a chunky desktop unit that comes in a lively black-and-yellow livery. It offers real-time control over every

Erica Synths have resurrected the now-legendary Hexinverter percussion modules to create a drum machine that's as immediate and playable as it is individual.

drum parameter, individual outputs for all voices, a hands-on 64-step sequencer with pattern memory, a song mode, a stereo master compressor and an overdrive circuit. Operation is intuitive, and if you've ever used a classic analogue drum machine, you'll barely need to open the manual.

At 455 x 216 x 91mm, HexDrums is a solid desktop unit. Its all-metal casing, black wooden side panels, mechanical cherry keys, and panel-mounted pots and switches make it feel robustly old-school. All the inputs and outputs on the rear are on proper grown-up quarter-inch jacks and 5-pin MIDI sockets. In short, the build quality is superb.

The Drums

The first of HexDrums' two bass drum voices is ideal for the heavy kick drum commonly used in techno, electro and many other electronic music genres. It's a combination of a triangle core VCO (square or sine wave), a sub-oscillator, a pitch envelope with variable decay, a pitch control with a range of several octaves and an adjustable click level. The results are very flexible, ranging from soft, short clicky

kick drums to long disco toms, typical 909 kicks, and massive gabber techno window-shakers.

The second bass drum is based on the TR-808 twin-T damped sine-wave oscillator. Unlike the TR-808, however, you can tune the fundamental frequency to match your song, or high enough to create tom sounds. It has just three controls (excluding level): pitch, decay and tone.

The Machine voice is the most complex of all, capable of a wide variety of percussion, synth voices and special effects. It consists of two synthesis parts, a dual-oscillator membrane and a noise section. With a variety of envelopes to provide amplitude and pitch variation (one of which can be permanently opened for drone sounds), you can achieve semi-realistic snares, dark techno rumbles, bass sounds, zaps, blips and noise blasts.

The dedicated snare drum voice uses two bridged-T sine-wave oscillators, adding a noise element, two octaves of tonal pitch control, and a resonant filter to create a surprisingly versatile snare. It will emulate 808 snares with ease, and 909 snares up to

a point (I couldn't quite get the 'snap' right).

The clap uses a linear feedback shift register to create the noise elements, essentially producing white noise that, when pitched down, becomes more digital-sounding (like 8-bit noise). There's a simulated reverb that doesn't sound like a real reverb but adds some low-frequency body, and a band-pass filter. Decay and filtering are also available for general tone shaping.

The Rimshot is next, and can be switched to sound like a Clave. There are the usual pitch and decay controls, plus a state-variable filter. With the wide pitch range and filter, the rimshot can be coaxed to produce kick drums and toms, and the filter, with high resonance and mod envelope, can tease out electronic zaps and special effects.

The Hi-hats circuit bundles both open and closed hats into a single 808-inspired section. The Decay controls the open hi-hat length. There's a band-pass filter for further shaping, and you can de-latch the hats so that a closed hi-hat will not choke the open hi-hat.

Finally, we come to the cymbal section, which is another dual-voice group. The Crash and Ride voices differ from all the others in that they use samples. Each cymbal has its own pitch and decay parameters. The pitch controls adjust the sample rate to alter the pitch, resulting in some wonderfully crunchy, lo-fi aliasing at lower pitches. The Ride also has a loop switch that repeats the sample for the duration of the decay, creating the illusion of a longer sample, or even a fake delay effect.

Both the Crash and Delay can be chosen from one of 10 sample sets, and one rather cool feature is that you can replace the Crash and Ride samples by

uploading your own 48kHz, 16-bit samples to Hexdrum's internal flash memory. You do this by entering USB mode, which makes the internal disk mountable on your PC or Mac. You can replace the factory samples with literally anything: vocal one-shots, your favourite kicks or snares, or, dare I suggest, some different crash and ride cymbals. Neat! But it is marred by the single shared selector control that changes both samples at once. You cannot, for example, select Ride sample 3 and Crash sample 7. You can choose only 3 and 3, or 7 and 7. It certainly won't replace your sampler, but it's still fun to import your own sounds into the system.

The Master Section

The master section includes two effects that can be applied to the master bus. Sounds coming from the individual outputs are not affected (in fact, if you have cables connected to the individual outputs, those voices are removed from the master outputs).

First up is the compressor: a classic soft-knee design with 40dB of gain, threshold, ratio and dry/wet mix controls. The exact ratios and threshold are unknown and set by ear. You can also switch response time between three options: slow, mid and fast. A single LED serves as a rudimentary gain-reduction meter.

Round The Back

In addition to the 12V DC external power supply socket and switch, HexDrums' rear panel houses the master L&R outputs, stereo headphone output, 10 individual drum voice outputs, MIDI In and Out on 5-pin DIN, USB-B for MIDI and firmware updates, and analogue clock input, output and reset. All audio inputs and outputs are on quarter-inch jacks.

Erica Synths x Hexinverter HexDrums

£1299

PROS

- Classic drum-machine layout and user experience.
- A robust sound that will fit right in with classic techno.
- Familiar sequencing, with enough new features to keep things interesting.
- Chunk, solid build quality.

CONS

- Hi-hat filtering can make hi-hats disappear.
- Cymbal samples have to be chosen in pairs.

SUMMARY

HexDrums feels like a love letter to the Roland TR-909. It's a 10-voice analogue drum machine, with a couple of sample-based cymbals that can be replaced with your own samples. All the usual 909-style sequencing tricks are there, but with enough modernisation to keep things feeling new and carve its own sonic identity.

In addition to the compressor is a drive circuit that adds some distortion. There's little to it, just a drive control and a blend (dry/wet) control. Overall, these two master effects do a great job of adding some weight and extra harmonics to the mixed output, and they feel well-tuned to the source material. These are master effects that you will actually use.

The Sequencer

Once again, the Roland influence is undeniable in the sequencer section. The row of mechanical cherry keys at the base of the unit comprises the transport and sequencer programming. Each sequence can be up to 64 steps in length, and each

»





■ The HexDrums measures 455 x 78 x 215mm and weighs a reassuring 2.65kg.

» voice can have a different number of steps. Perfect for polymeters.

Notes can be programmed, in typical x0x style, or tapped in and recorded live. Cherry keys aren't ideal for finger drumming, but the option is there if you need it.

With a total of 16 banks of 16 patterns, there should be enough storage space for anyone, and you can backup and restore patterns over USB if needed. You can copy patterns between banks and slots, making it a breeze to program pattern variations.

As well as having different lengths per voice, the sequencer also provides playback direction (forward, back, two ping-pong options, and random), shuffle, and clock divisions, which they call 'Scale' (1/8, 1/4, 1/2, 3/4, 1, 3/2, 2 and 4). These options really help craft patterns that escape the typical one-, two-, or four-bar loops that x0x-style drum machines can get trapped in.

There are some nifty per-step features, too. Like the TR-909, the HexDrums offers both global and per-voice accents that can add dynamics to a pattern. Accents are visualised using the LEDs embedded in each key. Maybe it's my eyes getting old, but I found it difficult to tell the difference between the full brightness used to show an accented step and the slightly dimmer brightness used to denote no accent.

Each step also supports ratchets, with up to eight repeats per step, probability (a percentage chance of the step triggering), odds (play the step every X repeats), and microtiming, which allows you to move the step off-grid.

Building songs and performances is also well thought out. Switching between

patterns on the fly is helped by the 'master length' function, which dictates when patterns should switch. For example, if this is set to 16 (steps), a pattern will always play to the end of a bar, before switching to the next pattern. Also, if you reprogram a pattern during a performance, you can hit the Reload button to return to the stored pattern. The flip side is that when programming, you need to remember to save your changes before moving to another pattern, or you'll lose them. Mutes and solos can also be employed during a performance, and a Roll function allows for some 'note repeat'-style drum rolls.

Song mode can chain up to 16 patterns from a single bank in any order and repeat them up to nine times if required (an odd choice, but I guess that's the highest number they could display on the LCD screen). Songs can loop, or come to a neat stop at the end of the pattern list. It's a shame that you can only make a song from patterns in a single bank.

In Use

If you are a fan of the Roland TR-909, you will likely adore the HexDrums. It has a big sound, which lends itself to classic techno, electro and house music. I expect there will be much online opinion about individual voices, and whether you happen to like the kick drums (I do) or the clap (I don't), will be down to personal preference.

What I love about the HexDrums is that it feels very immediate and responsive. The sequencer's timing is particularly pleasing, even when clocked by an external source

and getting the most out of it involves being hands-on: tweaking sounds as a pattern plays and reprogramming on the fly. Jeff Mills (seek out his legendary 909 performances) would do amazing things with the HexDrums.

There are a few minor design decisions that I found perplexing. Why, for example, do the hi-hats use a band-pass filter that almost entirely filters out the hi-hats when the Cutoff knob is set below 50 percent? Why do the Crash and Ride samples have to be selected together in pairs? Why do some Shift key functions not have a label printed on the front panel? (Shift+Last is the 'master length' parameter, but has no label to remind you.)

Some people may find the lack of parameter locking or automation recording disappointing. I debated not mentioning it because it is a feature outside the instrument's design scope. But people will ask if it's possible, and now they have their answer.

The HexDrums is a techno monster. It covers 'classic' territory with panache, and adds just enough modernisations to keep the 40-year-old drum machine design feeling fresh. It feels immediate and responsive. The timing is tight and the sound is raw and in-your-face. It invites performance, and sounds great whilst doing it. The HexInverter legacy is in safe hands with Erica Synths, who have proved that they know how to make a great drum machine. ■■■

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Win!

Expressive E Osmose 61

Worth €1999



First announced in 2019, Expressive E's original Osmose synthesizer and controller married the familiarity of a keyboard with the expressive possibilities of MIDI Polyphonic Expression (MPE). It took a while to come to market, but when we reviewed it in *SOS* April 2023 we hailed it as a "unique, surprisingly versatile and hugely inspiring" instrument, which both sounded superb on its own and worked as an excellent controller for MPE-compatible software and hardware instruments too.

With its 49 keys, capable of outputting tap, pressure, pitch-bend, vibrato, glide, note-off and aftertouch data on a per-note basis, the Osmose might well have been

synth, USB and 5-pin MIDI output, traditional pitch and mod wheels and colour screen interface — but adds a whole extra octave, for unrivalled MPE playability.

For this month's exclusive *SOS* competition, we're giving you the chance to win an Osmose 61 of your very own. To be in with

a chance of winning, simply follow the URL shown, and answer all the questions there, by Monday 2nd February 2026. Good luck! **///**

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To enter, please visit:
<https://sosm.ag/osmose-61-comp>

the ultimate MPE instrument — except that just as we were going to press this month, Expressive E announced its bigger brother. The Osmose 61 has all the same features as the original, including the powerful onboard 24-voice EaganMatrix

TalkBack

Galina Juritz

WILLIAM STOKES

Since graduating from the University of Cape Town with a degree in Violin Performance, British-South African artist and producer Galina Juritz has worked with groups like ShhArt Ensemble and Inclementine and composed numerous film and animation soundtracks. *One Weird Trick* is her first studio album, mingling jazz, electronic, ambient and orchestral sounds.

At the moment I can't stop listening to

Alas, like many other parents of toddlers, the *Go Jetters* soundtrack on CBeebies is the thing I hear day in and day out. It's got an uptempo disco intro that, to be fair, is a pretty vibey way to start the day at 7am! But some standout discoveries of the past while have been Alvarez's *La Línea Imaginaria*, Growth Eternal's *Bass Tone Paintings* and Reinier Baas' *Reinier Baas vs. Princess Discombobulatrix*.

Being part of the Kit Records family, I am blessed to have a world of weird and wonderful music in constant circulation — and many artists on the roster, and from elsewhere, passing through London have spent nights on my couch. At some point next year Cara Stacey and myself will be resuming our radio show *Run Amok*, where we dive into the vaults of music we've had on repeat.

The artist I'd most like to collaborate with

Again, so many! But maybe I'd form a trio with Jon Bap and the soundscape sorceress and composer Hildur Guðnadóttir, who created the *Chernobyl* soundtrack — two living legends of our time. I don't really know what they'd do with me, probably sample my whimpering murmurs of adulation and reharmonise them. Jon Bap is also partial to scruffy DIY aesthetics, so maybe the two of them could rehearse and send me out into the streets with an iPhone in the hope that I might record something interesting for an interlude! Hildur may allow me to scrape

a few stringy notes out, with the caveat they were recorded outside an industrial concrete carpark-turned-nightclub, with the groans of inebriated attendees covering a nice muddy frequency range. And if Louis Cole could pop into the room every now and then he'd lighten the mood and make sure the humour in the music was always detectable.

The first thing I look for in a studio

I look for a warm and inviting ambience. A space people would want to spend many hours in, like a kind of magical hideout den, away from regular life. You want that ease of workflow, so you can be spontaneous and capture new ideas quickly. I also always enjoy identifying unique pieces of gear in a person's studio — maybe a warm vintage compressor or an uncommon instrument lying around — and finding out what it means to them and why. People reveal so much about their musical identities through their relationships with these inanimate objects. For me, every piece of gear I own has a story, like the very first extremely cheap and nasty guitar multi-effects pedal I ever put my violin through that used to shriek with feedback, or the mini Fender bass I got from my partner for my birthday after recently moving back to London... or my TUL F47 microphone, dented in every direction from moving countries multiple times.

The person I would consider my mentor

From my classical violin training to my progression into composing, improvising and more electronic music, I've had a host of mentors along the way. During my studies in Cape Town I had a formidable violin teacher called Farida Bachorova, who had previously been the youngest female concert master in Russia. She has extreme attention to detail and doesn't suffer fools! Then, my lovely uncle David Juritz is a wonderful violinist who I have been fortunate enough to see perform over the years. The South African composer, playwright and installation artist Neo Muyanga really modelled how to be

brave and crazy, and how to put people from different disciplines on the same stage. And starting my journey as a producer, it was invaluable to be able to sit in a room and just watch producers work, asking questions along the way. The Capetonian group Mix n Blend and Zain Wolf were very kind to give me their time. Being part of egalitarian groups like the ShhArt Ensemble and Inclementine gave my friends and me a place to try out each other's compositions and workshop ideas. Matthijs van Dijk always had great advice on my scores and Cara Stacey would always gently encourage me out of my comfort zone — and also wipe away my pre-concert tears, of which there were many.

My go-to reference track or album

Thundercat's *Apocalypse* album has a sound that, for obvious reasons, feels centred around the low end of the spectrum, with lighter high-end vocals and strings washed out in reverb. I still don't know if my penchant for reverb stems from a desire to hide in a coat of shimmering obscurity, or if it's just an aesthetic preference; how much I use may change depending on what side of the bed I wake up on! It feels raw and not overly expensively produced, and emotionally more vulnerable for it. Then, for electronic music, there is no day of the year I wouldn't be soothed by the sound of Burial's 'Archangel'. The palette... those restless, unsettled drums, melancholy strings, chopped-up fragments of frantic vocals. It's right up my street.

My secret weapon in the studio is

When recording for other people, I'd say it's some kind of masochism-stamina-neurosis hybrid which allows me to record a thousand violin takes and add multiple parts! When I've started a string section I like to build upwards and just keep going. It gets addictive. In terms of my own music, I feel something I've been able to hone in sessions is the ability to embrace the



chaos and texture of spontaneous DIY improvisations, and then frame the results in a more solid harmonic structure. Also the Valhalla Supermassive reverb, iZotope Stutter Edit and also dramatic panning! Though the latter is probably an art school hangover from Goldsmiths University.

The studio session I wish I'd witnessed

The recording of an album called *Country Cooking* from 1988, by Chris McGregor's Brotherhood Of Breath. It's a brass-heavy big band and very close to my heart as my dad [Robert Juritz] often played saxophone and bassoon with them during my early childhood. The Brotherhood were composed of various exiled South African and British musicians. You can't overlook the band's political significance as an interracial band in the height of apartheid South Africa. With roots in Xhosa choral music, free jazz and the staggered cycles of Southern African vocal polyphony, this music spoke to

everything that the government was actively suppressing. There was a beautifully anarchic spirit to the band, the way they lived and the way they played.

I have memories of being a young child at the old water mill outside the French town of Agen, where before touring, the band used to have long rehearsals, and longer braais! In my mind it was a creative countryside wonderland. *Country Cooking* was born out of this wild time and remains one of my favourite albums to this day.

The producer I'd most like to work with

I'm going to say Tony Visconti, mainly because of David Bowie's *Blackstar*. The reason being that he must have a remarkable emotional competence to have been entrusted with producing someone's music while they were in the process of dying. To be able to hold the gravity of that, then incorporate all of the new influences Bowie was bringing in

like Kendrick Lamar, or Death Grips, and a more jazz-bound sound than he had ever explored in previous albums, while keeping everyone in the presence of those things simultaneously... I think it's an extraordinary feat of humanity and musicianship, and watching him speak back on the experience you can feel the love he has for them all.

In an alternate life I also feel like being produced by Steely Dan would be a baptism of fire, as there is no way they would let me get away with delivering anything shit. And I'd have to endure a lot of disdain and glaring through the studio glass! And Arca would probably be wonderful to work with. A deep educational dive into sound-world producing that I could get totally lost in.

The studio experience that taught me the most

When I was in high school we had to do a work experience week, and without much forethought I picked Cape Audio College and Milestone Studios, as they sounded grown-up and cool. Walking into this space for the first time I remember the feeling of freedom I experienced. No one was in a suit and tie, no one was being buzzed orders at or hunched next to a stack of dull papers — probably my idea of what 'real jobs' entailed. It was the first time I felt this radioactive potential, in the padded walls of a studio. Up to this point my first-hand experience of music had been my school training, playing in orchestras and ensembles, which were quite hierarchical and structured. It opened my eyes to a world which I knew I'd want to find my way towards one day.

The advice I'd give myself of 10 years ago

Stop operating from the outside in and be brutally honest about what you find interesting and what genuinely moves you. You could spend years trying to do things that have already been done better by other people and never achieve them, so you may as well start doing your own thing. There is a careful balance to be struck between experimenting, venturing, and then inviting critique, growth and refinement. You will make many mistakes and they won't kill you. In fact, they are a necessary facet of your own development, testing the parameters of where you are now and where you are going. And don't 'accidentally' lose a lot of your live recordings because you felt cringe-y about listening back to them! You will wish you had them down the line, warts and all! And of course, most importantly, don't drench absolutely everything in reverb. **///**



Arturia KeyStep MkII

Controller Keyboard & Sequencer

Arturia have reinvented the much-loved KeyStep controller keyboard.

ROBIN VINCENT

The KeyStep and its derivatives have become almost ubiquitous in the synth community. You see them everywhere at synthesizer shows and on YouTube videos and they're clearly a perfect fit for anyone working with MIDI, software and hardware. And so, with a product so useful and successfully baked into our workflows, it takes quite a bit of gumption to attempt to improve it. This will be a story of losses and gains, and whether that's a happy story depends on what you see as important in a multifunctional two-and-a-half octave compact MIDI controller.

New Hardware

My first impressions are that it's the same form factor of the original but with the knobs removed. It's fairly light, made of relatively robust plastic, and shallow in depth. The plastic knobs and squishy buttons have been replaced with a row of hard plastic but nicely clickable buttons, a tiny OLED screen and a silver hi-fi knob that stands a bit too proud. This feels like a regressive step back into screens and

menus, which causes me to wince a little, but we will see what the interface offers before passing judgment. More pleasing are the little graphical clues printed along the top of the keys that hint at patterns and note structures.

At the back, we have a USB-C port for power and connection next to a very welcome on/off switch, although we have lost the option for DC power. We then have full-size MIDI in and out ports, 3.5mm sync in and out, a quarter-inch sustain port and a bunch of 3.5mm CV outputs. We get an additional CV mod output to add to the original pitch, gate and mod jacks. Something I've always found curious about the KeyStep range is that Arturia talk about the compact depth enabling you to be closer to the thing you're controlling, except you have all these ports getting in the way as their cables spew out the back. I would have liked to have seen these ports running along the front panel, or pushed to the sides to enable you to butt the KeyStep up against its hardware destination. I would also like to have seen the labels of the ports printed on the front panel, as they do with the KeyStep Pro, to save me writing them on with a Sharpie so that I don't have to turn the keyboard on its edge every time I want to patch it in.

One thing I am very grateful for is the removal of the micro-switch sync options

panel. I imagine we have the screen to thank for that. In fact, there's quite a lot we can thank the screen for. I no longer have to connect to the MIDI Control Center in order to change any settings; it can all be done on the device itself. As we'll see, the screen is really the key to all the depth that's been brought to the MkII.

Arpeggiator

So, other than the interface, what's new? Arturia have pulled the sequencer and arpeggiator into an interesting new relationship, added new arp patterns, extended the chord function and are introducing the Mutate button, which will

Arturia KeyStep MkII

£109

PROS

- Mutate button.
- Sequencer to arpeggiator and back again.
- New arpeggiator patterns.
- Deeper Chord features.
- Spice and Gate strips.

CONS

- Screen driven.
- No knobs.
- Inconsistent pattern naming.

SUMMARY

The KeyStep MkII trades in its knob-led immediacy for button-led depth and expanded creativity.

reinvent your sequence. The keybed, on the other hand, remains the same half-decent experience with slim keys, velocity and aftertouch.

Let's start with the arpeggiator. On the original you could switch between all the usual arp modes on the turn of a knob. They were good and solid, and you could see at a glance which one you were on. With the KeyStep 37, they introduced a pattern mode as option 7, which is by far my favourite arp mode on any device I've ever encountered. In pattern mode, it just seems to generate a sequence of notes that sounds great. Whenever I want to try out some sounds or demo a synth I throw on arp mode 7, hit some keys, and it's perfect. Arturia noticed how great it was, pulled it across to the KeyStep MkII, and expanded it into nine fascinating variations.

As there's no knob, you select a mode by holding Shift and choose it from the 16 options spread across the keys. The little graphical clues are actually very helpful, and once you've tried them all out you know exactly what they are telling you. The first five are familiar ARP modes, number 6 is a chord mode, 7-15 are pattern modes, and 16 is a user mode. The screen and labelling are inconsistent here, where the modes 7-15 as written on the keys are called Patterns 1-9 on the screen. It makes you struggle to see what mode you're on and leaves you missing the original pattern knob all the more. In Pattern mode on the KeyStep 37, each time you press some keys you get a new, random and usually awesome pattern. On the KeyStep MkII, you now have nine patterns that follow their own distinct rhythm and character using ties and rests, but you get the same pattern every time. This would be disappointing if it weren't for the new expectation-beating Mutate button.

Hit the Mutate button and the pattern reforms into something funkier while staying within the rough structure of the selected pattern. Keep hitting it until you find something you really like. So,

you now have nine rhythmic variations of the KeyStep 37 arp number 7 that can be constantly evolved. When you find something you like, you can save it or manipulate it by squirting it over to the sequencer.

Sequencer & Chords

The sequencer takes the KeyStep beyond the enhancements found in the KeyStep 37 and into some solidly useful territory. The stealing of arpeggiations is delightfully fruitful because in sequencer mode I can transpose it with a finger, overdub on extra notes or edit the sequence note by note on the little screen. Starting from scratch, you can pump in notes SH-101-style or set it looping and throw in some keys. The screen tracks the loop progression, giving you visual clues of tempo and timing. At the top of the keyboard is a very useful Undo button that will remove everything you did from the last time you pressed Record. You can nudge notes, change octaves, double the pattern length and quantise, all from the Shift key. It's fast, fluid and entertaining. And in a confluent twist, if you save your sequence, which you can do in up to four banks of 16 slots, the rhythm and phrasing of that sequence will turn up as arp user mode 16.

But there's more. You can record automation from the mod and pitch-bend strips directly into your sequence. There's no editing for it, but you do have that very handy Undo button and the ability to clear automation via a menu. Hidden beneath the touch strips are two secret weapons called Gate and Spice, which can add a lot of flavour to your sequences and arpeggiations. Gate changes the gate size of the notes, which is far more satisfying than selecting a single gate size percentage like you could on the original. Spice is a weird little function that appears to throw in gate changes, ratchets and general messing about depending on how hard you lean into it. Both of these can also be recorded as automation into the sequencer.

It can get quite busy and complex once you start building up your sequence. This is where those save slots come in handy so that none of that hard work is wasted. You can also chain patterns together and have your KeyStep play back a whole song. But if you are struggling to develop beyond a basic melody line then the Mutate button is ready and able to give you new ideas. It's interesting how the Mutate function differs between arp and sequencer. With the arp it throws out a new idea in a similar rhythm, whereas with the sequencer it feels very much more derived from the notes and pattern you entered.

The Chord mode has acquired all sorts of depths with the help of the screen, similar to those in the KeyLab MkII. Along with holding the Chord button and recording the chord you are currently playing, you can also browse a bunch of chord presets, spread the voicing, add bass notes and turn it into a strum. The scales are also very welcome and you can set different ones for the keyboard and the pattern generation.

Conclusion

The new MkII makes some quite radical changes to the flow and intention of the KeyStep. The original was playful, very immediate, and great for playing your gear. The new one builds in the ability to craft and then save, mutate and save again, sequence, re-sequence, spice up and save again. It becomes a place where you can write melody and build songs. Some of the immediacy is certainly lost. The three knobs of the original made it the easiest and most convenient player of modular and MIDI boxes on the market, whereas the MkII requires you to dig a little deeper. For shows and demos it adds a level of unhelpful complexity, but it has huge potential for the home user and creative spaces where a little bit more effort will be rewarded. **///**

£ £109 including VAT.
W www.arturia.com



■ The back panel loses its DC power socket, but gains an on/off switch. The full-size MIDI ports remain mercifully intact, and the CV section has gained an extra mod output.

Future Sound Systems Cric

Analogue Synthesizer

Future Sound Systems' debut desktop is very much in a class of its own.

WILLIAM STOKES

Just when I thought Bristol's Future Sound Systems were getting 'serious' with the Cric, they put a cartoon frog in its manual with a speech bubble saying, "Hi there! My name's Francis and I'll be popping up here and there to guide you towards certain things to look out for when using Cric."

Francis Crick was, of course, one of the great British scientists, and such in-jokes would be incredibly annoying if they didn't somehow align with Future Sound Systems' design ethos perfectly. From their filters to their oscillators to their modular guitar preamps, the company's philosophy combines solid pragmatism, labcoat-worthy experimentalism and a certain

light-hearted — surreal, even — creativity. I say all of this because, in many ways, the Cric is the ultimate expression of this philosophy, which is the key to understanding it.

An all-analogue, patchable desktop synth, the Cric is quite a beast. At least, it's more beastly than anything Future Sound Systems have produced to date, but make no mistake, this is a premium synthesizer: patiently designed, impeccably built and, to be honest, with little precedent in its field. Aside from a few diminutive standalone offerings, the company are of course best known for their Eurorack acumen, from which I'm happy to report the Cric draws more than a little. Despite this, however, it's telling that at this year's Machina Bristonica exhibition, the Future Sound Systems crew didn't bother bringing any modular

goods with them at all, instead focusing solely on showcasing the Cric.

Complex Oscillations

That modular mindset is certainly still traceable here, however, in that the Cric's circuits can be patched any which way. Three oscillators and a white noise source constitute its sound generators, but it doesn't take long to realise that these coalesce to create something far more than the sum of their parts. The first is the Scissor oscillator, which is the most traditional of the three, though in fact it's also not very traditional at all. It has four wave shapes — sine, triangle, sawtooth and pulse — each with its own attenuator and output. So, right from the off, with these all sounding together in different combinations, there's already decent scope for sculpting composite



sounds with different waveforms. The Scissor oscillator doesn't stop there, though: beyond its knob for variable FM, switchable between linear or exponential, it has a shape control which of course varies the pulse width of the pulse wave, but also shifts the harmonic content of the sine and triangle waves. This functionality is dependent on a switch which, when on Clean, maintains a 'purer' version of those waveforms, and when on Shape imbues those triangle and sine waves with a type of notch at their peaks, resulting in a grittier texture reminiscent of older, characterful analogue synths, as well as FSS's own Cyclical Engine module.

Next are a pair of 'DNA' oscillators. These are curious things, delineated as +DNA and -DNA. The +DNA oscillator only outputs positively-rectified sine, triangle or saw waveforms (that is to say, with the negative-going half of the wave form flipped to positive-going) while -DNA outputs the inverse. These only have

one output apiece of their respective DNA Mixes, but the VCA levels of those constituent discrete waveforms can be individually attenuated and modulated thanks to individual pin inputs.

Pin Drop

This, tangentially, brings us to a conspicuous point of departure from the cabled world of modular: the Cric's pin matrix. The Cric is one of only a handful of synths of its size to sport one, but you'll be hard pressed to find a pin matrix-to-synth size ratio quite like the Cric's. If you're unused to pin matrixes, they can look dense and difficult, even a turn-off, but be not afraid. Even the most intimidating-looking pin matrix (the Cric's is 24x32) actually makes for quick work once you're used to it. The Cric ships with 25 coloured banana pins, which doesn't sound like many, but I rarely found myself needing more. Plus you can buy extras on the FSS site.

Pins also have more than a few advantages over patch cables, not least the fact that they are much tidier. Some synths' cabled patchbays are concentrated over to one side of their panels in the name of leaving the panel controls free of clutter, but to me, this can often end up feeling cramped and counterintuitive to patch. A pin matrix avoids this issue, since by definition its I/O must be concentrated into two simple axes. Dotted and solid lines are used here to differentiate sections and the patch points within them, and while it's not always very quick to trace an input or output up to the patch point, the included banana pins come in different colours which helps. This also makes unpatching and recreating patches much, much quicker, and to this end a notebook makes an ideal Cric companion.

If you are of the 3.5mm cable persuasion, the Cric has a handful of jacks above the pin matrix: four inputs and outputs, a global gate and volt-per-octave input, discrete Frequency

Future Sound Systems Cric

£2499

PROS

- Mammoth-sounding, with oodles of distortion, wave folding and filtering to explore.
- Hugely original and creative workflow.
- Excellent build quality.
- Plays very well alone or with other analogue gear.

CONS

- Lack of MIDI may disappoint some.
- It's expensive!

SUMMARY

Future Sound Systems have outdone themselves with a stunning synth. The Cric is as solid and reliable as it is explorative and creative.

CV inputs for the three oscillators and a trigger input for each of the Cyclical Function Generators.

Complex Oscillations... Cubed!

Which brings us neatly back to the other side of the panel and the Cric's core signal path. Considering there are discrete outputs for each of the Scissor oscillator's waveforms and discrete inputs for the DNA oscillators' frequencies — complete with FM attenuators on the panel — the you'd be forgiven for instinctively patching one oscillator into the frequency input of another when it comes to FM. This you can of course do, and with great results. But here is where the Cric really flexes its muscles, offering a number of handy switches for instantly adjusting the oscillators' interrelationship. In this respect, there's no signal normalising in the conventional sense, but there are interactions under the hood that can be instantiated natively. Generally speaking, analogue synths with more than one oscillator commonly place them all on equal footing — notwithstanding one usually having more waveform options or the like — but the »

■ The Cric's rear panel is where we find the power switch, four quarter-inch audio outputs, four quarter-inch audio inputs, a headphone port with level control, and an XLR input with gain control.





Measuring 550 x 310 x 80 mm and weighing 5.3kg, the Cric is also the vehicle for a particularly nice shade of blue.

» Cric handles things differently, using its switches to render the Scissor a kind of 'master oscillator' and the DNA oscillators its subordinates.

The first switch is Expo Lock, which forces the DNA oscillators to track the frequency of the Scissor oscillator by sending them the exponential frequency bus of the Scissor oscillator. Next is Sync Lock, which can either hard-sync or soft-sync the DNA oscillators to the Scissor oscillators. Then there's a switch simply labelled Scissor, which harks back to the FSS Recombination Engine Eurorack Module by sending pulse waves to modulate the amplitude of the DNA oscillators — essentially splicing them with on-off signals. This can be attenuated with the individual waveform VCAs of the DNA oscillators, either manually or via the pin matrix: no change at full, no signal at zero and even wave splicing in the middle. Lastly is PW Skew, which assigns the Shape modulation bus of the Scissor oscillator to the frequency of the positive and negative DNA oscillators, with that going to the negative one inverted. This combined with the fact that the DNA VCOs are inversely rectified means, in short, that (particularly

when used with the aforementioned Sync Lock or Scissor) the two will truly grind against one another to create some excellently guttural sounds, as throaty as a 303 and bursting with growling, formant-like timbral detail. It sounds absolutely fantastic.

The short version is that each switch does something, and each of those somethings is a lot of fun. This is as

“There's no other way to say it: the scope of the Cric's raw sound-sculpting capability is just massive.”

good a time as any to say that the Cric really does reward blind exploration magnificently. I felt nudged one way or the other at times, but only gently. Or, you might say, I felt a little lost at points, but never so much that I couldn't easily backtrack a couple of steps to get my bearings. It's a difficult balance to strike, and one I'd imagine took a fair amount of time in the R&D phase.

Form & Function

The four Buchla-inspired Cyclical Function Generators, or CFGs, each present a variable attack and decay stage with linear, exponential or logarithmic responses for both and variable-pulse outputs to boot. CFG 1 and 2 offer

independent slope time modulation on the pin matrix, while 2 and 3 offer one input for global modulation of the two. As the name suggests, these can be set to cycle or be used as more conventional envelopes with either AD or ASD topographies. To say their times are wide-ranging is a gross understatement. At maximum rates they go well into audio rate (approximately 3.6kHz, we're told)

and at minimum they can take over a quarter of an hour to complete a single cycle. Generative

ambient musicians rejoice! These do insist on resetting to the master Gate input even while set to cycle, so if you're wanting slower or asynchronous LFOs at play alongside the global gate, you'll have to plug a dummy cable into the 3.5mm input for any one of the CFGs.

Unfiltered Filtering

It's hard to believe, with all that going on in just the oscillator section, that we haven't even reached the filter. The one in question here is a driveable 18dB/octave state-variable design based on the FSS FIL4 Timbral Sculptor Eurorack module. I should know about the FIL4 Timbral Sculptor, since I reviewed it myself a while back over in the Modular

column. To quote that review: “at its mildest end the FIL4 very much does what it says on the tin: it sculpts timbre. At its most extreme end it creates new timbres entirely, mangling, morphing, ripping and pitch-shifting sounds into sonic creatures unknown.” And that’s about right for what goes on here.

Into the filter resonance chain can be inserted a Lockhart wavefolder for some very tasty distortion, full-wave or half-wave rectification and then a second (that’s right), more complex wavefolder can be placed before or after the filter processing. As with the oscillators, various combinations of these switches are where the magic happens, and to this end I can only say that hands-on exploration is the way to go. This filter does feel a touch tamer than the FIL4, which is no bad thing since that module — part of its charm — can quite easily bulldoze any sound if you’re not careful, and in the context of a larger instrument, it’s wholly appropriate that this one is a little more willing to play ball.

Risk & Reward

With this fundamental combination of an attitude-to-the-power-of-ten filter and various combinations of the aforementioned oscillator configurations, there’s no other way to say it: the scope of the Cric’s raw sound-sculpting capability is just massive. As for other circuits to augment that sculpting, there’s a standalone DC-coupled VCA, in addition to the various VCAs within the filter and oscillator sections, there’s a sample & hold which can draw on the noise as its source or an external input, and whose Hold input can be the pulse of CFG4 or come via the global gate input. There’s an XLR input with a low-noise THAT mic preamp capable of 60dB of gain, whose potential goes without saying considering the amount of distortion and filtering options on offer. When I say all-analogue, I really mean it: there is no MIDI, which may be a turn-off for some, but then again I also respect this, considering how MIDI-CV conversion can easily be handled externally.

I don’t know much about what went on behind the scenes during the making of the Cric, but I do know that it represents a risk of such proportion that most companies of Future Sound Systems’ size would never dare take it beyond the breadboard. “This instrument is the culmination of over a decade’s worth of tinkering, testing, exploration of and curiosity in all aspects of analogue sound synthesis,” says the manual’s foreword, and I don’t doubt it. Suffice to say, that risk has paid off. The Cric is an outstanding synth. Powerful, flexible, characterful and usable, it might take you a day to get used to its workflow, and its price might give you pause for thought before taking the plunge, but once initiated into its world I can say with some confidence that you are unlikely to regret it. **///**

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Aum Guitar Prana Pedal & Slide Saddle

Guitar Sustainer System

The Aum Guitar Prana Pedal is outwardly identical in connections and features to the pedal that came with the company's original Vibraslide, the brainchild of Aum Guitar's Gareth Whittock. Bearing some comparison with devices such as the Ebow and Sustainer guitars, the original Vibraslide used a hand-blown glass slide that connected to the pedal via a thin and flexible mini-jack cable. Whereas an Ebow houses both the pickup and string drive transducers, the Vibraslide used the output from the guitar's own pickups as the source, amplified the signal within the pedal, and then fed it back to the strings via a transducer within the slide. Like a Sustainer guitar, the Vibraslide was able to drive all the guitar strings at once.

This new system sees the pedal in a new green livery, but the key difference is in the slides, which are now available in glass or metal. Rather than the electronics being inside the slide, they are now attached to a small box fixed to the side of what's referred to as a Slide Saddle — this can be worn over a finger in the traditional way, rather than having to be held like a lap-steel guitar bar. However, if you want to use it as a lap-steel bar, you can, and the box section provides a useful handle. The strongest signal occurs when the box is on top of the slide rather than off to one side, but rocking the slide as you play provides a means of controlling the sustain intensity. As the slide can activate several strings at once and is just about long enough to cover a seven-string guitar, it could be interesting to experiment with alternate tunings.

The pedal requires a standard 9V, 300mA or higher, centre-negative PSU (not included), and a quiet, well-regulated supply is preferred, because a poor-quality supply can feed noise to the slide. There's

an on/off footswitch on the left of the pedal linked to a white status LED, while the rightmost switch operates a polarity inverter, which emphasises fifths or octaves (most noticeably on the lower strings), rather like the second switch setting on an Ebow, and in this mode the LED turns red.

A Pre knob sets the level of the guitar feeding the pedal, and if this is set too high there will be audible distortion — which may or may not be a good thing, depending on whether you can find a use for it! To the right is the Sys knob, which controls the level of the guitar signal being fed to the slide. The guitar connects, via the pedal's in and out jacks, to its usual amplifier, but there are two further jacks on the rear panel that function as an effects loop, allowing other pedals to be inserted into the signal path. These don't affect the guitar sound directly but do affect what's sent to the slide, giving rise to some interesting possibilities. For example, a wah pedal can change the overtones in the sustained sound.

In use, it's possible to feel small vibrations from the slide, and picked notes will sustain indefinitely if the Pre and Sys controls are set sufficiently high. If set too high, it becomes harder to control the strings that you don't want to hear vibrating, so it's useful to damp unwanted strings where possible when playing. I prefer these hollow Slide Saddles to the Vibraslide and, although there's a box and cable attached to the tubular slide section, it soon becomes second nature to position this somewhere comfortable. As with the original Vibraslide, the resulting sound can be very different from that of an Ebow, and not only because multiple strings can be sustained at the same time but also because it is possible to pick a string to produce a conventional attack and then have the note sustain. *Paul White*

£ Pedal with one Slide Saddle is €219, or €269 with two Slide Saddles. Additional Slide Saddles €60.

W <https://aumguitars.bigcartel.com>

Boss XS-1 & XS-100 Poly Shifters

Polyphonic Pitch-shift Pedals

Smooth, real-time polyphonic pitch-shifting remains a real technical challenge, and with most pitch-shift pedals processing artefacts are generally audible when shifting chords. Boss' latest attempts to rise to this challenge are the compact XS-1 and the larger, more controllable XS-100, both based on the same underlying pitch-shifting technology.

Adopting the familiar Boss compact pedal format, the XS-1 Poly Shifter is the more straightforward of the two new Boss pitch pedals, furnished with a Balance knob to blend between the dry and shifted signal, and a Shift Value rotary switch that determines how far the pitch is moved up or down. The first seven positions are semitones and the last three are octaves, with the choice of one-, two- or three-octave shifts. I found the numbers around the Shift Value control to be quite difficult to read, even when holding the pedal in front of me in daylight, so that could pose a challenge under subdued stage lighting.

A small three-way switch sets the direction of the shift or adds detuning if left in its middle position, and the main footswitch can be set either to toggle on and off in the usual way or to operate only when the switch is held down. In detuning mode, the Shift Value knob adjusts the amount of detuning up to ± 20 cents, and with the Balance control set around a quarter of the way up, this produces a very natural-sounding chorus effect. External footswitches can be connected to switch tunings.

With just a couple of knobs and a toggle switch, what's on offer? Set the Balance control fully clockwise and you hear only the shifted sound as set on the Shift Value dial. The first thing you notice on selecting a shifted sound is the lack of grainy artefacts. The sound is almost as smooth as using a capo, even on fancy chords, but there's no formant correction included in the processing, so the apparent guitar body resonance will change along with the pitch-shift. If only moving two or three semitones up/down you won't really notice, but if you shift up by an octave the guitar body will sound smaller, almost mandolin-like; conversely, if you shift down the body will sound larger. The harmonic content of the string is also shifted, so when shifted upwards the sound will be brighter



and harmonically different from the same note played unshifted. That being the case, I'd say that for the most natural-sounding shifts, moving up or down by just two or three semitones is safest. If you don't mind the tonal changes, then you can go further.

As a virtual capo then, the XS-1 is very practical, with downward shifts producing a very usable baritone guitar effect. Select an upward shift of one octave and then adjust the blend to produce an effective substitute for a 12-string guitar. OK, so you hear octaves on all six strings, not just the lower four, but in a live context nobody is going to notice. Detune plus a suitable blend setting gives you a natural-sounding chorus but without the obvious LFO modulation of a typical chorus pedal. It is also worth noting that any processing latency is so low as not to be significant. Some online comments suggest that the XS pedals are not great at following note bending, deep vibrato or pinched harmonics, but I tend to use all of those techniques and didn't find any obvious issues. In my tests I placed the XS-1 before any overdrive pedals.

With the XS-1 being the more straightforward of the two pedals, there are some limitations. For example, I would have liked to be able to use the virtual capo mode with octaves, but here you have to choose one or the other. You can't capo up two semitones and have octaves at the same time, so no capo'ed 12-string sounds. Similarly, you can't add detuning to a shift setting, only to the unison sound.

For those with more sophisticated needs, the Boss XS-100 Poly Shifter can store 30 presets and comes with an integrated expression pedal incorporating an assignable toe switch with the ability to pitch the note up another octave when pressed hard. This

makes it significantly larger than the XS-1 but it still maintains a reasonably narrow footprint, and the rear connectors help conserve pedalboard space. Its functions and form factor invite comparisons with the DigiTech Whammy DT, which, although far from new and not programmable, still offers cleaner pitch-shifting than most competing pedals.

The XS-100 has input and output jacks as well as a thru jack, mini-jacks for MIDI in and out, and a USB-C connector for facilitating firmware updates. The pedal's front panel is surprisingly sparse, with a turn/push encoder, two buttons, and two footswitches: Effect and Tune Down. Effect switches the pedal-controlled effect on or off, but if both switches are pressed together the two switches then act as preset up and down selectors (SW mode). Tune Down brings in the downtuning or capo effect as stored in the preset, unless the switches are in SW mode. Presets may also be navigated by turning the encoder.

A small display screen covers patch selection, parameter editing, tuning functions and so on. During

Unlike the XS-1, the XS-100 caters for external MIDI control.

performance the display shows the patch number to the right, while to the left there are values for the pitching functions. Unlike the XS-1, detuning and pitch-shifting can run at the same time.

The Effect and Memory/Write buttons allow users to scroll through presets, access the tuner, explore the edit menus and save user patches. Memory switches between Memory mode and System Settings mode, while Effect switches between Effect mode and Editor mode. A separate CTL 1/2 EXP 2 input allows a second expression pedal to be used to control additional parameters during performance, including pitch, volume or modulation intensity. Any custom switch settings are stored along with the preset.

Offering all that the XS-1 does and more, the XS-100 can also handle pitch-shifts of up to plus or minus four octaves — so your pet bat might need ear defenders. Whammy-style pitch bends can be actioned over a user-defined range as with the XS-1, and there's more flexibility in how the various pitch functions are combined.

The XS-1 and XS-100 show us that pitch-shifting is no longer the poor relation of the pedal world. Those warbly, slightly out-of-tune artefacts are no longer a problem, and though the shifted sounds still don't sound completely natural, especially at large shift values, they are sufficiently smooth and well behaved to be taken seriously. Until now the polyphonic pitch pedal market has been dominated by DigiTech, but now it has some real competition.

The compact XS-1 brings smooth pitch shifting into the compact pedal world, while the larger XS-100 has a built-in expression pedal, preset handling and MIDI support for external MIDI control and integration into MIDI-controlled rigs.

Clearly, Boss view the XS-100 as a programmable competitor to the DigiTech »



» Whammy DT — it even has a similar retail price! But while the XS-100 offers more flexibility, the XS-1's compact size and simplicity mean it should have good potential for users who just need a virtual capo that can shift down as well as up. It can also create a detuning chorus effect, or function as a 12-string guitar emulator. Add

an expression pedal and it can produce whammy-style pitch glides.

As commented, the XS-1 makes a practical alternative to a capo, and as it can pitch down as well as up, it can transform a regular guitar into a baritone or even a bass, while mixing in octave shifts can also mimic 12-string sounds. The XS-100 can do

all of that too but may be better suited to those who like the idea of a Whammy DT pedal but who need more customisable pitch effects during performance, along with user presets and MIDI integration. *Paul White*

£ XS-1 £198, XS-100 £299.

Prices include VAT.

W www.boss.info

Neural DSP Archetype: Misha Mansoor X

Guitar Rig Emulation Plug-in

Neural DSP's guitar-rig modelling plug-in range caters for almost every style of guitar tone you care to think of, but the company have won a particularly loyal fan base in the modern metal and rock community. Their latest release, Archetype: Misha Mansoor X, will certainly appeal to that audience, as Mansoor's innovative guitar playing, and his writing and production work with bands such as Periphery and Animals As Leaders, have influenced a whole generation of modern metal guitar players. Perhaps best known for using down-tuned and extended-range guitars (his endorsements include a signature seven-string guitar with Jackson), Misha has tailored his rig to work in that context and, of course, that's reflected here within Neural's virtual recreation.

I've explored the GUI of the Archetype plug-ins in previous Neural plug-in reviews, and as this release follows a similar format I won't repeat myself here, other than to say the polished GUI houses the familiar set of global controls (including access to presets, the excellent doubler, tuner and MIDI configuration options) and multiple tabs to access separate sections of the rig's signal flow. In this case, the sections comprise Special FX, Pre Effects, Amp, Cab, EQ and Post Effects options.

The three amp models are labelled Clean, Rhythm and Lead, and while they can all deliver a wide range of gain, things get hotter as you move through these options. These models will undoubtedly be inspired by some real amps, but I suspect they might well be modded versions and there may be, in this software version, a few additional mods (such as the Voicing switches) to add further tonal options. The three different cabs have corresponding amps, but they can be mixed and matched with the amps, and each includes a collection of IRs provided by Adam 'Nolly' Getgood. The Cab module also offers standard IR loading. Even if you



ignore the rest of what's offered, there is a huge range of tonal options available, from crystal cleans through to crushing high gain. However, even at higher gain settings, it's possible to avoid the low end getting out of control. I suspect seven- and eight-string players will find this low-end clarity particularly useful.

The Pre Effects include compression, tape delay, overdrive, octaver and distortion options, but it's the Special FX section that will catch the more experimental eye. The likely inspiration for the Laser pedal, which combines ring-modulation and pitch-shifting, might be familiar to Fractal Axe-Fx users (Misha has used Fractal units live for some time), but to fully exploit it really requires a suitable MIDI expression pedal. That's not the only creative option, though, as the Glitch pedal is just as wild a ride. Like Hologram's Microcosm hardware pedal, this uses granular processing, chopping your audio signal into grains and then reassembling them in all sorts of weird and wonderful ways. If you enjoy taking your guitar performance into experimental sound-design territory, then this combo might be reason enough to check out this plug-in.

After a nine-band EQ, the Post Effects section adds to the creative possibilities.

The Stereo Delay and Reverb pedals offer conventional ambience treatments but, with a neat ducker feature within the delay and a shimmer mode for the reverb, there are some cool sonic possibilities. However, the Modulator pedal, which features both expression pedal and 'auto' modes of operation, lets you experiment with sweeping chorus or flanger-based effects.

Like all Neural DSP's plug-ins, the quality of the modelling is excellent. If you're looking for your first top-shelf guitar rig plug-in and need it to cover all sonic bases, perhaps there are more obvious choices out there, but anyone looking for a selection of high-quality tones that span the modern metal and rock genres really should put this on their wishlist. And at the same time, you will also get a whole world of guitar sound design included for good measure. While it will obviously appeal to Misha's guitar-playing fanbase, you don't need to be a member to appreciate this plug-in! You can, of course, try before you buy (there's a 14-day trial), but if metal guitar is your thing and you can afford the price of entry, I can confirm that Archetype: Misha Mansoor X sounds excellent.

John Walden

£ €149 including VAT.

W www.neuraldsp.com

Decapitator

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From subtle to extreme, analog saturation is an integral part of great mixes. That's why we studied all the analog classics, and painstakingly created a plug-in that brings the best of analog saturation to your studio. Choose from five different models to add analog character to every track and bus in your mix.

"Decapitator is instavibe-in-a-box."

- Fabrice "Fab" Dupont
(André 3000, Shakira,
Jennifer Lopez, Marc
Ronson, Isaac Hayes)



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Spotlight

Eight-fader Control Surfaces

LUKE WOOD

The last time we looked at control surfaces, the focus was on single-fader units. And whilst many people are perfectly happy with just the one fader (or indeed, none at all!), others prefer working with something that feels a little more like a traditional console. At the top end of the market you have the likes of Avid's S4/S6 and Yamaha's Nuage — big, modular beasts designed to take the place of large-format analogue mixers in commercial mixing and post-production studios — but there's now a healthy selection of products aimed at home and project studios that still offer a useful amount of hands-on control, but in smaller, far more affordable packages. This month, we're rounding up a selection of eight-fader surfaces that offer DAW control either over protocols such as HUI and MCU, or in some cases, with some clever use of good old MIDI.

Akai MIDIMix

Akai's MIDIMix offers a compact and affordable way to add some physical controls to a computer-based setup. It features eight main faders and a dedicated master fader, along with record arm and mute/solo buttons and 24 knobs — they're not endless encoders, so

There's more choice than ever in the world of control surfaces. We check out some current options that won't relieve you of too much space.

they won't remain in sync to their mapped parameters when changing between tracks/instruments. It's a MIDI-only unit, so will only act as a DAW controller in software that supports parameters being mapped to MIDI commands. It does include a script that offers one-to-one control mapping with Ableton Live, though, and comes supplied with a copy of Live Lite, as well as a six-month subscription to Reason Studios' Reason+ upon registration.

£ £77 including VAT.

W www.akaipro.com

Avid S1

The smallest member of Avid's S-series range comes equipped with eight touch-sensitive motorised faders, each of which is joined by mute, solo, record arm and select buttons, an LCD screen, and a rotary encoder with another pair of push buttons. A row of function buttons along the top of the surface can be used to assign the encoders to various DAW and plug-in parameters. The surface is designed to be paired up with a tablet running the free Avid Control iOS/Android app, which although not strictly necessary, does add some useful functionality such as detailed metering

and an array of customisable soft keys. It communicates with a DAW using Avid's own EuCon protocol, which, as you'd expect, offers in-depth integration with Pro Tools, but is also officially supported in Logic Pro, Cubase and Nuendo. Although it's capable of functioning as a standalone surface,

the S1 pairs up neatly with the Avid Dock, which acts in a similar fashion to the master section modules found in the S4 and S6, adding a master fader, transport, automation controls and more in-depth session navigation tools. It's possible to combine up to four S1s and a Dock in a single setup, and they all lock neatly together thanks to magnetic strips tucked away in their side panels.

£ £1429 including VAT.

W www.avid.com

Behringer X-Touch / X-Touch Compact

Behringer's X-Touch includes main faders each equipped with an LCD scribble strip, a rotary encoder, LED metering and a set of select, record arm, mute and solo buttons; there's also a ninth fader that acts as a dedicated master. The unit's right-hand side is occupied by a wealth of controls that include mode selection for the encoders, assignable function



Akai MIDIMix



Avid S1



Behringer X-Touch

keys, DAW automation and utility controls, navigation tools, a transport section and a large jogwheel. Along with MIDI, the X-Touch supports both HUI and MCU, allowing it to integrate seamlessly with the majority of DAW platforms. If you want to expand your setup further down the line, there's the X-Touch Extender, which houses eight additional faders, screens, encoders and buttons that share the same layout as the controller's main section. The X-Touch Compact then offers a scaled-down version that features the same amount of faders but without the scribble strip, and replaces the array of function keys with a set of eight additional rotary encoders and a smaller transport section.

£ X-Touch £425, X-Touch Compact £303. Prices include VAT.

W www.behringer.com

Icon Pro Audio V1-M / P1-M

Icon Pro Audio's V1-M offers a wealth of hands-on control that comprises a selection of tactile controls and a grid of touchscreen buttons that can be assigned to everything from hotkeys to MIDI commands. There are nine motorised faders in total (one is a dedicated master fader), and each of the channel faders is accompanied by a rotary encoder and four function buttons. Plenty of visual feedback is available



Icon Pro Audio V1-M

thanks to a TFT meterbridge-style display, and the large touchscreen interface is joined by a set of transport buttons and a jogwheel. Everything can be configured using Icon's iMAP software application, which comes pre-loaded with custom mapping configurations for 18 DAWs — users are free to customise the included maps if they wish. The smaller P1-M then offers a more compact alternative, and saves on space by losing the row of dedicate buttons for each channel, as well as offering a reduced set of function keys and a smaller jogwheel; a customisable touchscreen display is still present. Both the V1-M and P1-M can be expanded with an additional set of eight faders, encoders and buttons by pairing them up with the V1-X and P1-X respectively, and Icon also offer an optional meterbridge-style row of display screens for both the P1-X and its extender.

£ V1-M £1186.80, P1-M £646.80. Prices include VAT.

W www.iconproaudio.com

JLCooper FaderMaster Professional

JLCooper's FaderMaster Professional is about as simple as a control surface can get, housing just a set of eight 100mm faders and eight buttons, although it is possible to link multiple units to create a larger surface. MIDI I/O is provided on 5-pin DIN sockets, and



JLCooper FaderMaster Professional

there are two quarter-inch TRS sockets that can be used to integrate an external foot pedal and footswitch. All of the controls are freely assignable, and the surface can be configured using a Mac-based software application. As it's a MIDI-only device with no support for protocols like HUI or MCU, the FaderMaster Professional will need to be used alongside a DAW that supports MIDI mapping.

£ \$999.95.

W www.jlcooper.com

Korg NanoKontrol 2 / Studio

If you're tight on space, then it's worth taking a look at Korg's NanoKontrol 2. The latest iteration of the ultra-compact controller packs in eight mini faders, each of which is joined by a rotary control and dedicated solo, mute and record-arm buttons. The left-hand side of the unit features a set of transport controls along with buttons for engaging loop/cycle playback modes, setting and navigating track markers and DAW track navigation. Everything can be customised using the Korg Kontrol Editor software, which allows users to define the type of MIDI message transmitted by each control, as well as setting their minimum/maximum values. Because the NanoKontrol 2 is a MIDI controller, it's possible to use it to control software instruments and sample libraries as well as a DAW. The larger (but still relatively compact) NanoKontrol Studio then offers a similar layout, but with the addition of per-channel select buttons,



Korg NanoKontrol 2

» a jogwheel, built-in scene switching and Bluetooth connectivity.

£ NanoKontrol 2 £49, NanoKontrol Studio £120. Prices include VAT.

W www.korg.com

Mackie MCU Pro

The MCU Pro is the latest in a long line of control surfaces from Mackie, who actually developed the MCU (and earlier HUI) protocols that have become the standard method of communication between many DAWs and control surfaces. The current model's main section sports nine touch-sensitive motorised faders (eight channel faders plus a dedicated main fader), along with eight assignable V-Pot rotary controllers and dedicated record-arm, solo, mute and select buttons. A section above the main faders offers Track, Send, Pan/Surround, Plug-in, EQ and Instrument modes for the V-Pots, along with buttons for bank navigation and fader modes. The right-hand side of the controller packs in an array of assignable function buttons, as well as offering dedicated sections for selecting automation modes, triggering keyboard modifier keys and selecting important menu functions such as undo and save. The lower-right corner features a transport section, navigation/zoom controls and a jogwheel. As for visual feedback, there's a digital scribble strip above the eight channel faders and a counter that can display time, bars/beats or SMPTE timecode. Thanks to a built-in MIDI interface, it's possible to connect up to three of Mackie's XT Pro units, each of which adds another bank of eight faders, V-Pots and channel function buttons.

£ £1274 including VAT.

W www.mackie.com



■ Mackie MCU Pro



■ Nektar Panorama P1

Nektar Panorama P1

Occupying around the same space as a 17-inch laptop, Nektar's Panorama P1 houses a set of nine 45mm faders, 16 rotary encoders, 10 LED buttons and 28 standard push buttons — those looking to get some more limbs involved will be pleased to see there's also a footswitch input. A full transport section is present, and this helpfully doubles up as an additional set of function keys that can be configured to transmit MIDI commands or QWERTY keyboard macros. Visual feedback for the controller's various operation modes is provided by a 3.5-inch colour display, which also makes it possible to carry out MIDI mapping assignments on the unit itself. As the Panorama P1 is a MIDI device, it can be used to control a whole host of software and virtual instruments. However, Nektar have provided pre-built configurations that offer in-depth integration with a selection of DAWs straight out of the box. Currently, users of Bitwig Studio, Cubase, Nuendo, Logic Pro, Reaper, Reason and Studio One all enjoy dedicated Mixer, Instrument and Transport modes, along with a fourth mode that provides access to standard MIDI control.

£ £259 including VAT.

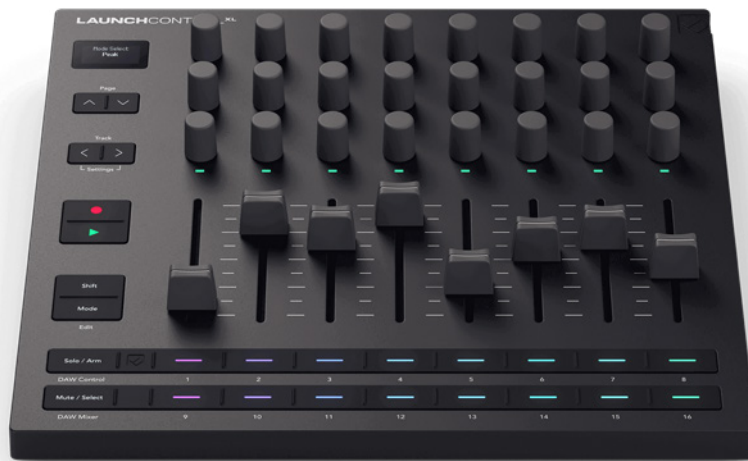
W www.nektartech.com

Novation Launch Control XL Mk3

The latest iteration of Novation's MIDI fader controller offers a wealth of hands-on control, and thanks to a set of custom scripts, can be used to control a selection of DAWs (Ableton Live 12 or later, Logic Pro 11 and later, Cubase 12 and later, FL Studio, Bitwig) as well as any MIDI-capable hardware and software instruments. There are eight 60mm faders, 24 endless encoders with RGB LEDs and 16 assignable buttons, along with dedicated navigation, page and play/stop and record buttons — the Mk3 version also introduced an OLED display for some visual feedback. The controls can be custom mapped to suit just about any instrument or setup, and it's possible to store up to 15 configurations. As for connectivity, the unit offers MIDI I/O over USB as well as 5-pin DIN sockets. The Launch Control XL Mk3 is available to pre-order at the time of writing, and should be shipping by the time you read this!

£ £189 including VAT.

W www.novationmusic.com



■ Novation Launch Control XL Mk3



PreSonus FaderPort 8

PreSonus FaderPort 8

PreSonus' take on an eight-fader control surface comes in the form of the FaderPort 8, which offers a middle ground between the FaderPort (a single-fader surface) and the larger FaderPort 16. You get eight 100mm, touch-sensitive motorised faders, each of which is accompanied by an LED scribble strip and select, mute and solo buttons, while a multi-mode encoder at the top left of the surface offers control over the likes of panning and plug-in parameters. A collection of 65 buttons offers hands-on access to 78 functions, and there are sections dedicated to automation modes, track navigation, zooming/scrolling and transport. As you'd expect, the FaderPort 8 boasts in-depth integration with PreSonus' Studio One DAW (and comes with a six-month membership for the Studio One Pro+ subscription), and there's also a customised mode that extends the courtesy to Ableton Live users. Other DAWs aren't left out, though, as support for MCU and HUI offers compatibility with all of the major alternatives.

£ £469 including VAT.

W www.presonus.com

Softube Console 1 Fader MkIII

Softube's Console 1 Fader MkIII breaks convention in that it actually has 10 faders, but given its size, it seems reasonable to include it here anyway! The motorised faders have been upgraded in the MkIII version, and now offer haptic feedback that can emulate detents,



Softube Console 1 Fader MkIII

along with touch sensitivity. They are joined by select, solo and bypass buttons, the latter two of which double up as polarity and bypass function switches, and a row of 10 OLED displays that offer a digital scribble strip and parameter readouts. As with its predecessors, the unit integrates tightly with Softube's Console 1 software, but also includes Pro Tools HUI and Generic

MCU modes that allow it to function as a standard controller in all major DAWs, along with an Apollo Central mode that allows it to control Universal Audio's interface software mixer. Helpfully, it can also be used as a MIDI CC controller. Thanks to dual USB-C ports, it's possible to link up to four units without a hub, and there are a selection of mounting options available, including a rackmount kit, a VESA mount, and a stand that allows the Fader MkIII to be housed below a Console 1 Channel MkIII.

£ £939 including VAT.

W www.softube.com

SSL UF8

The SSL UF8 features eight 100mm touch-sensitive motorised faders that are each joined by a high-resolution colour display that offers a virtual scribble strip, level metering and channel routing information. Eight dedicated select, mute and solo buttons are present, along with eight multi-function V-POT encoders derived from SSL's large-format Duality and AWS consoles. As with the other members of SSL's controller family, the UF8 integrates with the company's SSL 360 software, which acts as a central control



SSL UF8

point and integrates the surfaces with SSL's 360-enabled plug-ins and the user's DAW. All of the major DAWs are officially supported out of the box — the UF8 can control up to three simultaneously — and it's possible to control software that isn't currently on the list thanks to HUI, MCU and MIDI CC compatibility. There are also two quarter-inch inputs that can be used to assign a variety of functions to external footswitches. Up to four UF8s can be linked together to build a larger surface, and it's possible to team the unit up with SSL's UF1 and UC1 in order to build a virtual SSL console with a dedicated master fader, full transport section, additional metering and hands-on channel strip/bus compressor controls.

£ £1198.80 including VAT.

W www.solidstatelogs.com

Studiologic SL Mixface

Studiologic's ultra-compact SL Mixface has been designed as a mobile solution for those working on the go. It



Studiologic SL Mixface

features nine faders (eight main faders along with a master), along with a set of eight buttons and rotary controls.

Despite the minimal footprint, Studiologic have found space on the SL Mixface's panel for a set of transport controls, and it will also attach magnetically to the company's range of controller keyboards and act as an expander for their built-in controls. Although it's a MIDI-only controller, it does offer automatic parameter mapping for all of the major DAW platforms — a full compatibility list is available on the company's website.

£ £169 including VAT.

W www.studiologic-music.com

Emergence Audio Infinite Guitar

Kontakt Instrument

★★★★★

Emergence Audio have a reputation for creating filmic instruments for NI's Kontakt, and the latest addition is Infinite Guitar. The source material is mainly electric-guitar-based but these are configured as layered, modulating textures best described as pads, ambient swells, granular-like beds, harmonic drones and hybrid tones that would fit in very well with film (especially sci-fi), TV or ambient music mixes. Underpinning the instrument is Emergence Audio's Infinite Motion Engine, which is responsible for producing a sense of continuous and non-repeating motion within each note. The majority of the individual samples are very long, which is why the library takes up almost 47GB when installed, and each of them evolves tonally — some smoothly, others pulsing or incorporating seemingly random delays and granular shimmers.

The Infinite Motion Engine allows for two samples to be layered, with modulation options to continuously change the balance between the two layers. The LFO section can also be used to modulate other parameters with independent depths of modulation: Pitch, Pan, Resonance, Effect Send amount and Filter Cutoff can all be modulated, and all the usual modulation waveshapes are available. The filters can be set to low-pass or high-pass, and the individual samples can be freely retuned up to two octaves in either direction. There's also a so-called intelligent randomisation function that switches out the samples and changes the pan and filter settings, and this often comes up with very usable patches. Each sample layer has full ADSR control, and there's a dedicated effects section that includes a convolution engine for lifelike reverb and other effects.

Working through the generous set of factory samples gives a good idea of what Infinite Guitar can do, though creating your own patches — even if just by switching out samples in an existing patch (or pressing the Random button) — can be very rewarding. Many of the sounds work well with just a single note or perhaps very simple chords, and the sense of movement means you can hold them for as long as you like without the user experiencing that sense of repetition that you get with straightforward



looping. I found that some of the preset sounds seemed a little low in level (which is apparently intentional to preserve dynamics), but these are easily tweaked, and for some of the sounds I found that following the instrument with a compressor plug-in worked really well. There are deep, ominous drones, haunting meditations, glittering ear candy and soothing ambient backdrops that literally go on forever (or at least until the next power cut). I came away very impressed by the way these carefully crafted sounds evolve in an almost hypnotic way, and I can definitely see a role for them in my own ambient creations. *Paul White*
£229
www.emergenceaudio.com

Soniccouture Deep Felt

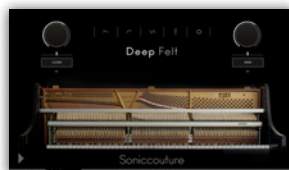
Kontakt
Instrument

★★★★★

Soniccouture's Deep Felt is a beautiful Kontakt instrument for a specific piano timbre that continues to be very on-trend, thanks to its usage by numerous electronic artists and soundtrack composers.

The principle behind this piano sound relies upon a pedal commonly found inside upright pianos, which drops a layer of felt between the hammer and strings. Often described as the practice pedal (with the original intent being to decrease the instrument's volume for the purposes of practice), it has now become a piano category all of its own.

Soniccouture have sampled an Essex upright, which they cruelly describe as "nothing fancy!" I would disagree, as the Essex brand is part of the Steinway family, albeit manufactured in the Far East, rather than Hamburg or New York. The sampling process employed a pair of Schoeps MK4 mics taking care of a more intimate close-up signal, with an equally stunning pair of Earthworks used as a more distant spaced pair. The net result is a visually appealing minimal interface, with volume controls for each of the signals. Even in its default mode, the quality of the sampling is truly engaging



Sample Libraries

when played, sounding more European than I might have expected.

An enormous amount of attention has been paid to the resonance across the range of the instrument. Despite the felt, initial string strikes provide a full tone, coupled with a beautifully sympathetic cabinet resonance which is particularly evident in the upper register of the instrument. It's very organic and natural in sound. There are also volume adjustments for the actions, such as the hammer against the felt and the pedal mechanism. This is a medium-sized upright, so bass reproduction isn't huge, but it is firmly in keeping with the intimate nature of the instrument.

This library does, however, move way beyond the basic felt piano sound. SC provide five menus that allow control of envelope, filter and LFO modulation. This may feel counterintuitive, being a piano library, but the lengths to which you can go to create interesting snapshots (patches) is very welcome. As if to prove the point, SC provide 36 snapshots that demonstrate everything from natural usage to overt sound design. This is heavily bolstered by the effects section, which provides some exemplary treatments, such as bit-crushing and saturation — both surprisingly useful!

There is also a Generative Tools page, which includes sequencers named Jammer and Weaver. The latter provides the ability to host up to eight lanes of sequence, so that when you play a chord, each note will be directly triggered and effected by the Weaver lane, in a form of sequential playback. This is totally mesmerising, providing beautiful pulsating Steve Reich-like backdrops with incredible ease.

Like so many previous encounters with Soniccouture, this library, which provides up to 18 velocity layers within 6.4GB of compressed sampled content, begins with a beautifully captured instrument. The additional elements are an enormous bonus, extending a beautiful library way beyond its first impression. *Dave Gale*
£119

www.soniccouture.com

Sonora Cinematic Aethervox

Soundbox Instrument



For those yet unfamiliar with Audiomodern's Soundbox Player, it is a sample playback engine that can host and blend four sample-based presets. A free version is available, and it runs inside iOS, macOS or Windows DAWs as a plug-in. A paid version is available with additional editing features such as a sample and mapping editor. Most of the available sound packs are very attractively priced and, once installed, samples from any of the packs may be combined to create new user presets. Soundbox Player has separate envelope controls for each sample, modulation families, transpose options, arpeggiators and a dynamic blend panel that creates movement by applying a choice of geometrical shapes to a blend cursor. It supports MPE (MIDI Polyphonic Expression) and is also multi-touch enabled, making it one of the few samplers that offer both MPE and full multi-touch support on iPad.

There are a couple of free sound packs that are worth exploring, but the focus of this review is the new Sonora Cinematic Aethervox, a pack that provides, as the name suggests, vocals of an ethereal nature. Sonora Cinematic have produced some wonderful vocal sound libraries and while Aethervox may only comprise 15 presets, up to four presets may be combined to create vocal textures that shift and evolve in a very emotive way. Apparently Aethervox's source sounds are in the spirit of Sonora Cinematic's respected Aria Vocalscapes and Emma Legato Kontakt libraries (I can't say whether the samples used here have been taken from those libraries or not), and then formatted to produce this set of cinematic-style vocal textures designed specifically to run in Soundbox. The whole library occupies only 350MB of drive space too.

There are pads, evolving textures, drones and haunting mini phrases (non-tempo-sync'ed vocal improvisations over suspended chords) that are well suited to film/TV scoring, ambient production and experimental electronic music. Some of the sounds are fixed notes, albeit very expressive ones with some lovely natural vibrato creeping in as the notes evolve. Others comprise a series of perhaps three notes that sound wonderfully evocative



when played individually. There are some lovely hums and also sibilant whispered S to Ahh and S to Ooh sounds. There's a lot of scope for building up vocal layers and I also managed to create some very useful ambient pads by combining Aethervox sounds

with some of the free sound pack presets.

Some sounds can be played as you would a choir pad, others with the ability to conjure up emotion by holding down a single note and letting the sounds evolve. The sound design is well balanced between full-sounding timbres and more airy layers, and can add a genuine sense of magic and mystery to a composition. Soundbox Player's controls provide the means to blend textures and adjust dynamics without the user having to get bogged down in complexities, while the ability to add automatic movement really helps keep things interesting. This may be a relatively small library and one that falls at the lower end of the price bracket at that, but it really is capable of some beautiful-sounding results. *Paul White*

£15

www.sonoracinematic.com

Edu Prado Sounds Dark Mandolin

Kontakt Instrument



Busy media composers

looking for a source of sounds off the mainstream beaten track could do a lot worse than checking out the catalogue of Kontakt-based instruments from Edu Prado Sounds. We reviewed their excellent Extended Electric Guitar library in the August 2025 issue; it's niche, compact, accessibly priced and full of character. Their latest release is Dark Mandolin, a collaboration between Edu Prado and Jan Pfitzer, and the product blurb suggests it might tick a number of the same sorts of boxes. If you have an affinity for the conventional made unconventional, then read on.

Dark Mandolin is a Kontakt-based instrument (including the free Kontakt Player) and features a finger-played mandolin recorded in parallel paths to both pristine digital and analogue tape recorders. Samples from both pathways are available to the user and presented within Kontakt as five main instrument categories — Natural, Tape, Room, Duo and Swarm — with velocity

layers and round robins included to ensure realistic performance dynamics. The first two instrument categories are self-explanatory. Room places the sounds within a small warm space, while Duo provides a wide doubling effect with different samples placed left/right of the stereo image. Swarm is a little more experimental with what seem like rapidly played repeating notes.

All these instrument categories offer a selection of presets within them and, while the Natural and Tape presets include conventional mandolin sounds suitable for conventional performance styles, as the screenshot here suggests, the UI provides a selection of sound-design options that let you take the underlying sounds of the mandolin in a somewhat less conventional direction, and many of the presets demonstrate what's possible here.

The Envelope, Filters and Space (with various convolution reverb options) sections do exactly what you might expect. However, you also get a compact step sequencer with variable step count (up to a maximum of 16 steps), a selection of arpeggiation options, preset velocity patterns and the option to adjust the step length. The Heat and Burn sections provide gentle saturation and distortion respectively. The Lo-Fi effect can

add wow/flutter and bit reduction, while the Carve effect seems to provide emphasis to the attack or sustain portions of the sound

via a combination of EQ and/or transient processing. Finally, the Crystal effect lets you dial in a sort of dreamy, ethereal element to the sound. It's a cool effect and can take the mandolin starting point into soundscape territory.

The end results can be intimate and realistic or beautiful and otherworldly, going from simple plucks through to enchanting pads. Yes, it is a library with a very particular sonic signature, but it is full of character and surprisingly flexible and modestly priced. If you like the unique flavour that boutique sample libraries can bring to your compositional palette, then Dark Mandolin is well worth auditioning via the Edu Prado Sounds website. *John Walden*

£35

www.sounds.eduprado.com



Audio examples of this month's libraries are available at www.soundonsound.com.

THE SOS TEAM

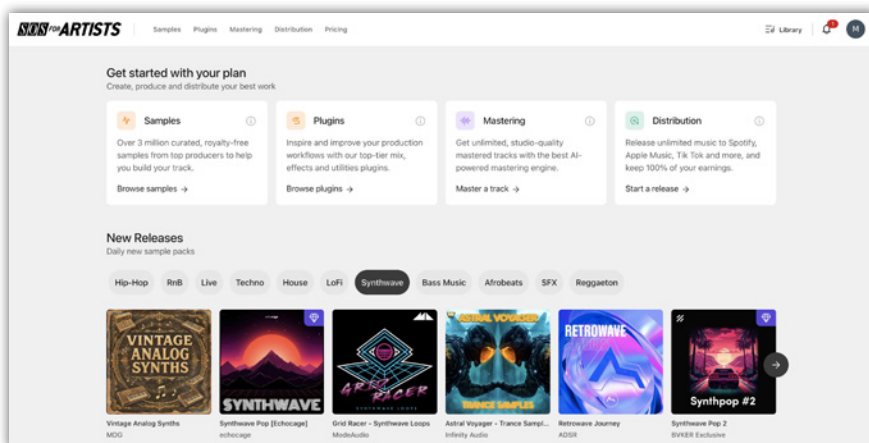
When we first launched SOS For Artists (SOSFA), it wasn't all that easy to try before you buy. Now, thankfully, you have a number of options. Not only do we have two subscription tiers available — Essentials and Pro (of which more another time) — but so that you can get a good taste of what's possible before you commit to a sub, you can now sign up to a free version of SOSFA, and test-drive a more feature-rich seven-day trial version. Obviously there are plenty of features you can only find in the main subs — we couldn't give away *everyone's* plug-ins for free forever, for example! — but you should find that there's more than enough in the free and trial versions to allow you to experiment with ideas, plan projects and generally get a feel for whether our platform is a good fit for you. In this article, we'll explore this free version in more detail, as well as running through the added extras in the seven-day trial.

Exploring the Free Version

In case you're completely new to this, creating an SOSFA account is really straightforward: just provide an email address and password, and you can then immediately access the dashboard. The GUI is pretty clean and we hope you'll find it intuitively organised! At the top are four main tabs: Samples, Plugins, Mastering and Distribution. Clicking on a tab takes you to a brief overview of what you can do, and even the free version of SOSFA allows for meaningful exploration of what's on offer. Even if you're just starting out, you should find it easy enough to navigate that you can jump straight into previewing loops, structuring tracks and planning a release, all from this single interface.

The **Samples** tab is where many people's creative journey will start. Here, you can not only browse thousands of loops and one-shots, but also audition them directly in the browser and, better still, test combinations of them using the built-in eight-track Creator. This offers an interface and engine that allow you to layer multiple samples, adjust them in volume and experiment with arrangements. Plus, you can save favourites to folders so you can revisit them later. Even though there are some limitations compared with

You can now explore SOS For Artists with a free trial!



■ You can now create an SOSFA account without paying to subscribe, and can start exploring the content and features found in the four main tabs.

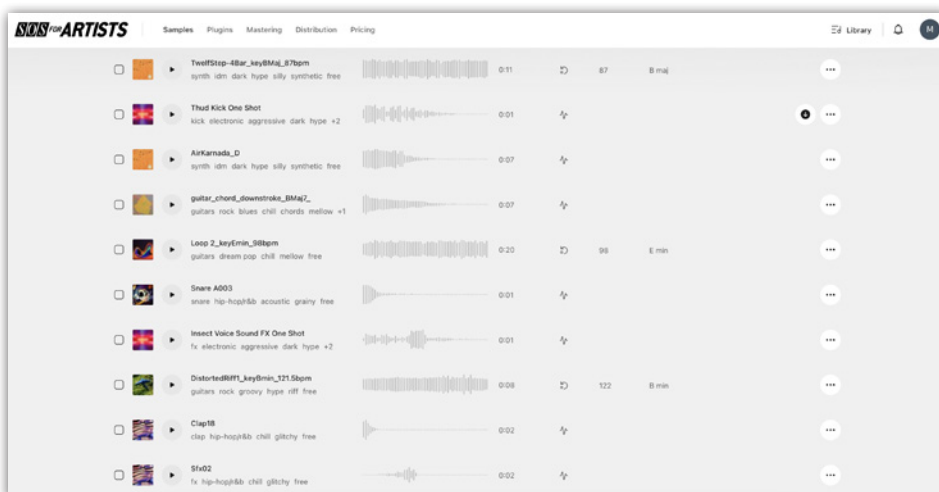
the paid subs, this tab is invaluable for testing ideas and building future projects. You can quickly audition combinations and find inspiration for beats, harmonies or textures. SOSFA offers a wide breadth of options, but with the added advantage of dropping these sounds into the Creator section, so that you can audition samples together without needing to export them elsewhere.

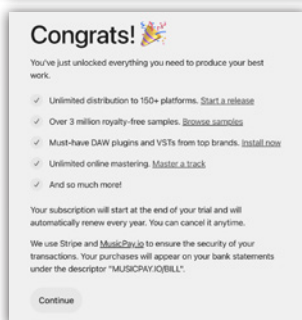
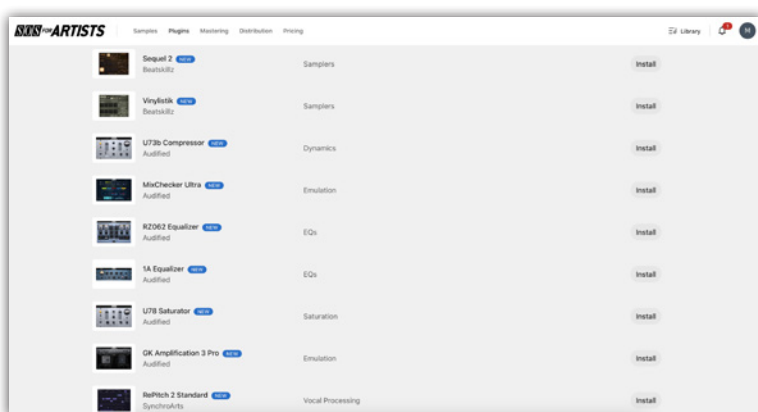
The **Plugins** tab gives an overview of more than 40 professional tools that are compatible with all the major DAWs. So that you can gain insights into how a plug-in might (or might not!) enhance your setup or workflow before installing it, you can view specifications, explore features and watch demonstration videos, all without leaving SOSFA and having to trawl the Web. Obviously, the plug-in licences themselves aren't included in the free version but you can test quite a number of them in the seven-day trial

(more on that later), and the demos and documentation should give you enough information to plan your projects and understand how each tool could be applied in your productions.

In the **Mastering** tab, you gain some access to the LANDR automated mastering service. Here, you can upload tracks and experiment with different mastering presets. The free version lets you hear how your track would sound after mastering, to give you a sense of the sort of polish that you might achieve on a track before you commit financially. Downloads are limited, and you'd need to subscribe before you could export the results as high-quality files. But even as it stands, this free version could be a useful preview tool — hearing a mastered version of a track can often

■ In the free version's Samples tab, you can search for and audition lots of samples spanning a huge range of genres.





■ The seven-day trial gives you access not only to more features, but also evaluation versions of various plug-ins.

planning. By preparing everything in advance, you can streamline the process for when you decide to go live.

This preparation step is a subtle but valuable feature, allowing musicians to organise releases, catalogue their work and familiarise themselves with the data entry process. Even in a free account, SOS For Artists encourages structured planning from the very start.

Testing the Seven-Day Trial

Once you've had a nose around the free account, and are ready to dive in a little deeper, you can sign up for the seven-day trial. This unlocks more access to the Essentials or Pro plan, so you can develop a greater understanding of how well our platform might work for you. In order to activate the plan, you will need to

highlight potential areas for improvement that had escaped your attention when mixing, particularly in terms of balance, EQ and dynamics.

The **Distribution** tab allows you to start building a release by uploading audio, adding artwork and filling in metadata. While you can't distribute tracks without a subscription, this feature is excellent for

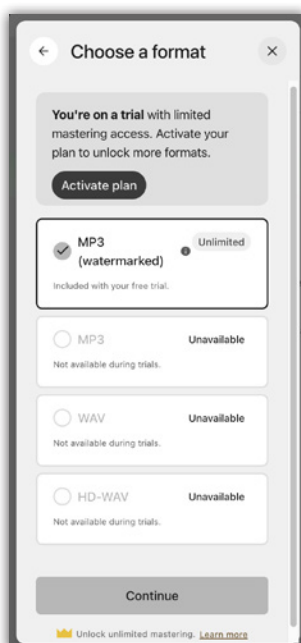
add payment card details — but you don't pay up front, and you can cancel before the trial expires should you wish.

Once signed up, there are still some limitations, but you can dig into each of the sections more deeply and play with more of the tools.

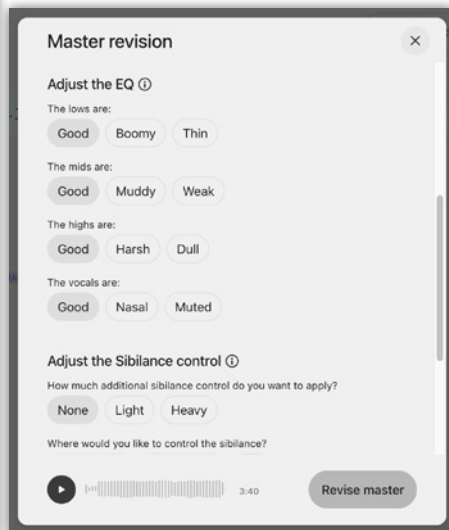
Starting in the Samples tab again, you can open up the search bar and type in "free". You'll be able to browse and download a range of samples to try out in your first project, from 166 tabs of one-shots and loops, including synth pads, guitars, synths, percussion and more. There's plenty here to suit all sorts of genres. In the Plugins tab, you can download, install and try out 22 of the available plug-ins for the duration of the trial. The choices include Synchro Arts' RePitch 2 Standard and VocAlign 6 Pro, LANDR Stems, and a range of Audified's plug-ins including MixChecker. Also included in this trial version is LANDR's sampler, which is a great way to organise and make good use of any samples you may have downloaded from the previous tab.

In the Mastering tab, you now have the option of downloading an unlimited number of watermarked MP3 versions of your tracks — this means you can test them in various settings, for example putting them in playlists alongside commercial tracks, which again can be a great way to get a new perspective on your mix, or get feedback from bandmates and so on. Some other features in this tab let you revise your master before downloading too. For example, having heard the automated result, you can adjust the EQ settings, widen or narrow the stereo field and introduce some sibilance control if needed. This trial version gives you 4GB of storage (there's a whopping 2TB on the full plan, once the trial has ended), so you have plenty of room to run a few tests and finalise your mixes before committing.

Once your masters are finalised, you can even set a release in motion as we described in much more detail last month — the option to fully release your tracks is included in the trial! ■■■



■ The trial also extends the options available in the online mastering service, including the ability to render a watermarked MP3 version to play on other systems.



Free & Trial Versions' Key Features

The new free version of SOS For Artists has some limitations compared with a subscription, but it still offers lots of tangible benefits:

- You can experiment with loops and sound design for free.
- Plug-in demos can help you understand their potential in your projects.
- You can hear mastered versions of your tracks, and glean insights into your mixes.
- You can prepare your release metadata and artwork before subscribing.

If you want more than that but aren't yet ready to commit to a sub, you can check out the trial version. This lasts seven days, and offers deeper access to the platform's features: with included samples, trial plug-ins and the option to release your tracks, you have the tools you need to get your first track created and launched.

JOHN WALDEN

When you're mixing, there are lots of processing options that can make a significant contribution to your acoustic drum sound, but dynamics processing can be particularly useful when the musical style or nature of the mix mean the drums must make a larger-than-life impact. Such processing might come in the form of gating, expansion, compression, limiting, transient shaping, clipping and even as a side-effect of tape emulations, amp/speaker sims, saturation and distortion.

So what, on this front, does Cubase's stock plug-in collection have to offer? With the aid of some audio examples (available on the SOS website: <https://sosm.ag/cubase-1225>), we'll explore all that in this article. It's a big subject, though, so to keep things focused and ensure that all the techniques I cover below can be achieved in any Cubase versions from Elements upwards, I'll concentrate mostly on compression, transient shaping and saturation.

And I'll add one final note before we dive in: remember that the multitude of options at your disposal always includes doing nothing! Your first job is always to listen carefully, and if your drums were recorded well and sound great, then just balancing the faders and allowing the glorious natural dynamics of the kit and performance to be the star of the show may well be enough, particularly if you have any bus processing going on. It's not an unrealistic prospect, if you're happy to live without winning an award for crafting the loudest mix in history!

Drums, Naturally

Normally, though, you'll want at least to reign in some aspects of the performance/recording just a little, even if you don't want to overtly change the energy or character of the part. For such duties, you don't need to look for anything special: Cubase's standard Compressor plug-in is a transparent and very flexible tool, and well up to the job. An instance, or possibly two in series, placed on your drum subgroup might well be all that's required.

The first screen shows a setup with two Compressor plug-ins, each instance having different attack and release times so they react differently. The first is intended just to catch the highest peaks in the signal. Its threshold has

Add impact and attitude to your drum parts with Cubase's stock dynamics plug-ins!



With two instances of Compressor on your drum bus, you can both tame transients and add some extra punch, without losing the natural sound of the kit.

been adjusted so that compression is triggered only by the loudest section of a hit: the initial transient. But, the very fast attack and release times mean that as soon as the transient has passed through (and thus the signal level drops below the threshold), the compression ceases quickly — the gain reduction meter should almost just 'flicker' to indicate this short-lived processing.

The ratio you can set to taste, and you can juggle the ratio and threshold controls to finesse the amount and character of gain reduction this compressor gives you. But note that generally you'll want more assertive settings than the (roughly) 2:1 you might opt for on a mix bus; I used 5:1 in this example. Incidentally, if you crank the ratio up to 8:1 (the maximum ratio the Compressor plug-in offers), it starts to become something like limiting, and if that's what you're after, you could easily use an instance of Cubase's Limiter here instead. Personally, I like having the additional control offered by Compressor. Once configured, and you've noted the maximum gain reduction (around 6dB in

this example), you can then apply a little make-up gain (I used a fairly conservative 3dB here). Because we're pulling down the peaks, the end result is a sound with less overall dynamic range, but with the sustained portion of each hit (where Compressor has fully released) sounding a little louder (because of the make-up gain). Judge it right, and the result should be more impactful drums.

The second Compressor instance is optional. The first Compressor has already handled the highest peaks, so this one targets a different aspect of the drum sound. A slower attack time allows the transients to pass through unscathed before the gain reduction kicks in, and the combination of a longer release and a lower ratio (3:1) than on the first instance results in smoother-sounding compression. Again, the threshold can be adjusted to refine the overall result, and if required (it depends what you want to achieve here) a little make-up gain can be applied to restore the final level of the drum mix. As with any compressor, the various controls interact, but unless

you push specific controls to an extreme, it's relatively easy to retain a very natural sound, and the gentle compression of the kit's overall dynamic range makes it easier to integrate into your overall mix. Compressor might not have the vibe of some more esoteric dynamics plug-ins, but it is capable, flexible and transparent.

More Control

Sometimes, you're going to want more detailed control over the dynamics, and if that's the case then you might want to consider compression of the drum kit's close-mic channels. If a transparent result remains your aim, then the same dual Compressor configuration we used for the drum bus is a perfectly suitable starting point for individual kit pieces. But you'll want to tailor the settings to suit the dynamics of each drum element, of course. This is particularly useful for the key kick, snare and hi-hat or spot-cymbal mics.

Depending on how you're approaching mixing your drums, you might want to configure this compression with any drum bus dynamics processing you have in place bypassed — in changing the peaks and levels of the individual mic channels, you'll change how the bus compressor reacts. Note, though, that each mic will have captured spill from other kit pieces, so if you wish to do any of this work with your drum element soloed to better hear what the compressor is doing to the track, you'll still need to listen to the processing in the context of the whole kit — particularly overheads and room mics, with which all the close mics will interact.

Once the individual drum dynamics have been tweaked, you can re-engage the bus-level processing, which might need small adjustments. For example, if your channel-level processing is now catching the initial transients, you may be able to back off the same processing on the drum bus. Or if your kick and snare are now louder because of make-up gain, they might be tripping the bus compressor more, and you might either want to back off the threshold a little, or use a side-chain filter to reduce the compressor's sensitivity to the kick's low frequencies to avoid pumping.

Targeting individual mic channels also lets you explore additional options on a specific drum, without worrying too much about undue impacts elsewhere in the kit. Yes, as I said, there's often spill and with it a risk of changing phase relationships other mics, but the risk of

significant problems is less than with radical EQ. One of those additional options might include replacing Compressor with one of Cubase's more characterful compressors or other dynamics processors. The bundled compressor options depend on the Cubase version you're using, but there are plenty of free alternatives if you need one, such as Klanghelm's DC1A and MJUC jr. In this example, I've gone for two additional (and I suspect often overlooked) options that are available in all versions from Elements upwards: EnvelopeShaper and DaTube.

Throwing Shapes

In the second screenshot, I've placed an instance of EnvelopeShaper after my dual Compressor configuration (you can also use this without the separate compression, by the way!). This allows you to shape the transient and sustained (release) portions of a typical drum hit independently and in some interesting ways.

The Length control sets the duration of the 'attack', so determines how much of the drum's initial transient is influenced by gain applied using the attack knob. You can subdue the transients by choosing a negative attack value, or emphasise them with a positive one. The latter can add extra 'click' to a kick or initial

'snap' to a snare for better definition, for example. But be careful — you're likely boosting the peaks, which could lead to the signal either overloading any plug-ins you have downstream, or peeping above the threshold of your bus compressors, making them react differently.

The release control then applies a gain adjustment to the 'sustain' (non-transient) portion of the drum sound. So positive values tend to make the 'ring' or 'body' of a kick, snare or tom more obvious, while negative values generally give you a tighter-sounding drum. Negative values will also tend to reduce any bleed from other drums, along with any room sound, rather like a gate can. There are more sophisticated transient shaping plug-ins out there, but do you really need them? I find EnvelopeShaper easy to use and very effective!

What A Character!

Some compressors (in hardware or plug-in formats) don't just turn the signal up and down — they can also add additional character to your drum sounds, with subtle transformer- and valve-like distortion that we might call 'saturation' or 'harmonics'. To emulate this, there are various options in the different versions of Cubase, including in the channel strip, »



At the channel level, it's pretty easy to achieve additional control over a specific drum sound — but always listen out for interactions with the rest of the kit and your bus processing.



» and plenty of third-party options including freebies such as those mentioned above. Since all Cubase versions come with DaTube, I've used that here.

As shown in the screenshot for my kick drum channel, I pushed the Drive control quite hard and then backed off the output to balance (to taste, by ear) the processed and unprocessed signal. In this case, DaTube's harmonics add a lovely warmth to the kick tone, and result in a more solid 'thump'.

Two Busses At Once

Another technique that's well worth exploring for acoustic drums is parallel compression. Some plug-ins let you do this internally — that's what a wet/dry mix control gives you — but there are often advantages to adopting the 'old-school' approach of putting a duplicate signal through a compressor on a separate track,

and I've done that here using a send to a dedicated effects track. This allows you, for instance, to EQ the 'wet' part of the signal different from the dry one, which can be helpful.

In fact, as you can see in the final screenshot, I chose to create two such effects busses for this project. The first receives sends from just the drum shell channels (the kick, snare and toms), while the second gets the hi-hat, overheads and room mics. The purpose of this separation is to allow you to be more assertive with the close-miked drum-shell processing than with the metalwork, which can start to sound weird if overcompressed, or when the longer sustains are compressed in response to loud close-mic hits.

There are plenty of creative options that you might explore on the 'shells only' bus, but I've kept it relatively simple in this case with an instance of the

■ If you want to move beyond a natural sound, parallel processing of the kick, snare and tom mics can be used to dial in some additional grit without making your cymbals suffer.

Tube Compressor pretty much 'slammed', followed by the Distortion plug-in to add extra grit. As demonstrated in the audio examples, blending just a tiny amount of the resulting signal into the overall drum mix injects a much more aggressive character to the overall kit sound. But you can easily dial in something super-grungy if you really want to. Importantly, the cymbals, having not been subject to this OTT processing, retain their definition — they sound neither 'splashy' nor too harsh.

Bypass Plug-ins, Not Taste

So whatever version of Cubase you have, you have plenty of scope to experiment. That said, it's fairly easy to take your perfectly decent acoustic drum recording from 'natural and dynamic' to 'flat and overcooked', so make sure you avoid the potential for a taste bypass by regularly bypassing your dynamics plug-ins for an A/B reality check. Side-step that possibility and, with a combination of taming your transients, reduced dynamic range, and possibly a little colourful saturation, you can increase the overall level, and dial in some attitude, so that your acoustic (or multi-out virtual instrument) acoustic drum mix delivers just what the song requires. ■■■



■ Klanghelm's MJUC Jr — one of many great-sounding free 'character' compressor plug-ins you can check out if your version of Cubase has fewer options.

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Master Faders offer much more than just level control!

JULIAN RODGERS

Master Faders in Pro Tools aren't created by default in a session, and things seem to work fine without them, so you might wonder why they exist at all. The reason is that they solve problems. They monitor and trim signal paths, and their inserts and metering work slightly different to those in other track types. This is great for tackling that often talked about and sometimes misunderstood aspect of audio production: gain management.

You might never move the fader that gives a Master Fader its name, but having one across, for example, the output bus of your session, is very useful. The meter in a Master Fader is often more important than the fader itself, and can be reason enough on its own to use Master Faders throughout your session.

Meter Reading

If you see that you're not managing headroom effectively, the Master Fader is all you need to correct the situation without disturbing your existing automation and balances. Master Faders control paths, whether outputs or internal busses. They consume no additional system resources and are available for all track formats. While many people only think of placing them across the main output, in practice you'll often find uses for them across stems, submixes, and even headphone sends.

The scaling of the faders in Pro Tools does mean it's easier to make fine adjustments towards the top of the range, but beyond that, there's no sonic change created by adjusting levels using a Master Fader, unless you deliberately introduce non-linearities by using analogue-modelled plug-ins. For this reason, some longstanding rules of thumb from the analogue world don't apply inside Pro Tools, and Master Faders provide a good example.

I think a Master Fader is best thought of as a meter with an associated trim control. It targets a specific output or bus path, and while they can be used to control the level of multiple tracks together, there are other, often better ways to do that. The meter shows the level of that output or bus path at the summing point in the Pro Tools mixer.



Avid's Pro Limiter has loudness and true peak metering, and is best placed on a Master Fader where the inserts are post-fader.

Meters on Master Faders always show the post-fader level, so any adjustment you make with the fader is reflected regardless of other metering settings. You can even choose a different metering type for Master Faders from the rest of your tracks — a valuable option when you want finer resolution for monitoring headroom.

There are only five insert slots on a Master Fader, but as they operate differently to inserts on tracks, you're unlikely to need more than five. More on this difference later.

Level Management

As mentioned, Master Faders are one way to control the level of groups of tracks in Pro Tools, but there are others and it's important to understand how they differ:

- **VCA faders** are the simplest way. They act like a remote control for the channel faders of tracks in a Mix Group, and don't pass audio.
- **Auxiliary inputs** and **routing folders** sum the audio outputs of the tracks feeding them. They're often used for group processing, much like a subgroup on an analogue console.
- **Master Faders** don't have audio inputs at all. They directly control the level of a path.

Because of this, Master Faders complement rather than replace VCAs or auxes. You might use a VCA for automation moves, an aux for bus compression, and a Master Fader for accurate post-fader metering and headroom management. Thinking of

them as different tools for different jobs is a good way to avoid overlap or misuse.

So why bother with Master Faders at all? For one, they offer fast headroom management.

A Master Fader lets you pull back a path that's clipping in a single move. Whether it's your main mix, a cue send, an effects path or a stem output, it's far quicker than adjusting multiple faders, which can upset balances, automation or timbre. Trimming the overall level at the Master Fader avoids these issues. This makes Master Faders the quickest, cleanest way to apply a global fix.

They also offer accurate path metering: because the meters are post-fader, you always see the level actually hitting the bus or hardware output. This makes them invaluable for preventing clipping.

Certain types of plug-in are also best placed on a Master Fader — loudness meters and dither plug-ins, for example. This is because Master Fader inserts sit post-fader, where they receive the true signal leaving the system. By

Master Faders & EuCon

If you use a EuCon control surface such as the Avid Dock, there's a neat shortcut for tying Master Faders into your monitoring workflow. Right-click any Master Fader and assign it as the EuCon Monitor. This lets you control that fader from a hardware pot on your EuCon surface — for example, the Monitor knob on the Avid Dock. Because it's so quick to assign, you can easily switch the monitor control to a different Master Fader when needed, and switch it back just as fast.



■ In this screenshot, the mix bus processing is carried out on an aux input fed by an internal bus (labelled Mix Bus), which in turn is routed to Out 1-2 and controlled by a Master Fader.

contrast, bus compressors are better off sitting on auxes, where their behaviour won't be affected by fader moves. One classic mistake, though, is to place a bus compressor on a Master Fader and then attempt a fade-out. The compressor threshold shifts relative to the signal as the fader drops, and the sound changes in undesirable ways.

Master Faders aren't just for the main output. Place them on stems, effect sends, or cue mixes, and you instantly gain reliable metering and level control.

Faders & Auxes

A very common workflow is to combine an auxiliary input (or routing folder) with a Master Fader, using each for what it does best.

Rather than routing everything to the main output, create an internal bus named Mix Bus. Instantiate an aux input called Mixbus and use this for bus compression or analogue-modelled plug-ins. Route this aux to the main output, then create a Master Fader on that output.

This gives you a pre-fader insert point for mix processing and a post-fader insert point for utilities, along with accurate metering and level trim at the end of the chain. The same approach works well with stems. For

processing on the aux, and reliable metering and level control on the master.

VCAs can also be used alongside this setup for fader control, but bear in mind that VCA metering behaves differently and doesn't provide the same accurate sum as a Master Fader.

Max Headroom

Because you can use different metering types on tracks and Master Faders, you can keep something like a VU-style meter on source tracks to reflect musical balance, but set your Master Fader to Linear for detailed peak resolution at the top of the scale.

When it comes to level targets, it's important to remember that 24-bit converters offer generous headroom.

example, a dialogue stem might be routed to an aux for EQ or dynamics, then to a dedicated Master Fader for headroom management and metering before printing.

If you're working with stems — common in post-production for dialogue, music, and effects, but also useful in music sessions for groups such as drums or guitars — you can create Master Faders for each. These work just like a main output master, giving you post-fader metering and trim on the stem. Combined with an aux or routing folder for pre-fader inserts, this provides the same two-stage approach:

Unless you're driving analogue-modelled plug-ins or HEAT, there's no sonic benefit to running hot, but there is a real risk of clipping. When tracking, I aim for peaks averaging around -18dBFS, which leaves plenty of headroom. Also keep in mind the distinction between indicated clips and true peaks. For streaming delivery in particular, make sure your chosen limiter or meter warns you of true peaks, not just sample peaks.

Master Faders aren't as uninteresting as they might at first appear. They are path controllers with post-fader inserts and post-fader meters, and they achieve essential tasks in the cleanest, most effective way possible. If I were introducing them to someone new, I'd call them Master Meters. The meter is their most important feature, with the fader and post-fader inserts acting as tools attached to that meter. Wherever you have a bus or an output path, there may be a strong reason to display a master meter on it.

One alternative worth knowing is the hidden meter in the floating output window. On any track routed to a hardware output or internal bus, click the output window button next to the output selector. Expand it with the disclosure arrow at the bottom right, and you'll see a second meter showing the level of that bus or output. This is another excellent way to keep an eye on headroom, much like using a Master Fader. ■■■



■ The expanded output window (shown here on aux 4) features the same type of metering as a Master Fader.

ROBIN VINCENT

There are times when your best course of action is to record the guitar first and worry about the tone later. With the right equipment and some useful Studio One know-how, you can try all sorts of different ways of amplifying your guitar before you commit it to the mix. This is known as re-amping.

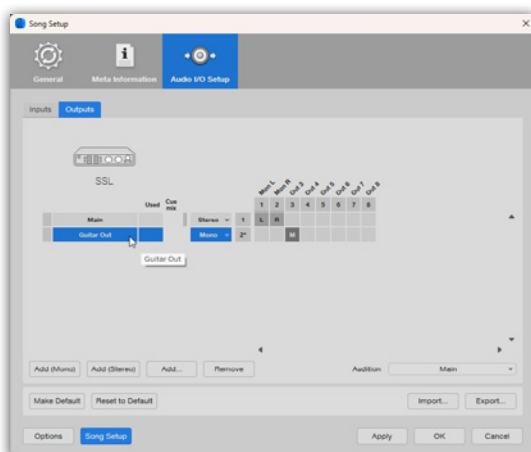
The basic idea is that you record your guitar completely dry by direct injection, and then route that track out of the DAW and into a guitar amp, which you mic up and record. Re-amping gives you a considerable amount of versatility and the opportunity to try different effects, tones, setups and configurations that perhaps weren't available at the time of recording. Studio One Professional includes a rather neat plug-in solution for accomplishing this, called Pipeline XT. We're also going to look at how to do it manually for those of you who are running the Artist version of Studio One, and don't have that plug-in in your box of tricks.

Before we turn to the computer, we need to make sure we have the right hardware in place to make re-amping possible, or at least not rubbish.

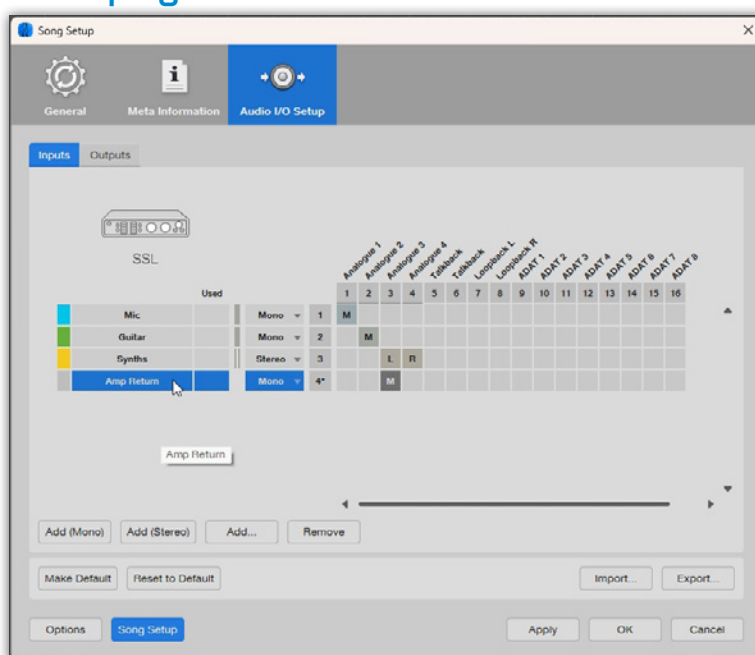
High & Dry

To record dry guitar, you will need an audio interface with a high-impedance mono unbalanced input. Most audio interfaces have this and it's usually marked as 'Hi-Z' or 'instrument'. Impedance is a complex and meandering concept that revolves around the total resistance and reactance of a circuit. Hugh Robjohns wrote an excellent article all about it in SOS January 2003 (www.soundonsound.com/techniques/understanding-impedance). All we really need to know is that electric guitars have a high-impedance output and would very much like a high-impedance input to plug into, or they tend to lose their high end and sustain. If your audio interface doesn't have a Hi-Z input, you'll need a DI (Direct Injection) box, which converts the guitar signal into a balanced, low-impedance signal that can plug into a standard microphone input.

To do the re-amping, your audio interface will need a spare output, and another input for capturing the sound of your amp. This loop has certain requirements to make it work properly. First of all, the dry guitar



We talk you through the ins and outs of re-amping in Studio One.



The Inputs tab of the Audio I/O Setup window. Here's where you assign a physical input on your audio interface for capturing your re-amped guitar sound.

signal you are sending out is emerging from a balanced line output on your audio interface. Your amp is expecting the mono, unbalanced, high-impedance output of a guitar, and so we need to convert the signal into something resembling that.

For the level disparity, you could simply reduce the output level for the channel routing to the amp. The problem with doing it digitally in the DAW is that you are effectively reducing the dynamic range of the signal, which can result in unwelcome noise. To change to a higher impedance and an unbalanced connection, you can use a passive DI box in reverse. However, by far the easiest method is to use a re-amping box. It takes the output from the interface, lowers its level, increases the impedance and makes it unbalanced. It usually has a ground-lift switch too, which can massively reduce the amount of noise and interference being generated.

You'll also need a spare output to send your clean DI signal out to your amp.

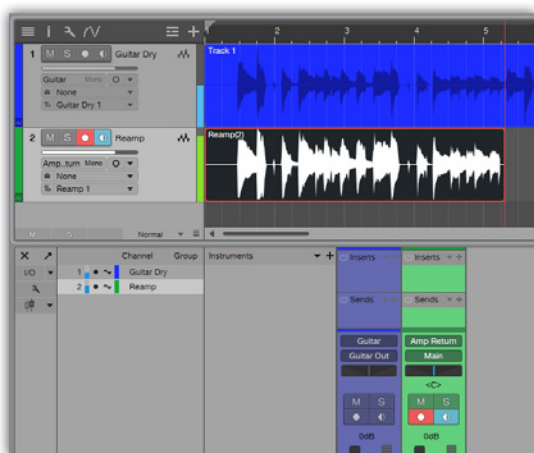
Re-amp boxes tend to be passive and rugged things that can handle themselves in a chaotic studio environment. Prices can vary from under £100 from manufacturers such as ART or Palmer, to nearer £300 for a Radial re-amp box. They all can achieve a similar outcome, but do check to make sure the one you choose has the features you need for your setup.

Now that you have a good signal for the amp, you have to consider how to capture it again. You could mic up the cabinet, but this requires you to fully embrace the loudness of your guitar rig. You might use an amp with a direct, speaker-simulated output, or you could connect your amp to a 'dummy load' box, which fools the amp into thinking it's connected to a cabinet and gives you a direct output suitable for recording directly through your audio interface. There's so much to think about with re-amping that one could be tempted to simply use the Ampire software amps built into Studio One.

However you choose to do it, the next step is sorting out the routing in Studio One. We'll look at the manual approach first.

Route Planner

Click on Song and select Song Setup from the menu. Click on the Audio I/O tab and you'll see the ins and outs of your audio interface. I have the SSL 12 from Solid State Logic, which gives me four analogue



Recording the re-amp. Here, the dry guitar signal has been assigned to the Guitar Out output, and a new mono audio track has been created, with its input set to Amp Return, in order to capture the re-amped guitar sound.

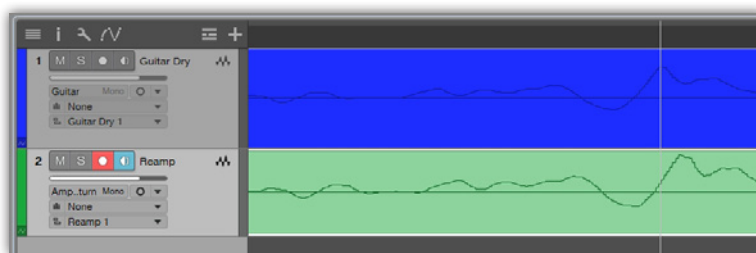
inputs. The first two can accept instrument signals, and I normally use input 2 for recording guitar. Inputs 3+4 usually carry a stereo mix of hardware synthesizers, but for re-amping I'm going to repurpose one as the return signal from the amp.

Next, I'm going to add a mono output and allocate it to one of my spare analogue outputs. If I label them usefully, then it's going to be really easy to set it up on my guitar track.

When it comes to recording your guitar dry you are, obviously, playing without any effects or tone that you would normally use during a performance. This might not be ideal, as you may want to be able to lean into some kind of tone to influence your playing. To get around this, you could drop the Ampire plug-in onto the track and monitor through that as you're recording. Alternatively, you could use the re-amp loop that's already in place to monitor your guitar through as you play. The setup is the same as doing it after the recording so follow on to see what that is.

This is easiest to see if you open up the Console. On the channel strip for your recorded dry guitar, you need to set the output to 'Guitar Out', or whatever you called the output you're using for the re-amp. You'll find this just above the panning control and below the input field. Create a new mono audio track and set its input to 'Amp Return' or whatever you've called the input for the amplified guitar signal. Record-enable the re-amp track and hit Record. It will then record the amplified signal to the track. You can do this multiple times with different amp settings until you find the exact tone you're after.

There's one little bit of housekeeping you might have to do after your recording. As you've sent your guitar track out of your audio interface and back in again, you will have accrued some latency. With a decent audio interface we're only talking a couple of milliseconds, but it might make your track feel draggy or a little less tight. There's no magic answer to this other than moving it yourself. So turn off Snap, zoom into the audio and find a peak to help you line it back up with the DI signal.



After you've recorded your re-amp, you may need to pull the recording back in time to compensate for the added latency.

Pipeline XT

Moving on to the plug-in that does it all for you, drag Pipeline XT onto your dry guitar track. (It doesn't really do it all for you; it's more that it's convenient, handles latency for you and lets you save the routings so you can use it every time without having to set it up again.)

Pipeline XT acts as a hardware insert, breaking the signal flow of the channel and routing the audio out of the computer and back in again. You still need to create the Guitar Out and Amp Return channels in the Audio I/O setup, and you'll still need a track to record the output to, so those elements are the same as doing it manually. However, this time we set up the routing in the plug-in rather than in the Console.

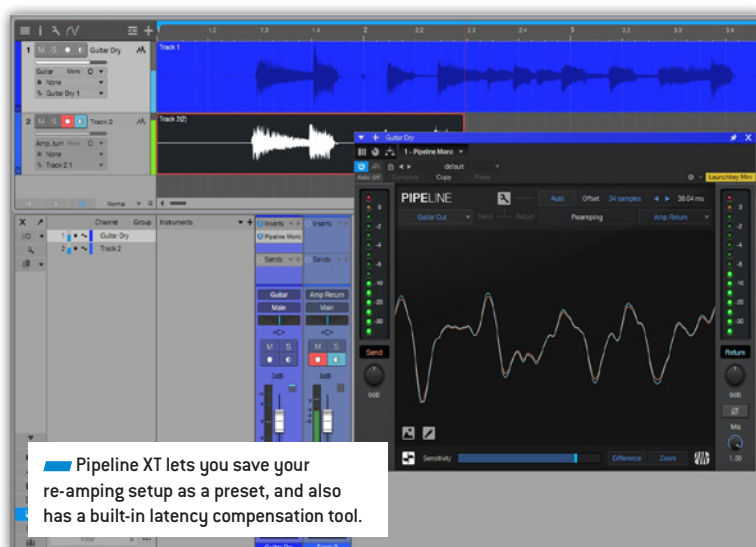
So, in the Pipeline plug-in, you set your output on the left and the returning input on the right. You can name the plug-in to remind you what you're doing

and even take a photo of your amp and stick it in the middle, which is a pretty genius idea if you're likely to forget settings.

Now we have an opportunity to deal with the latency issue. Click on the little spanner icon and Pipeline will ping itself with a short, sharp burst of audio and show the result on the display, where you'll see two traces. The orange one is the original output, and the blue one is the signal coming back in. You'll see that they are currently offset. Click on the Auto button and Pipeline will measure the latency and adjust so that the two signals are aligned. You can manually offset this as well if you like. There are controls over the level going out and coming back in, there's a wet/dry mix control for parallel processing, you can change the sensitivity of the display, zoom in/out, and also just see a trace of the difference between them.

All that's left to do is create a new mono audio track and set its input to the Amp Return path we set up earlier. Click Record, and you will capture the guitar going through your amp.

Pipeline XT is also great for setting up effects loops through hardware, and you can save every configuration as a preset so that you can come back and use the same setup again. ■■■



Pipeline XT lets you save your re-amping setup as a preset, and also has a built-in latency compensation tool.

With Flashback Capture, you need never lose a magical take again!

SIMON PATERSON

Like in *Citizen Kane* or *Pulp Fiction*, interesting things happen when a story is subjected to a flashback. Imagine you've played something brilliant, but were not in record. Wouldn't it be useful to step back in time and capture those inspired moments? Echoic memory is all very well and good — that aural image in our short-term memory that might help us remember what we just played — but all too often, the more we grasp at the musical idea that once was, the more we're aware that that particular moment of magic, that unique combination of phrasing, dynamics, note choice and intonation/inflection, has passed. Could Flashback Capture be the answer?

Sliding DAWs

Logic Pro's Capture Recording feature was updated in version 11.2, and renamed to Flashback Capture. The original Capture Recording function wasn't particularly prominent, but was incredibly useful nonetheless. Indeed, whenever I showed it to students in workshops, even in its original incarnation, it would always elicit audible gasps of amazement — perhaps more than any other feature.

It's a function I use every time I record myself. I detest the red light and the tyranny of the ticking metronome. It unnerves me. I much prefer just happily jamming along until I feel I'm getting



Screen 1: The Flashback Capture button lives next to the usual Record button, but will be greyed out until there's something to recover from the Flashback buffer.

somewhere. And if I haven't yet played well enough, it doesn't matter, I'll just go again. But once I've come close enough to getting the part right, I simply press Shift+R and... hey presto! The way Logic instantly and seamlessly reveals a fully intact region containing all those nuggets of nice notes does seem like magic. And not a single red light has reared its beastly head.

You Don't Have To Put On The Red Light

As a producer, I'm constantly striving to capture that elusive musical magic, to trap that lightning in a bottle. When working with any artist, I tend to record absolutely everything. I leave the tape running, as it were, even whilst setting levels and warming up. Why? Because unique moments of magic can and very often do happen when an artist performs for the first time. In these precious moments, they're not necessarily focused on attaining a perfect technique. They might be looser, more inventive, more daring, more instinctive.

The trouble is, something shifts in our brains when the red light comes on and we shift into 'record mode'. Is it the fear of being chronicled for posterity; our truths, our stories, our ephemeral outputs made concrete? A permanent record of our

disk needn't be a scary thing. Like any number of visual and aural illusions that consistently fool us, even when we know how they manipulate us or, like setting clocks a little fast so we leave the house earlier (again, we know it's a ruse!), we can use Flashback Capture as that one weird trick to reveal recordings of performances made free from fear and prejudice. After all, so many great moments in musical history have occurred when artists didn't know the tape was running. We would never have known the beauty of someone merely tuning their guitar if Teo Marcero hadn't recorded John McLaughlin doing exactly that at the beginning of Miles Davis' beautiful 'In A Silent Way'.

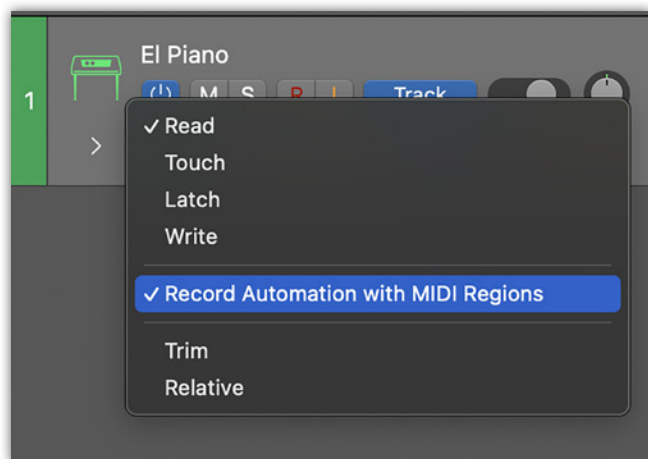
Flashback To The Future

Here's how it works. Since the 11.2 update, you may have noticed a new extra button next to Record in your toolbar. This is the Flashback Capture button (see Screen 1). If it's not there, click the Display Mode pop-up menu (the triangle to the right of the LCD) and select Customise Control Bar and Display. Towards the bottom of the second column, tick Flashback Capture. It will appear greyed out until there's something in the buffer to render, at which point it will become solid.

Like Big Brother, Logic Pro is always listening (but, thankfully, only to your musical ideas). Assuming you have a track header selected and record-enabled, and Logic is playing (Flashback Capture doesn't work while Logic is stopped), it's always recording into a buffer. And you can tap into this recording repository to bring out any snippets of audio or MIDI you played whilst you were warming up and setting your levels.

Audio Vs MIDI

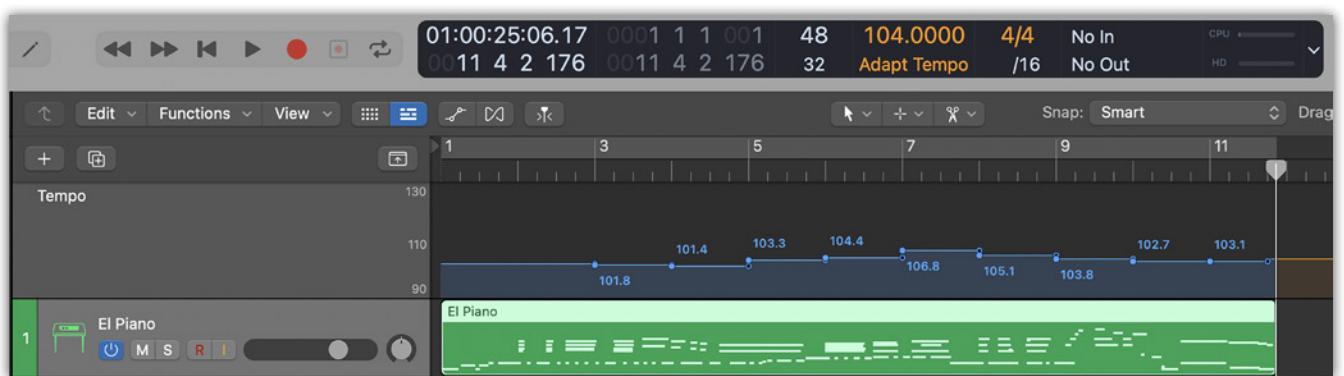
The behaviour for MIDI and audio is very similar, but there are crucial differences. As long as MIDI is being received (check the MIDI Indicator in the LCD), and tracks are record-enabled (this is the default for all software instrument tracks), it will all be recorded into the Flashback buffer.



Screen 2: As well as audio and MIDI takes, Flashback Capture can store your automation, provided this option is ticked in the Automation Mode menu.

skills, our technique, our choices, our taste? Is our entire legacy at stake? Argh! French soprano Régine Crespin once said, "The microphone waits, unpitying, insensitive, and ultra-sensitive at the same time, and when it speaks, it's to repeat everything you've said word for word. The beast." I can definitely relate.

But committing musical ideas to hard



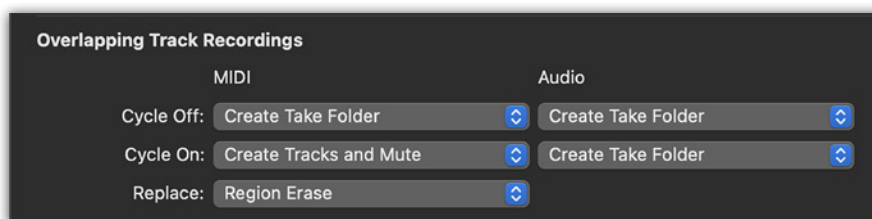
Screen 3: Unlike with audio, MIDI Flashback Capture data is created even when your project isn't playing, so your unrecorded MIDI noodles can be recovered. You can even combine this with Logic's Adapt Tempo feature to create a tempo map around your Flashback Captured performances.

But there's more! Unlike with audio, MIDI Flashback Capture works whether the project is playing or not. This means you can engage Flashback Capture to render a new region from whatever MIDI data has been received since the last playback. This not only includes MIDI CC data, it also records automation if Record Automation with MIDI Regions is selected (click-hold on the Automation Mode pop-up menu) — even in Read mode (see Screen 2). Combine this with Adapt Tempo (see Paul White's Logic Pro workshop from SOS September 2025), and Logic will have worked out the tempo for you (Screen 3). This saves you from being tied to the metronome — a genuinely useful benefit at the start of any new composition session. Indeed, this is now all set up as such in my current default project template.

To use Flashback Capture with audio, ensure you've set the correct input and record-enabled your audio track. Note that Flashback Capture will only work if the project has been playing for at least four seconds, and then only on a single audio track, no matter how many tracks are record-enabled.

Recording Settings

Before using Flashback Capture, make sure you're happy with the settings for overlapping recordings, as these will also apply to Flashback Capture.



Screen 4: It's worth making sure your overlapping track options are set up to your liking. Logic's default behaviour is to always merge your MIDI recordings, which may not be what you want!

In Logic / Settings / Recording (these are the global preferences, not project settings), you'll find options that cover what happens when you record multiple takes on the same track, with options for recording with the cycle on or off, and for both audio or MIDI.

For MIDI, the default setting will, inexplicably, always merge your MIDI recordings together. In my view, it's much better to have Logic record individual takes either on automatically created tracks (Create Tracks and Mute) or in a take folder. I tend to prefer recording audio into a take folder as it cycles round, because quick-swipe comping is so intuitive and effective, but I set my MIDI tracks to Create Tracks and Mute because MIDI take folders don't work in quite the same way. If I wanted to select whole takes, then a MIDI take folder is great, but I'm a bass player not a pianist, so I'll inevitably need to grab either the marquee or the scissors tool (both permanently set up as my Command-click and right-click tools respectively) and construct my own composite takes from MIDI regions. Screen 4 shows my options.

Otherwise, it's that simple. Press Play, then play or sing and as long as Logic has detected audio or MIDI (obviously, it won't work with silence), and the project has been playing for at least four seconds, it will always be recording. If something

magical happens, press Shift+R, click-hold on the Record button in the transport and select Flashback Capture, or press the dedicated Flashback Capture button in the toolbar.

Rolling With The Punch-ins

I now use Flashback in lieu of punch in/out, especially when I'm recording myself. Autopunch, which automatically drops in and out of recording over a predefined section, is wonderful, but crucially, any audio either side of the punch-in zone is not recorded. So, if you've played a note slightly early or not quite contained the end of your performance within the boundary, there's no chance of recovering it. However, I've discovered that this is never an issue if you use Flashback Capture.

Here's the trick. Let's say you're patching up a performance which has already been recorded into a take folder, and there's a four-bar section you'd like to play again. I'd start by duplicating the comp as it is, because that will make inserting the new section much neater in the long run. To do this, click on the triangle at the top of the comp region, and select Duplicate Comp from the drop-down menu. So now you have two identical comps, Comp A and Comp B. Whilst in Comp B, remove the four-bar section you want to revisit, spin the playhead back a few bars, and press Play. Once you're happy with a take, render that performance as a region using Flashback Capture. Don't worry that your new take has just ruined the comp — that's exactly why we duplicated it. Simply revert back to Comp A, then add the new four-bar section using quick swipe.

Now, when's Flashforward Capture going to drop? **///**

Q How do I stop my USB mixer feeding back to my DAW?

I'm an experienced computer user, but have relatively little experience using Windows with a DAW. I had a configuration that was working well a year or so ago, but recently started using it again and now have a problem that has defeated me! I use a Windows 10 (version 22H2) computer with a Mackie ProFX6v3 USB mixer, with Reaper as my DAW. I have the latest Mackie driver installed and the latest Reaper version. The problem is that everything going to the Mackie over USB (on channels 3+4) is coming back on the computer input channels 1+2, whether it's the metronome or playback of other tracks. Even general computer audio (also set to use ProFX channels 3+4) shows up as an input. Input on channels 1+2 is working as it should, but intermixed with the unwanted output signals. So far I have tried:

- Checking and double checking the driver settings and routing matrix;
- Uninstalling and reinstalling the latest Mackie driver, Reaper version, and installing all driver updates from Windows Update;
- Swapping the Mackie for a small Behringer mixer that uses completely different drivers (not ASIO); and
- Using Audacity instead of Reaper.



While the Mackie ProFX mixers can receive multiple USB channels from your DAW, only USB 1+2 are fed direct to the control room (ie. monitor and headphone outputs). The others will, like any other source, be fed to the main input channels and thus the main mix, from which your computer/DAW takes its stereo input.

The issue persists through all of this, which leads me to believe that it is something within Windows. I've gone into the advanced sound settings and reset everything to the defaults. Curiously the 'normal' audio on the computer works fine: output to Mackie channels 3+4, and input from the webcam's microphone. Can you suggest what on Earth might be going on?

SOS Forum post

Hugh Robjohns, Technical Editor

I'd recommend carefully reading the manual for the mixer again. I think that the monitoring outputs from your

computer/DAW software need to be routed through the USB 1+2 outputs — these feed only the mixer's control room outs and headphones, and not the mixer's main stereo mix bus. In contrast, the USB 3+4 outputs from the computer that you say you're using appear on one of the mixer's stereo channels, and that gets mixed with everything else, before appearing on the mixer's stereo bus — which you're then recording via the USB 1+2 inputs to the computer. In other words, while I understand the head-scratching, it's a DAW output configuration problem. Thankfully, it's easily solved at no cost!

“Your software needs to be routed through the USB 1+2 outputs — these feed only the control room outs and headphones, not the mixer's main stereo mix bus.”

Q If I swap from Windows to Mac, can I still open my old Reaper projects?

I've been using Reaper on Windows for years, but I need a new computer and I'm thinking of getting a Mac. I have a pair of external drives for all my projects. Will Reaper on Mac be able to open them from these drives as they are, or do I need to tweak anything?

Pete Taylor via email

Matt Houghton, Reviews Editor

I use Reaper on both platforms. The DAW won't have any problem reading and opening your projects,

but might not be able to run them. First, Reaper will load all its own plug-ins just fine and many third-party ones too (assuming you've installed them on the new machine), but there may be a few of the latter that won't play ball. To minimise the chances of disruption here, ensure you install the VST versions of your plug-ins, not just the AU ones (a Mac format that Reaper can host alongside VST2.x, VST3 and its own formats). You might find that a few sample-based instruments need reminding where their samples are, but that's easy to put right.

Perhaps the biggest issue is that while Macs can all read the data on NTFS drives (this is Windows' default drive format), unless

you install special software they won't be able to write to those drives. So if your external project drives are NTFS types, you need to copy the files to a drive Macs can use. The macOS default drive type is Apple File System (APFS), which is no good for Windows. But if you format your drives as ExFAT, both Mac and Windows operating systems will be able to read and write to them. I'd suggest obtaining another drive; they're not crippling expensive. Format it as ExFAT, and copy your projects to it. You probably know how to format drives on your current system, but on Mac you can use the Disk Utility app in Applications / Utilities. Choose Erase and select the ExFAT option (note that this wipes any existing data on the drive; you can't just convert your existing drives while keeping the data on them!). If an old drive is big enough, you can put it to very good use as a Time Machine backup drive. **///**

“If you format drives as ExFAT, then both Mac and Windows operating systems can read and write to them.”

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SOS WHY I LOVE... FOUND SOUNDS

BENJAMIN MCCULLOCH

I think it's important to have open ears when you're a music-maker. I like to listen to everything, because it's all got musical potential in some form or other. I feel like the sounds of nature, a bustling city, or even devices such as dishwashers and cars are just as interesting as acoustic or electronic instruments. I wonder if other musicians feel the same?

We often focus on craft when we're producing music. It requires a huge time investment to acquire the skill to play an instrument well. It also takes time to develop our own style and taste. Songwriting is a craft that needs constant effort too, and engineers and producers need to learn to get results from equipment that often resembles NASA mission

control... and then you also need to develop unique social skills to work with performers. When you consider all the craft that needs to go into music-making, it might seem that simply recording things like birdsong, waves on the beach or cicadas is cheating. We might say "Anyone can do that!", and that's partially true. Anyone can do it — so long as they have the motivation to go and find the thing, the luck to capture it (the world doesn't always perform on cue), the skill to record it well, and then the technique and imagination to turn everyday sounds into music that people want to hear.

When trying to record these sounds you'll often feel like a surfer waiting for the perfect wave. You're ready, but you have to be patient and wait for nature to get ready too. It can be relaxing if

you're somewhere nice, or it can be incredibly frustrating if you're not in record when the sound appears. Sometimes it's dangerous! You have to keep your wits about you when recording wild animals. Is it worth the effort? I think so. Listen to 'Blackbird' on the Beatles' white album. The song would still be great without the bird recording, but for me the whole thing comes across so much better because Paul McCartney's performance features a guest spot from a blackbird! It's beautiful and unique.

When you dig into music production techniques, you realise that many musicians were inspired by sounds that weren't from music. Loads of Björk's albums include found sounds and natural recordings, but a great example is *Vespertine* — listen to it and you will feel like you're in an Icelandic winter landscape with ice cracking under your feet, because the producers really recorded cracking ice for the songs. Billie Eilish's producer Finneas O'Connell

is known for recording interesting sounds and working them into his tracks. For example, in the chorus of 'Bad Guy' you can hear the sound of a pedestrian crossing recorded in Sydney. It's a prominent part of the rhythm! Who else would have used that sound in that way? That's why I find it so exciting. We can remix the world in our own unique style.

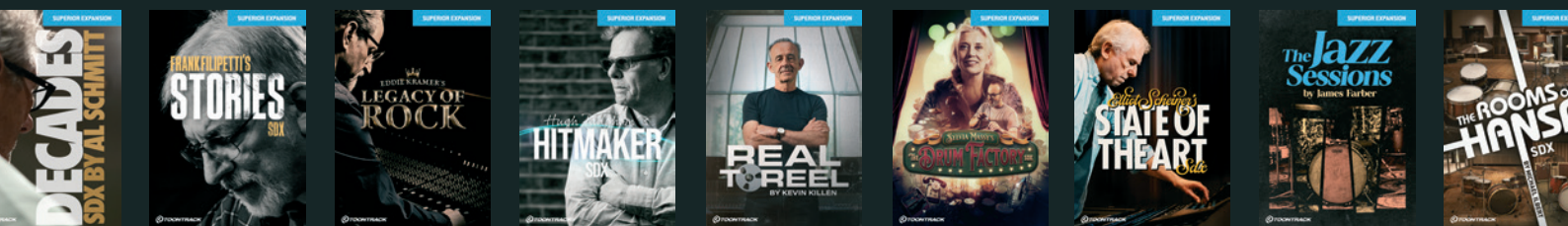
I've been recording a forest near Prague for my music recently, because it's full of amazing sounds. Why don't you try recording found sounds too? You'll get the best results with a high-end portable recorder and good mics, but you don't need that yet. Just put your phone into record mode for a moment, go outside, and try to capture anything that interests you. What do you hear that inspires you to make music? Will you include the sounds you record, or just use them as inspiration? I promise nobody else will make the same choices as you — there's just too much sonic variety out there! **///**

"When trying to record these sounds you'll often feel like a surfer waiting for the perfect wave."



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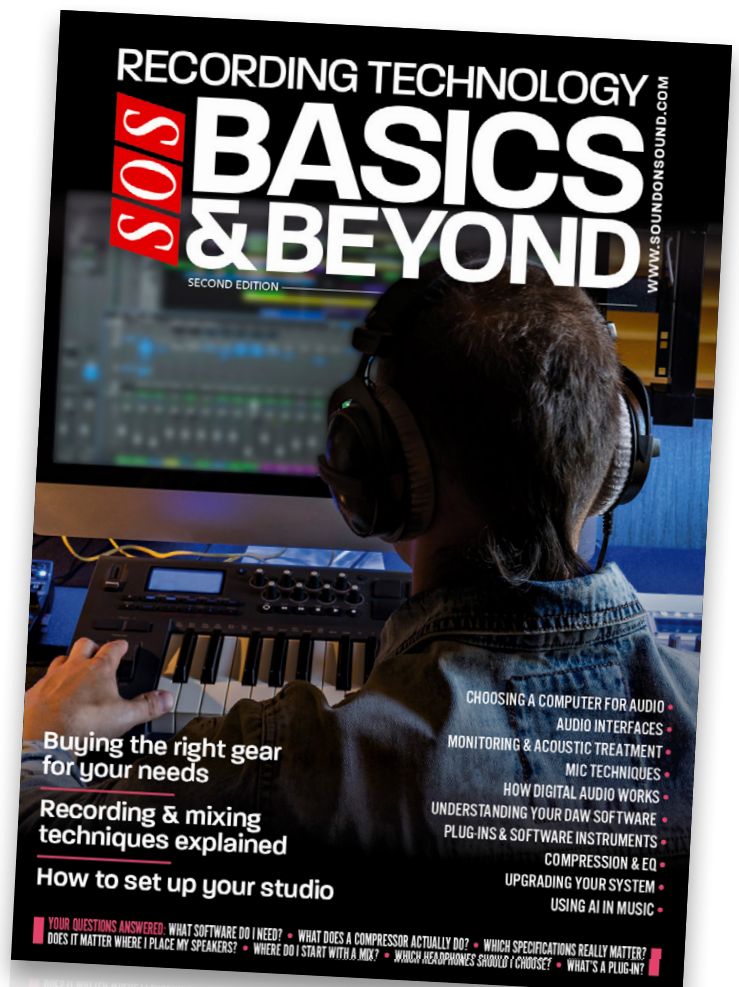
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SOUND ON SOUND

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