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BACK TO BASICS

When it comes to dynamics processing, we've never had it so good. Once upon a time, even the largest studios had only basic compressors, noise gates and perhaps de-essers. You were lucky if a hardware compressor offered fully variable attack and release times: luckier still if its actual behaviour bore any resemblance to the numbers printed on the front. Today, by contrast, our plug-in folders bristle with brickwall limiters, multiband compressors, dynamic EQs, resonance suppressors, transient shapers, parallel compressors, upwards compressors and countless other options. But how much better are our mixes?

Now I'd be the last person to discourage readers from trying and buying new things. New equipment is the lifeblood of this magazine. It can inspire us, help us to get results faster, or just push us in different directions, and that's all great. But the key to getting the best from it is to have a clear understanding of the basics. As the engineers of the past well knew, a straightforward compressor with the regulation threshold, ratio, attack and release controls is actually an incredible versatile device.

Even now, understanding how to get the best from a generic compressor can get us out of some pretty big holes. In our studios we can cheerfully fire up the latest CLA-endorsed drum smashalizer or vocal smarmalator without bothering to find out what it actually does — but what will we do when the band asks if we're available to mix front of house at short notice? And when none of the 'bass guitar' presets seem to work on the actual bass guitar in the project we're mixing, how will we know what to change to make it right?

Sound On Sound has published a few beginners' guides to compression over the years, by experts such as Paul White, Hugh Robjohns and Mike Senior. All are available as part of the vast treasure trove of articles on the *SOS* website, and the great thing is that each of these authors brings a new perspective on a familiar topic. But with compression being such a basic technique in modern mixing, I don't think you can have too many explanations, and this month I've tried to take a different approach to unravelling its mysteries.

One of the main barriers to learning in any field is jargon, and music technology is no exception. Standard compression parameters such as Threshold, Knee and Ratio can seem needlessly obscure to newcomers. My hope is that by providing a clear and concise explanation of what these actually do, we can help people overcome that barrier and begin to use compressors with confidence. After all, it's only once we have a clear idea of what's possible with these basic controls that we'll really be in a position to judge whether those more exotic dynamics plug-ins offer anything truly new.

Sam Inglis
Editor In Chief

"As the engineers of the past well knew, a straightforward compressor with the regulation threshold, ratio, attack and release controls is actually an incredible versatile device."

SOUND ON SOUND

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IN THIS ISSUE

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November 2021 / issue 1 / **volume 37**

FEATURES

SOS AWARDS
VOTE FOR YOUR FAVOURITE NEW GEAR!
PAGE 159

120 POLYEND TRACKER

ON TEST

WORKSHOPS

MXL Revelation Mini FET

Cardioid Capacitor Microphone

MXL's latest mic is as versatile as it is affordable.

PAUL WHITE

Back in September 2010, we reviewed MXL's original Revelation valve microphone, and more recently, in July 2021, we checked out the MkII version. Now those mics have an affordable solid-state sibling: the Revelation Mini FET.

This new mic uses a 32mm capsule, with a centre-terminated, six-micron-thick, gold-sputtered Mylar diaphragm. Its preamp circuit is, we're informed, designed to "provide a classic sound with modern flexibility" and low noise, though oddly no noise figure is included in the spec sheet. Unlike the tube models, which have switchable polar patterns, the Mini FET has a fixed cardioid pattern. Unscrewing the base and removing the body sleeve reveals a double-sided circuit board populated mainly with surface-mount parts.

Supplied in a small camera-style case with an elastic-cradle shockmount included, the mic's styling has a lot in common with its tube counterparts, specifically its general shape and its deep blue finish with bright, black-chrome trimmings. The overall size of the mic is more compact, though, at 59 x 158mm, and it weighs a reassuring 454g. I found the mic to be a secure fit in the shockmount, and spare elastic hoops are provided, which is thoughtful.

The mic features a three-way pad switch, offering 0, -10 and -20 dB options that allow the mic to be used on a variety of sources, including the very loud. MXL's aim for the Mini FET was apparently to get the sound of this mic as close as possible to that of its Revelation valve counterparts. The Revelation Mini FET is touted as an all-rounder that's as happy recording drums and electric guitar as it is vocals or acoustic guitar, making it a good choice for those on a tight budget who need one mic that can cover multiple applications. With the 20dB pad engaged, the mic can handle a massive 158dB SPL before the THD (total harmonic distortion) figure exceeds 0.5 percent, so things like close-miked drums and brass instruments should pose no problems. The quoted frequency response of 20Hz-20kHz doesn't tell you much about the sound of this microphone, but the response graph is more revealing, showing a modest bump at around 120Hz and two equally gentle HF boosts centred on 4kHz and 10kHz. Overall the response only varies by around ±2 or 3 dB, but these gentle lifts are still enough to influence the microphone's character. Its sensitivity is 25mV/Pa, which is very typical of this sort of microphone, as is its 150Ω output impedance. Operation requires a standard 48V phantom power source.

Revel Yells

Used on voice, the mic gives a good account of itself, particularly the way it balances clarity with warmth. As always, you need to evaluate how it performs on your own voice, but the lack of a heavy-handed voicing means the mic is likely to work over a wide range of vocal types. I would certainly be happy use this mic for podcasts as well as music recording, though the lack of a bass-cut switch means that I'd recommend also getting a pop screen.

Moving to acoustic guitar, I was again pleasantly surprised at the quality of the results given that this is such an affordable microphone. Some microphones just seem to make the acoustic guitar sound harsh or gritty, but at the traditional 'where the neck joins the body' position, this one produced instantly usable results with a nice density of tone, lively highs and a smooth midrange. In fact it produced subjectively better results than some far more expensive mics I've tried in the past!

Capacitor mics can give mixed results on electric guitar amps, with some mics capturing close to what you hear in the room and others just sounding a bit strangled or harsh. The Revelation Mini FET did pretty well in this respect, and I got the best results off-axis and around 300mm from the speaker of my Blues Cube combo. The sound didn't seem quite as warm as what I was hearing in the room, probably because the mic was closer to the amp than my ears were, but the mildest of low-mid EQ lifts got it to sound very close.

For a mic that you can pick up for little over £200, the Revelation Mini FET puts up a strong fight, and it appears to live up to its promise of doing a decent job on just about any sound source. Its subtle, mildly flattering character keeps things sounding natural while helping project warmth and clarity. Definitely worth a closer look — and listen!

summary

An affordable mic that nonetheless provides a classy, subtly flattering sound. Well worth considering for someone who's looking for a 'mic of all trades' on a budget.

£ £219 including VAT.
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E sales@musictrack.co.uk
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DAVE GALE

Slate + Ash Landforms

Sample Library

Slate + Ash take a different approach to the orchestral sample library.

The Kontakt instrument format seems to have become an ecosystem all of its own in recent years, with more and more developers seeking to provide content to keep musicians and producers stimulated. While many of these instruments adopt a relatively familiar form, there are occasions when developers seek to shake up the Kontakt norm. This could certainly be said of Simon Ashdown and Will Slater, who form the partnership Slate + Ash. Their latest Kontakt-based instrument, Landforms, draws upon their impressive output and credentials as composers and sound designers.

Sampling From The Real World

At the core of Landforms is a beautifully curated set of acoustic samples, with a little bit of source content in the guise of sound design. Sample capture was undertaken at Real World Studios, in the Wood Room. The vast majority of Landforms instruments are assembled in groups of three; while these instruments conform to section types, some of the blends are really quite interesting. For example, the low woodwind samples consist of a combination of bass clarinet, bassoon and baritone sax. This keeps things blended within the section construct, but as with any traditional nuclear family, there's enough diversity to keep things interesting.

This concept doesn't end here, though. Each instrument has been meticulously spot-miked, and the Wood Room has also been strewn with ambient mics in order to capture all the nuances and resonances that the room can offer. As we'll find later, this provides an irresistible level of control, which extends well beyond the traditional concept of blending the room reverberation with the instrumental sample.

Returning to the sample content, instrument collectives are elegantly organised. The strings are presented in groups of basses, cellos and violins, while the woodwinds are split into high winds and low winds, with an additional trio of flutes. The brass are a little more generic, being split between high and low instrumentation. Flute and Violin are also featured in solo form, while a section labelled Other mops up bonus content, looped and processed elements. This is where the sound-design stuff lives, although substantial interest can be obtained from the main sampled content, heading way out to the leftfield!

Getting Layered

One aspect of Landforms that is relatively traditional is its capacity to layer sounds. A patch or snapshot employs up to two sound layers, labelled A and B, with the ability to control and edit key-zone range, tuning, filtering and envelope controls.

What sets Landforms apart from other more traditional orchestral offerings is that the sampled content

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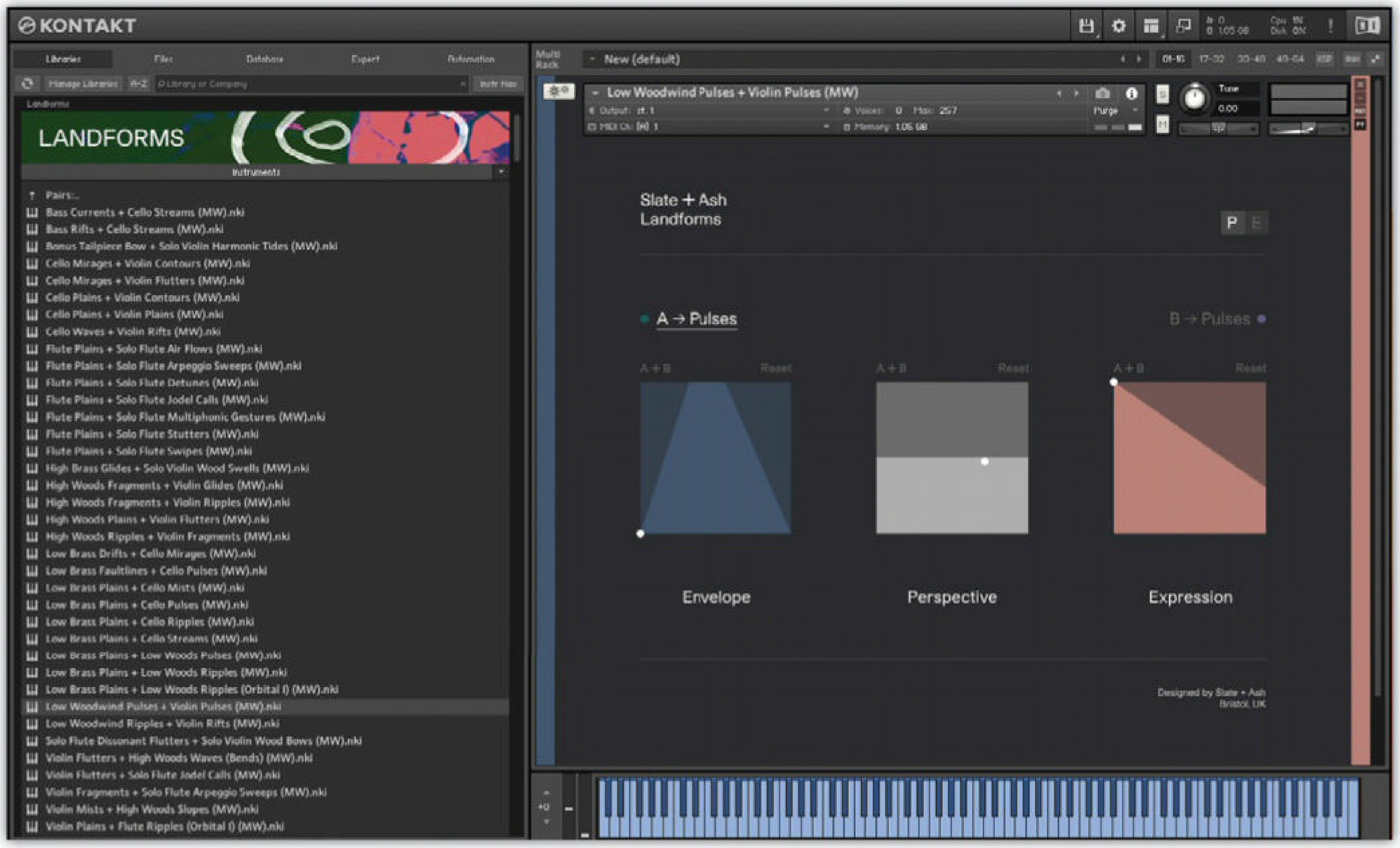


Recordings took place in the Wood Room at Real World Studios.

» is a fair distance from the usual set of long, short and legato articulations. Each instrument set boasts a complement of 13 articulations. I'll call them articulations, but this term tends to suggest the front of a note, and the concept is far more interesting than a plain old staccato or marcato! Slate + Ash use descriptions such as 'flossing' or 'flutters', which can feel a little elusive. It's not always useful if you're in a hurry to create a patch. Maybe that's the whole point; you shouldn't be a in a hurry, because many of these samples take a while to play out, and there are some real prizes to be found among terms such as 'pulses'.

Take a timbre such as the High Brass 'Waves' sample. The instrumentalist's volume increases and decreases in waves, so arguably, you begin to understand the naming protocol. Even at this most basic level, you can hear the quality of recording and instrumental presentation. Humanistic detuning is captured and burnt-in, but never distracting, providing an organic engagement that feels natural.

Other articulation types yield everything from scrapes and harmonics to detuning, repeated notes and much more. Some are very pleasant, others very angsty!

Taking Orbit

The ability to choose any of the aforementioned samples and place them in a layer-cake employing parts A and B is made very easy thanks to a sleek interface, which is relatively minimal in design. Once sample selection has taken place, the main Performance page offers visual representation of an amplitude envelope, which can be assigned to one or both parts. While the envelope is only a three-stage attack-sustain-release design, the phases are generous, with the attack phase capable of a 15-second rise time and the release phase offering up to 25 seconds.

Next to the envelope, the Perspective 'block' allows elegant control over the microphone signals. Clicking in this box throws up a visual represention of the microphone positions, from the three spot mics to the most distant ambient signal capture. This in itself is really terrific, but click on the legend below and you enter a more detailed portrayal of the evidence. This is where you will find the Movement control, which sanctions the ability to apply an LFO across the microphone sources, or better still, draw and record your own wander across the microphone field, which can then be replayed at the stroke of each note. You remember our Brass 'Waves' sample that comes and goes? Now imagine that with movement across the mic field, and that's before we've even driven into the virtual effects arena.

From the main page, clicking the 'E' toggles you into the effects page, moving away from the 'P' for performance page. This section is a huge resource, with further blocks which make easy work of electing and applying effects. As Slate + Ash endorse the use of slightly abstract wording, you will find many favourites lurking beneath less usual names. Effects range from filters, tape delays and tremolos to reverbs and bit-crushers, and everything else in between. Each is presented in a X-Y pad formation, with a wet/dry control to the right of each. In much the same way that a movement element can be applied to the mic placement, the same is true at the effects level.

Making Landforms

Landforms comes equipped with a wealth of content. In total there are nearly 130,000 samples, taking up ahefty 104GB of uncompressed disk space. There's also a complement of 421 really clever Kontakt patches, all curated and divided into menus.

I have to say that Landforms is far more engaging as a creative instrument than the more usual ROMpler libraries, with a potential to get totally lost in sound design that will allow for some very unique and inspiring backdrops. This suite certainly lends itself to the more cinematic and ambient end of musical musings, but could easily find placements in a diverse set of production settings. It's an incredibly inspiring Kontakt instrument, with an elegantly engaging interface that is utter joy to use.

summary

In the pursuit of new ways to present acoustic instrumentation in sampled form, Slate + Ash have captured a perfect balance between classy sonics, an interface which invites tinkering, and boundless amounts of inspiration for production and composition.

£ £349 including VAT.
W www.slateandash.com

Austrian Audio Hi-X15 Closed-back Headphones

SAM INGLIS

The Hi-X15s bring Austrian Audio's proprietary driver technology to the mass market.

One effective way of establishing a new brand is to start at the top. Begin by launching a premium product range, and you entrench a perception that the brand stands for quality. With a bit of luck, this perception will then rub off on more affordable mass-market products further down the line.

This seems to be a strategy that Austrian Audio have pursued, and with considerable success. Their OC-series microphones have won acclaim from professional engineers and have proved able to hold their own against strong competition from established names; likewise, until now, their Hi-X headphones have been targeted at the upper end of the project-studio market. With the new Hi-X15 model, however, Austrian Audio have made available their 'high excursion' driver at a much more affordable price.

Designated Drivers

Physically speaking, there's a family resemblance to earlier Hi-X models, especially in the jaunty angle at which the earcups sit relative to the headband. At a more detailed level, however, pretty much everything is different. The new model uses a much narrower headband, which in turn means different hinge and gimbal designs. There's perhaps a bit more plastic and certainly more colour in the Hi-X15s, but they don't feel cheap or uncomfortable. The earcups rotate freely in both the vertical and the 'fore and aft' plane, the headband is nicely padded and the overall weight is a welcome 50g or so lighter than that of the Hi-X50s, whilst offering a perfectly decent level of isolation. The new design also retains the additional hinges that allow the earcups to be folded right into the headband for compact and safe transportation. At this price it's no surprise that you don't get a proper case, but a soft bag is supplied.

Another welcome design feature that's carried over from the more expensive models is a detachable cable, which attaches to the left earcup using a twist-lock mini-jack. Less welcome is the fact that said cable is only 1.4m long: fine for mixing on a laptop, but not really long enough for general studio use. (Austrian Audio have also launched a variant called the Hi-X25, which adds Bluetooth wireless connectivity to the Hi-X15.)

Look Lively

The Hi-X15s are 5dB or so less sensitive than the Hi-X50s, but given that they still manage a figure of 113dBSPL/V and that their nominal impedance is only 25Ω, you'll have no trouble getting loud working levels from them. In most respects, their specifications are similar to those of the more expensive model, but it's notable that no THD figure is listed for the Hi-X15s.

The outward family resemblance that's visible in the design of the two models is also apparent in their sound. The existing Hi-X models all have a pleasant but noticeably 'scooped' tonality to my ears, which slightly foregrounds the bass and the upper midrange at the expense of the bit in between. The Hi-X15s offer a less refined take on the same sound, with a low end that is sometimes borderline bloated and a really quite aggressive treble boost. I think it's still a tonality you could learn and mentally adjust for, but it certainly isn't neutral, and could become fatiguing during the course of a long session. The Hi-X15s perhaps wouldn't be my first choice as mixing headphones — but at the price, that's hardly surprising. As long as you have a few cable extenders to hand, these have plenty of other roles to play, and I think they'd be an excellent, affordable choice for general-purpose studio headphones.

summary

These affordable, well-built headphones travel well and offer a bright, forward sound.

£ £89 including VAT.
W www.soundtech.co.uk
W www.austrian.audio

Sample Logic Animation Station

Sequencing Plug-in

Sample Logic's first plug-in can bring your samples and music to life.

JOHN WALDEN

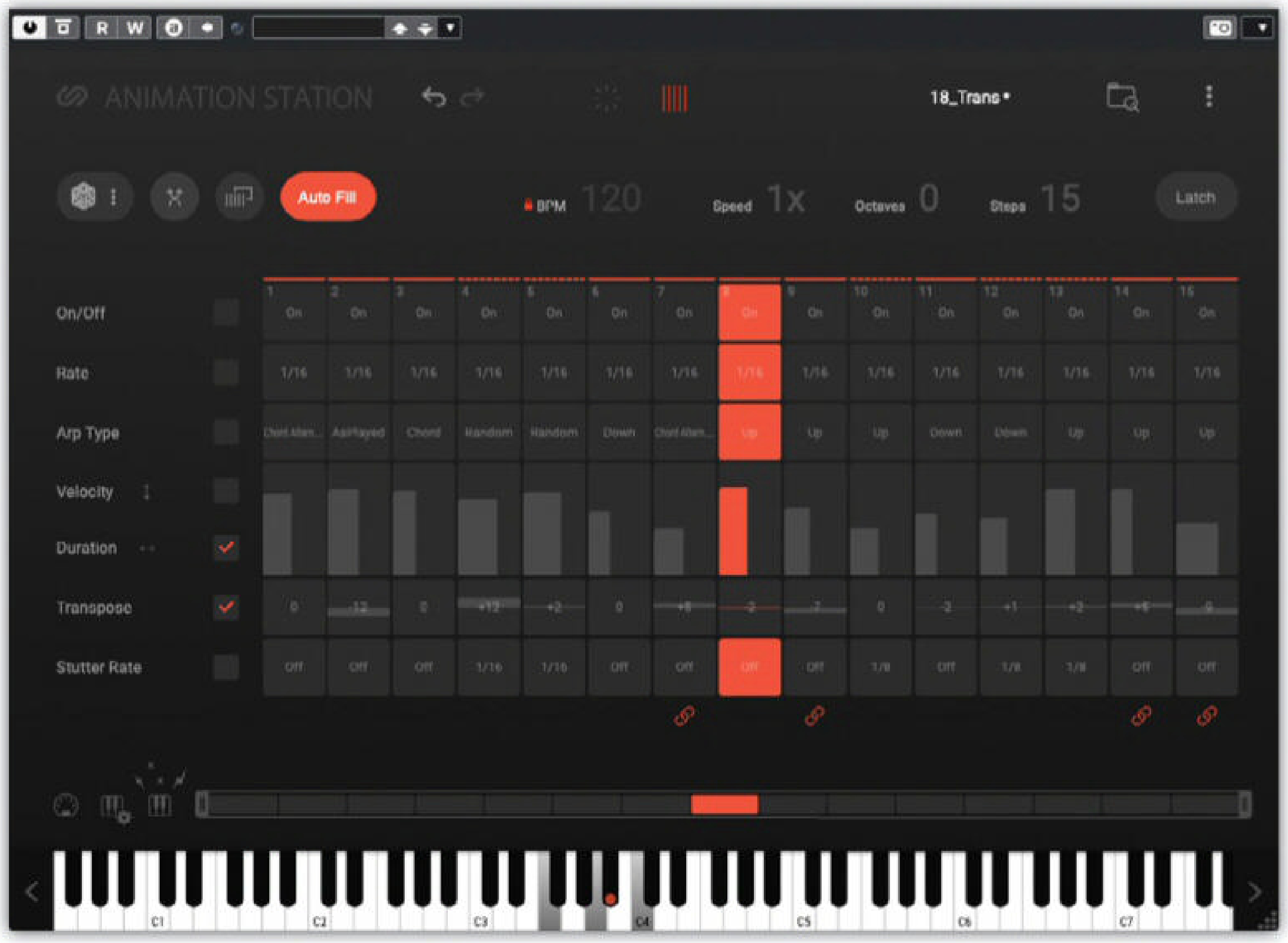
Sample Logic have a well-established catalogue of sophisticated virtual instruments, and have just released their first plug-in. Animation Station delivers a combination of the pattern- and arpeggiation-creation tools found in many of their own instruments as a standalone plug-in, which can be employed with any of the user's favourite virtual instruments.

First Steps

Animation Station is, in essence, a standalone step sequencer/arpeggiator. It allows you to create patterns of up to 64 steps in length and provides control over seven parameters for each step: step on/off, playback rate, arpeggio type, velocity, duration of note within step, pitch transpose of step above trigger note, and stutter rate. Hovering over any lane label produces a drop-down menu to access presets for just that lane. You can also 'link' steps using a button that appears as you hover over a step. This creates a group of linked steps (they don't have to be adjacent steps), where edits made to one step within the group are then applied to all linked steps.

As described more fully below, one of the main features of Animation Station is its emphasis on randomness. You can opt for totally random pattern generation if you are feeling particularly lucky (or experimental!), but you can also constrain the randomisation process in a number of ways to generate more subtle variations on your sequence.

The vast majority of the controls are contained within the default Linear View screen. This includes access



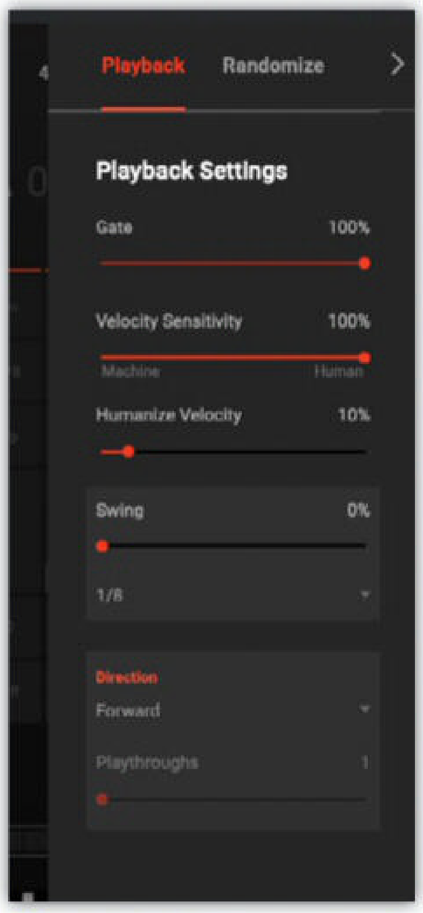
to the preset browser, undo/redo buttons, tempo, playback speed, adding additional playback octaves, number of steps and a latch option. At the bottom of the screen is a zoom bar, MIDI export, Mod Wheel (it can control either velocity or duration) and keyswitch assignment, and a MIDI panic button.

Top-centre, you can switch between Linear View and Circle View. The latter offers only limited editing options, but the circular layout provides rather cool looper-style playback visuals. Top-right, clicking on the three vertically arranged dots opens a two-part settings panel. The Playback Settings panel lets you set a balance between the velocity set in the pattern editor and any trigger note velocity. You can also 'humanise' the velocity, add a degree of swing and change the playback direction mode. I'll come back to the Random Settings panel below. The three dots next to the main 'dice' randomise button pops open tick boxes at the left end of each parameter lane (these are visible in the main screenshot), allowing the user to include or exclude parameter lanes from the randomisation process. Usefully, if the combined length of your pattern's steps does not complete a full bar, the Auto Fill button appears; one click and extra steps will be automatically generated to tidy things up.

Configuring Animation Station for use with one of your virtual instruments is straightforward but somewhat DAW-dependent. The PDF manual provides straightforward instructions for multiple mainstream DAWs, and I was up and running within Cubase in just a few clicks.

Writers Unblocked?

Animation Station's primary function is to provide new musical inspiration with minimal effort. To get you started, the plug-in includes around 300 preset patterns organised into four categories. The Standard and Transposed groups contain a wide variety of pattern styles, with the key difference being »



Amongst other things, the Playback Settings panel gives you options in terms of velocity response and timing.

that the latter feature scale-based pitch shifts in the Transpose lane. Patterns in the Glitched category make full use of the Stutter Rate lane to provide plenty of electronica-friendly timing quirkiness.

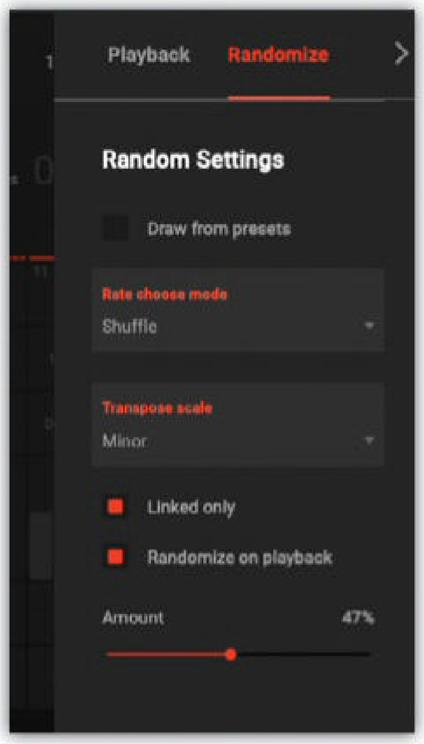
Perhaps more surprising — and a lot of fun to explore — is the Drum Machine category. You can use the patterns to target a single drum sound (for example, to create a pattern for just a kick or just a hi-hat) but holding down the appropriate keys for multiple drums (for example, kick, snare and hi-hat of your drum VST) at the same time can also produce some very cool grooves. It also works brilliantly with orchestral percussion or hand percussion such as claps and stomps.

Roll The Dice

The presets are great, but the real fun of Animation Station lies in using the various randomisation options to supply an assisted musical nudge in creating your own patterns. Sample Logic have struck a very sensible balance here between features and ease of use. providing enough possibilities but without bogging you down in overly complex control sets. That's not to say I can't think of a feature or two I'd love to see added (more on that in a minute), but what's here is both creative and straightforward.

The lane tick boxes mentioned earlier allow you to limit which parameters are subjected to a roll of the randomisation dice, so you can go fully random or just let Animation Station gently stir the pot for a select parameter or two. Also cool is the Shuffle button (the icon with the double arrows next to the Dice button). This retains all the settings for each step, but randomly reorders the steps, which can be very effective in creating pattern variations that retain a sense of being related.

The Random Settings panel provides further control. For example, selecting Draw From Presets forces Animation Station to use the factory presets as a pool of settings for randomisation options.



The Random Settings panel lets you control just how far the randomization process can go in changing your current pattern.

Choosing the Shuffle option for the Rate Choose Mode setting means that Rate settings are swapped with other steps, preserving the pattern length. Enabling Linked Only is particularly useful in that you can limit the randomization to just the selected steps. Equally, the Amount slider can be used to constrain the degree of change the randomisation makes to parameters. And, if you limit the randomisation to just a few lanes and/or steps, enabling the Randomise On Playback option can be very cool; randomisation is applied each time the pattern is cycled through, creating endless variations.

While routing Animation Station's MIDI output to a suitable virtual instrument is very easy, you can also export MIDI for your patterns for further editing and arranging within your host DAW. This works very smoothly and, combined with the randomisation options, it's easy to build longer sequences with subtle variations or to create contrasting song sections from multiple patterns.

Fast Track To Finished

I had a lot of fun using the plug-in. Whether it was to generate some synth bass-line ideas, or arpeggiated/full chords with some interesting rhythmic patterns, Animation Station makes an excellent 'ideas assistant'. Used with a staccato string sound — and avoiding the Stutter Rate lane — it was also an interesting choice for creating string ostinatos. It also worked nicely with a well-sampled 'clap' virtual instrument and produced some very cool acoustic and electronic drum patterns.

That said, if I was prone to greediness, it would be great to see a couple of additions to the feature set that, hopefully, wouldn't compromise the easy-to-use workflow. First, it would be awesome to see a couple of MIDI CC lanes added with the option to specify the CC number target within the virtual instrument being used for playback. This would open up a huge range of useful possibilities for 'animating' your sounds. Second, it would be great if you could trigger the Randomly Reorder Steps button in real time from a keyswitch, and even better if you had the option to apply this action to just linked steps. I've no idea if these suggestions are technically possible, but they would add another layer of icing onto an already well-iced cake.

Like other standalone arpeggio/pattern generators (for example, Sugar Bytes' mind-bending Thesys or Consequence), its price tag makes it more than a casual purchas, but Animation Station does exactly what it is designed to do; it helps you spark a new musical idea, and creating a few variations on a cool pattern can take you from zero to musical hero in a very short space of time. With MIDI export, you can do as much, or as little, hands-on personalisation as you think is required. This is excellent stuff from Sample Logic and I'm sure Animation Station will have considerable appeal for busy media composers or producers who regularly need a fast-track start to a new musical idea.

summary

Sample Logic's first plug-in is both impressive and inspiring. Animation Station is very useful tool for anyone needing new musical inspiration in a couple of clicks.

£ $149 including VAT.
W www.samplelogic.com

NICK ROTHWELL

Bitwig Studio 4

Music Production Software

Bitwig continues to blur the boundaries between instrument and DAW, while offering some welcome practical enhancements.

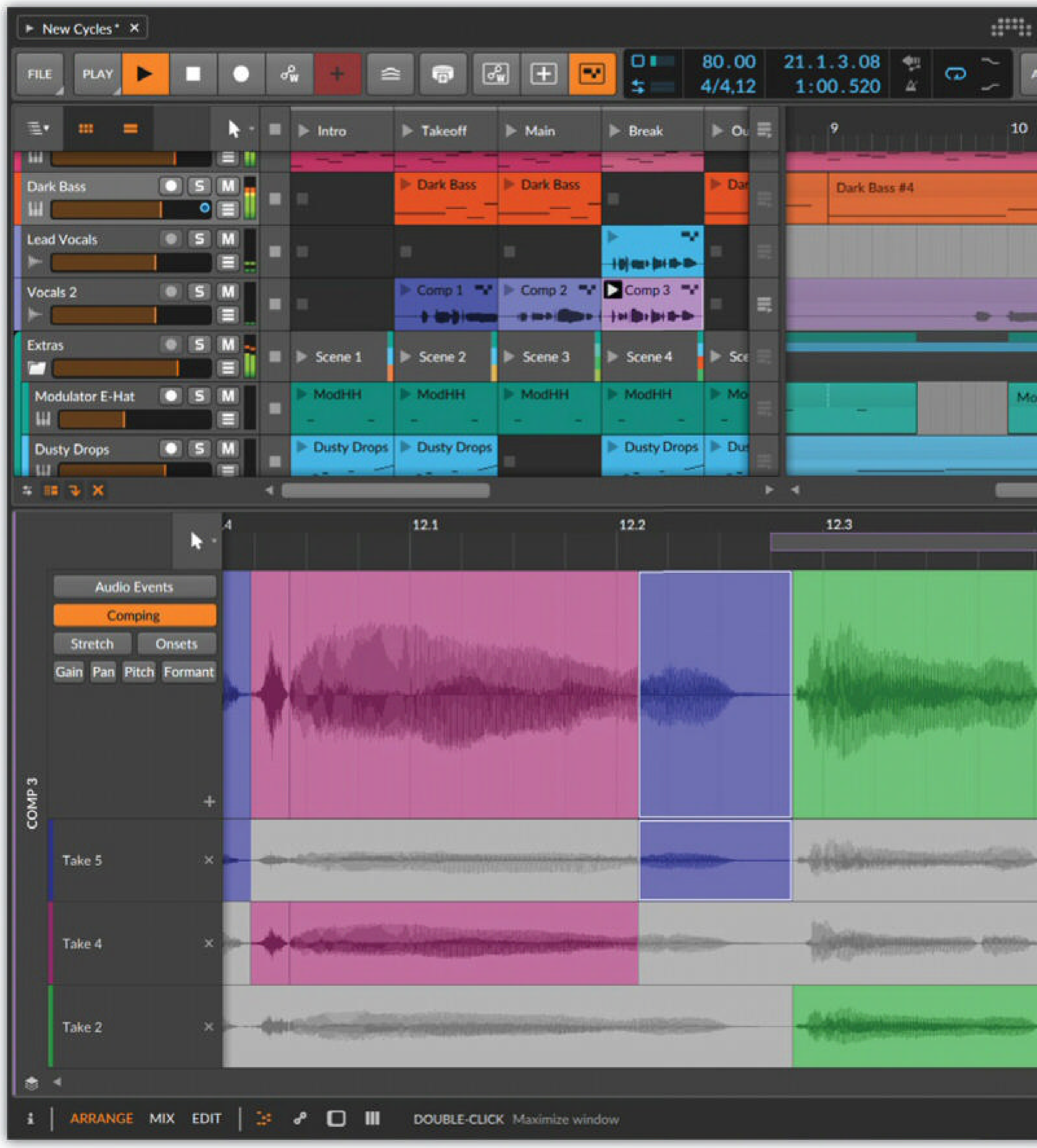
Bitwig Studio seems to be living its life at something of an accelerated rate at the moment. We are seeing a major release once every couple of years, which is quite an aggressive schedule, and the expansion of features from each version to the next also tends to be quite considerable. Version 3 brought us The Grid, a fully integrated modular synthesis environment, nudging Bitwig into becoming an instrument or algorithmic composition tool. Version 4 is another step in that direction, but mostly at the level of sequencer events: notes, audio fragments and loops. Let's see what's in the box.

Comping

Bitwig Studio now supports audio comping. Users of Ableton Live 11 will be familiar with comping as a way of laying down additional takes within the linear arrangement. Bitwig supports this too, but its notion of 'take' is slightly different. To fully make sense of how Bitwig supports comping, it might help to recall how it structures clips. A Bitwig audio clip is not a single piece of audio: it's actually a sequence of consecutive audio 'events', analogous to notes in a MIDI clip. Think of comping as a way of also stacking audio events vertically, in separate takes. What makes Bitwig's comping unusual — and powerful — is that the takes are encapsulated within clips, not just in the project's linear arranger. You want multiple takes in the clip launcher? You got them. If you copy and paste clips, the takes come too.

The obvious way to work with comping is to set a loop in the arranger. Each trip round the loop adds a take to any audio clips which are currently recording. If the recording process starts before the beginning of the loop, then the first take will be longer than the subsequent ones.

Any attempt to record over an existing clip will incorporate a new take into it, rather than replace it. If you do a recording pass over a sequence of clips, each will accumulate a new take, while any gaps in the existing track will



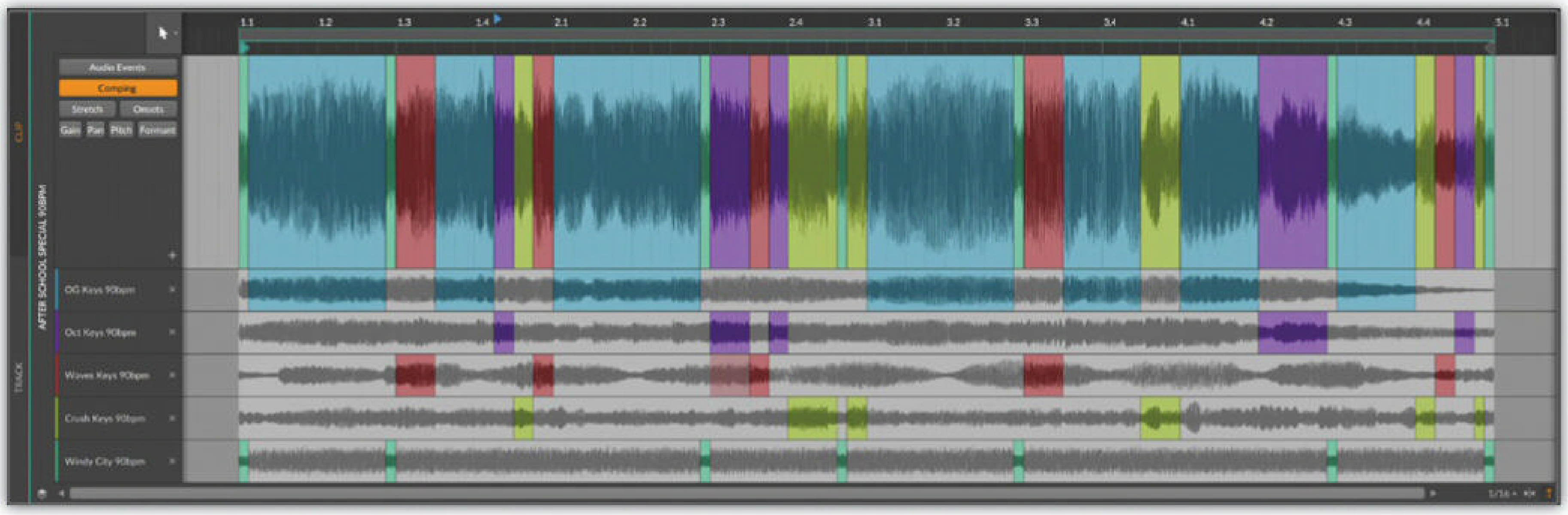
Bitwig Studio 4

€399

PROS
- Powerful comping feature operating inside audio clips.
- Expression spread adds randomness and variation to notes and audio events.
- Operators for notes and audio act as algorithmic compositional tools.

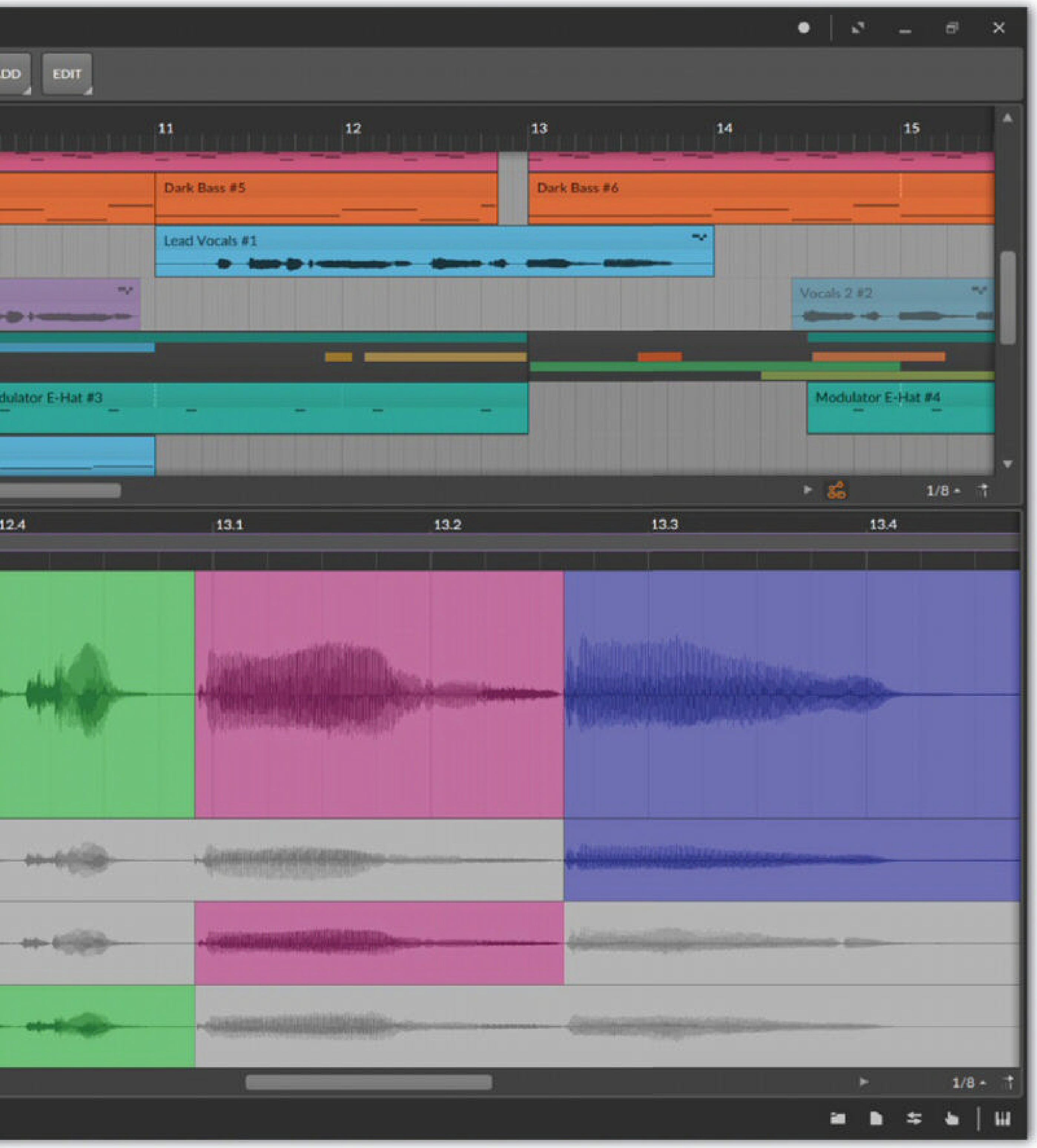
CONS
- No comping of MIDI input.
- Import of Ableton Live sets is partial at best.

SUMMARY
Bitwig Studio 4 strengthens the DAW's position as part composition system, part modular music engine. Comping combines multiple takes with creative audio slicing at the clip level, while operators facilitate patterns and add randomness to notes and audio events. The result is a bigger box of creative tools for experimentation and production.



The comping view of a clip, with multiple take lanes.

be populated with new clips. This does mean that a ‘single’ recording can get rather confusingly fragmented across multiple clips, although the track will play back fine. (A subtle bug which can cause audible clicks in this situation should be fixed by the time you read this.) If this sounds confusing or inconvenient, I recommend clearing out any audio track that you are planning to record into.

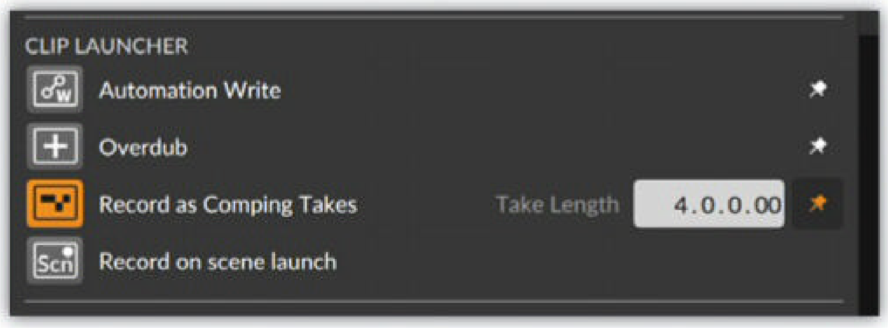


I’ll mention one ‘gotcha’ in the linear take recording process: if the arranger clip has its own internal loop then any attempt to record over it will, well, record over it. Adding takes in this situation was perhaps considered too complicated to implement, or perhaps too complicated to be useful.

Since the clip launcher doesn’t have a timeline or a global loop, setting up comping here requires the take length to be set explicitly. Turn on Record as Comping Takes from the Play menu and set the take length in bars and beats at the same time. Any clip containing multiple takes is marked with a little icon at the right hand end of its title tab.

Stop recording half way through a clip, and you end up with a combination of the last two takes, split at the ‘punch out’ point. Split points separate editing regions across all the takes: click into any take, and its audio for the clicked region will be made active at the top of the clip. You can walk around the various takes and regions with the arrow keys, auditioning different combinations. To subdivide into smaller regions, click and drag horizontally inside a take. You can drag the boundaries between regions to change their size, or you can even ‘peel back’ a region in the composite view to create a gap into which you can link another take.

If you click the Audio Events button on the left of the clip view, you’re back in the world of multiple sequential audio events within a single clip, with no sign of the comping machinery. Well, almost: the audio regions are still coloured, indicating that these are still links into the original takes, even though Bitwig’s menus now refer to these audio areas as ‘events’ »

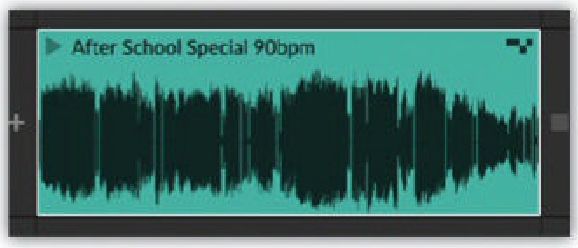


Setting the take length when comping in the clip launcher.

and not 'regions'. Perform any serious editing here, and the link to any take is broken, as a 'region' gets demoted into a plain old 'event'. Switch back to the comping view, and the editing area is now grey, showing its disconnection. (You can still click on another take region to replace it.)

There are a few editing operations which can be applied to take regions without 'breaking' them. Regions can be resized by moving a left or right boundary, or sliced into smaller portions. Their audio gain can be adjusted, and the underlying audio can be nudged forwards or backwards in time.

Bitwig's implementation of comping is versatile and logical. Implementing comping within clips, rather than within tracks, initially seems a little strange, and results in some rather odd outcomes: a single linear recording can create takes within several clips, and there is no overall 'comp view' across the arrangement. But for modern, bar-oriented music, the comping machinery can be repurposed as a creative tool at the clip level, which is a more-than-fair trade-off. Being able to edit comps from rhythmic fragments of material whilst working in the clip launcher feels like a really empowering workflow development.



Multiple takes inside a clip are marked with an icon.

Operators

Bitwig Studio has slowly been growing into a kind of hybrid environment somewhere between DAW and modular synthesizer. Version 2 saw the arrival of 'modulators', which allowed arbitrary plug-ins to be extended with additional control components like LFOs, envelopes and step sequencers. The concept was pushed further with The Grid, the modular synthesis environment introduced in v3.

Any kind of modulation system, especially one which can randomise its behaviour, potentially pushes the composition process away from literal notes and towards musical 'systems' where the composer builds structures which generate note-like events, possibly in unpredictable ways. Bitwig 4's operators (unrelated to FM synthesizer operators) are an attempt to bring in a kind of modulation feature at the level of sequence data, whether that's MIDI notes or audio events in a clip. From a synthesist's perspective, you can think of operators as a kind of modulation that's applied whenever a note is played or an audio event triggered.

Before we look at operators proper, it's probably worth revisiting a feature of Bitwig that's been present since version 1, namely histogram editing. Select a set of events such as notes or automation points, and it's possible to edit them as a group: a histogram display shows all the values as a continuum, and they can be shifted, scaled or randomised in a single operation. A histogram is purely a temporary editing shortcut, an alternative to manually editing each of the individual events. Once altered, values remain fixed until another edit.

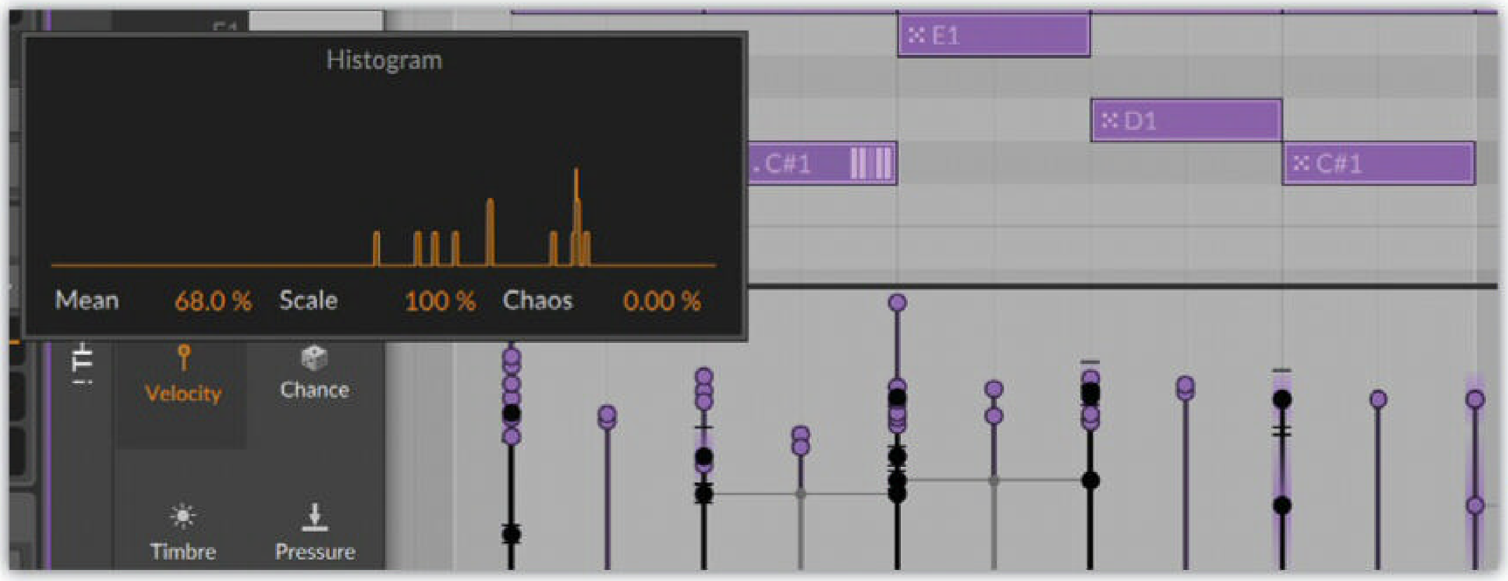
Test Spec

Apple MacBook Pro (Mid 2014)
macOS Catalina 10.15.7
Bitwig Studio 4.0.1

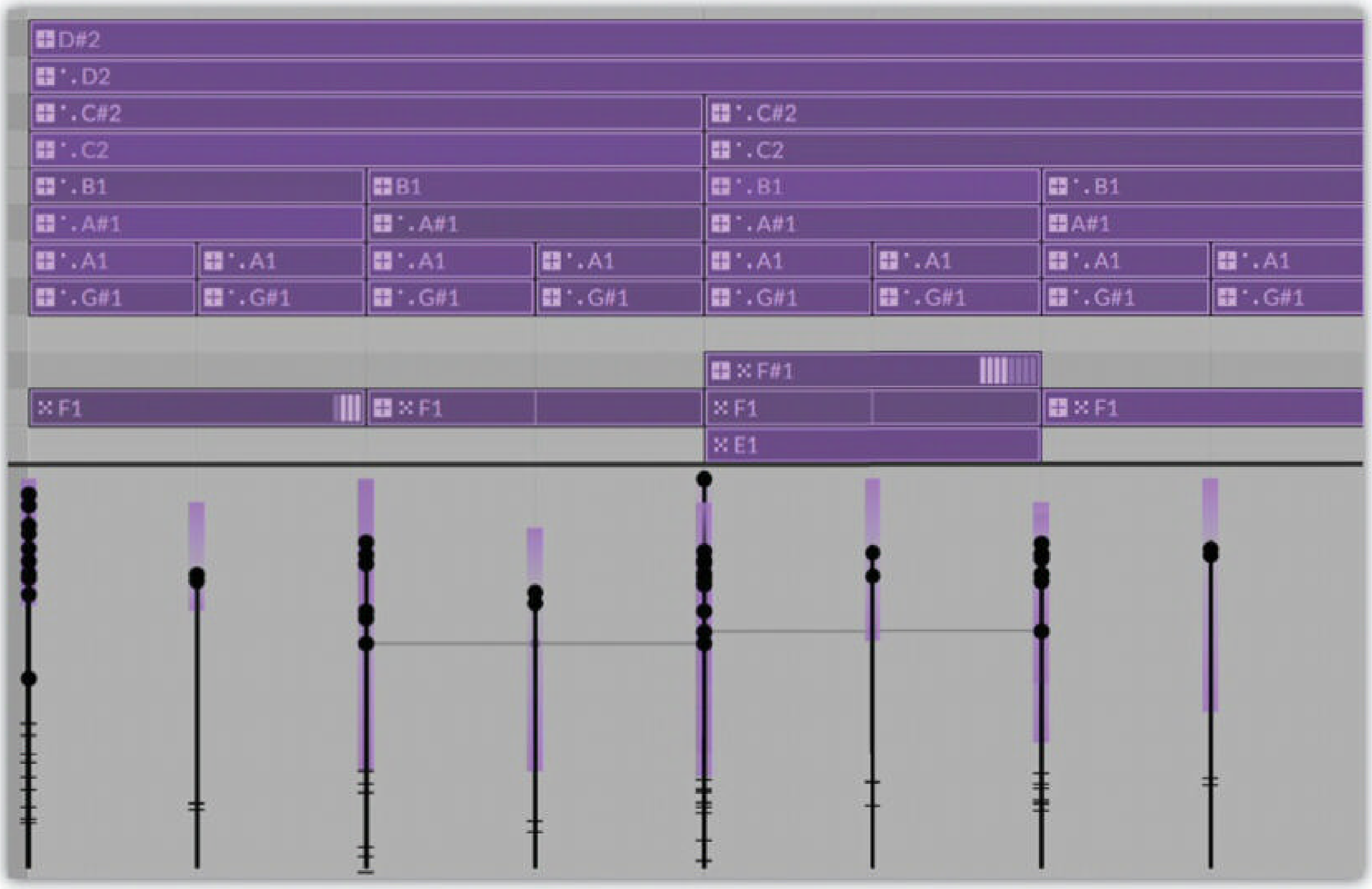
The first step into the world of operators is through a feature called 'expression spread'. This is similar to scaling using a histogram, but is dynamic: there's no editing of the values, but before every playback pass, the values are randomised within a spread range.

Looking at MIDI data rather than audio for a moment, it's possible to apply a 'spread' to note velocities — rather than having a single velocity, a note can be assigned a velocity range (a percentage variance from a central value). Every time a clip is triggered, or each time its loop restarts, actual velocities are assigned randomly within the range and the notes are played. Not all notes have to have the same spread, or any at all — it's on a note-by-note basis. And to add a layer of editing sophistication, since velocity spreads are editable properties of notes, you can select a whole set of notes and then alter the spreads in a histogram.

How much effect velocity randomisation has depends, of course, on the instrument that the notes are playing. For most off-the-shelf instruments. you'll probably get some variance of loudness or brightness, but this kind of controlled randomness might hint at some sound-design ideas: start with a quantised MIDI loop for drums or a rhythmic synth motif, and dive into editing the response to velocity in more radical, unconventional ways. Even a simple trick like layering two instruments on the same track, using note filters to pick instrument



Editing a selection of velocities with a histogram.



Velocity spreads in a tight rhythm clip.

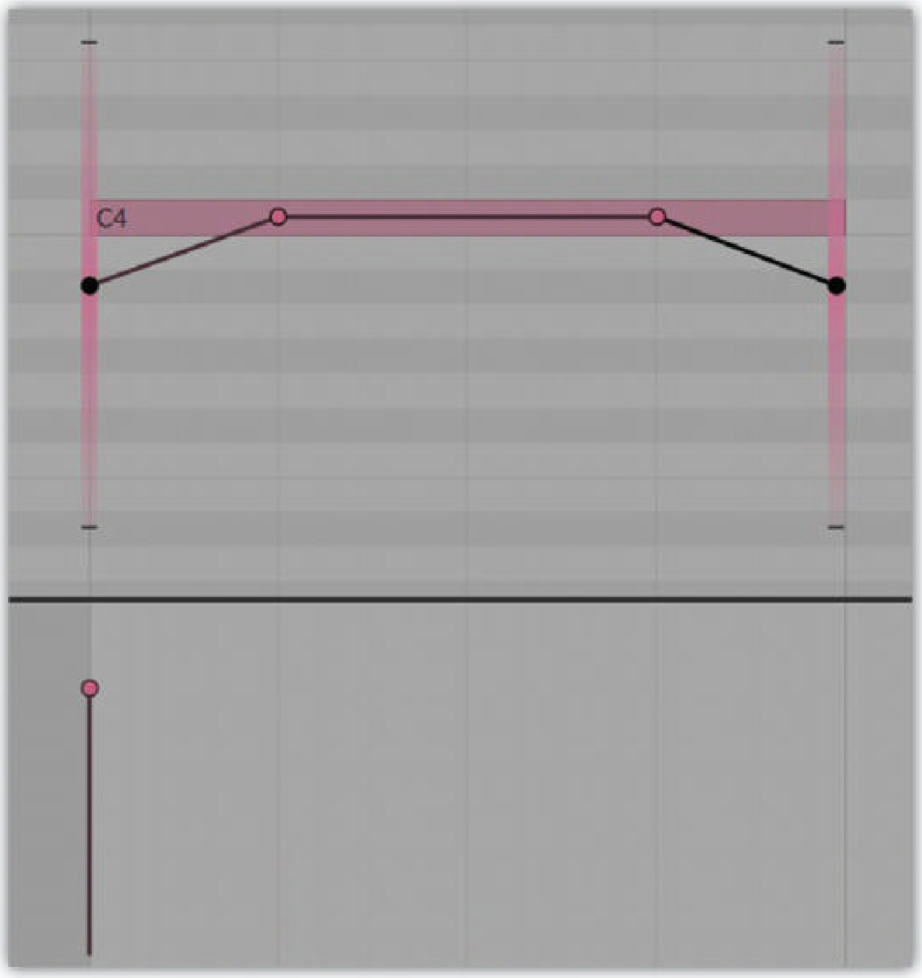
according to velocity, can really animate a simple musical sequence.

For instruments that support MPE, we can play similar games with per-note performance data (timbre, pressure and pitch-bend). Unlike velocities, this data is notionally continuous, like automation, so the spread is applied to control points of the data, not just at note-on. I'm struggling a bit to think why I'd want to randomise pitch (except perhaps for slight detuning), but random timbre or pressure could animate the tone of individual notes within a chord each time it's played. Audio clips support expression spread as well, for gain, pan, pitch and formant.

Random Interlude

At this stage we should probably talk a little about how randomisation works. It's all well and good to apply random processes to musical composition, but sometimes a particular roll of the dice will yield a good result which you want to keep (or which your client wants to keep), making a degree of reproducibility important. Based on what we've discussed so far, each clip launch or trip around a clip loop starts with a randomisation process which replaces what happened the last time through.

The arranger is generally where we want to start setting material into its final, mixed, form. Bounce-record a clip from the launcher into the arranger, and sure enough the randomisation when the clip is played is completely repeatable. If the clip has a loop, the loops will be distinct, but the clip as a whole will always sound the same. You can even start playback part way through the clip, several loops in, and have it behave as if all the randomness has been rendered in place, regardless of how many times it's been looped.



Spreading the MPE pitch-bend in a single note.

This stability is achieved by random seeding. There's no such thing as a truly random number generator as far as computers are concerned, so what they do instead is build repeatable chains of randomness from a single initial numeric seed value. If that seed can itself be randomised in some manner every time a clip is launched, then everything else seems to be random. Fix a clip's seed, and all of its random behaviour becomes repeatable. Every clip has a seed setting:

A fixed random seed for a clip's spread and operators.

if it's set to 'random', then the chain of random values is recalculated on every clip launch. Click to 'capture' a random seed, denoted by a pattern of rectangles, and everything becomes repeatable.

Operators Redux

Let's look at operators proper. There's a group of condition functions that can be applied to any event — a MIDI note or an audio event in a clip. These act as filters, deciding whether that event will play or not. In fact, only the first of these functions, prosaically called 'chance', makes use of the random seeding described above. The others are more computational, generally depending on the current state of a clip whilst it's playing.

Chance is simply a probability that a note or audio event plays — or doesn't. Select one or more events and set the chance to any value between zero and 100 percent. In the clip view, the event is annotated with a small dice icon, showing a number of dots between one and five as a rough visual indicator of the chance, and there's also a graphical editor view — and the histogram view — for editing them.

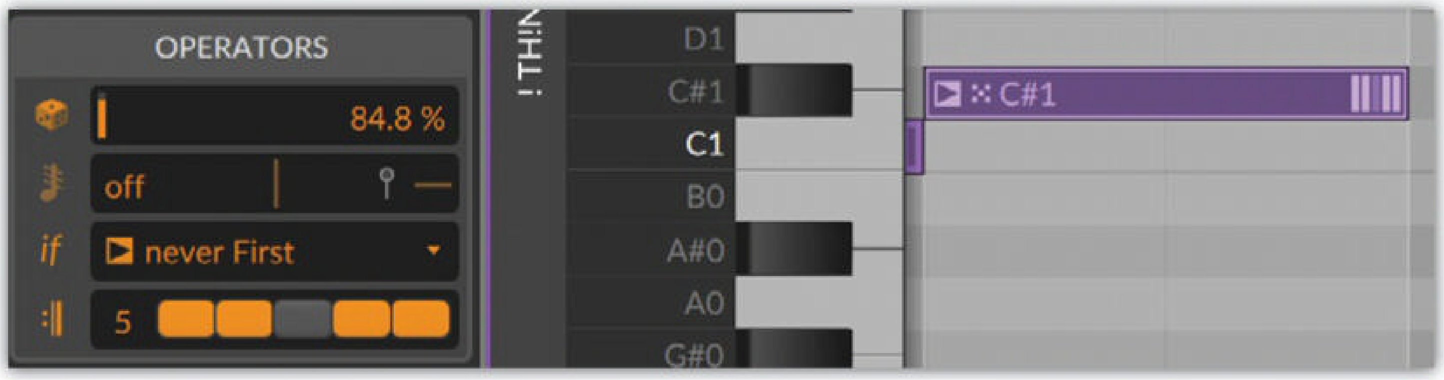
"I've been growing more fond of Bitwig Studio through the years as it has matured from DAW into a well-rounded compositional system, with sophisticated modulation features, an internal modular synth and considerable versatility in some of its built-in instruments."

Like spreads, the event chances in a clip are calculated when the clip is launched and again every time it loops: the events are visually highlighted each time according to whether they will play or not.

The repeats operator effectively cuts a note or audio event into slices, although the operation is non-destructive: think of it as a stuttering effect, or adding grace notes in a musical score. The subdivision can be calculated according to the length of the original event (cut into halves, thirds, etc.) or according to the time grid, placing the repeats on a subdivision of beats. In the former case, the division points will scale or 'stretch' with the event if it is resized; in the latter, division points will be added or removed to keep the specified grid spacing. The grid-based divisions are not restricted to powers of two, either: if you want 37 repeated notes in a bar, perhaps to add a bit of a Conlon Nancarrow groove to a piano part, feel free. Certainly, divisions into thirds and sixths give you plenty of options for polyrhythms.

For MIDI notes, a response curve allows the velocity to be scaled up or down along the repeats, allowing for some dynamic change along the event; sadly, there's no way to apply such a curve to the amplitude of repeated audio events, so that would need to be done with automation. Also, it would be nice to be able to add a curve for other properties, such as pitch or timbre, through the repeats.

A feature which is shared between notes and audio events is the ability to shift the timing of the repeats into an accelerando or ritardando phrasing, to add some speeding-up or slowing-down interest to drum fills or other effects, for a bit of a Rival Consoles vibe. A note-with-repeats or audio-event-with-repeats is still a single event from an editing perspective, so other processing like spreads and chance »



Operators applied to a note.

» applies to the entire event with repeats, not the individual repeats themselves.

The recurrence function is another filter, causing events to either play or not. For an event you can specify a recurrence length (from two up to eight) which is used to count the clip loops as they repeat. For each 'lap' of the loop in this recurrence length, there's a toggle box to indicate whether the event plays. Once past the recurrence length, the process starts again. The best way to think about this (or, at least, the way which works for me) is to imagine the clip loop unrolled linearly over time. A recurrence length of two makes the loop twice as long, duplicating all the events but letting some of them play only in the first half, or the second. A recurrence of three makes the loop three times as long, and so on. If you stack up some dense drum parts in a short loop and then thin them out by adding recurrence with differing lengths, you get longer, more engaging rhythms really quickly. At the very least, if you often find yourself doubling the length of a MIDI loop, duplicating all the notes, and then selectively editing or removing notes in one half or the other, This will save you a lot of time and be more manageable. Little icons at the end of notes or audio event tabs in the clip view let you keep an eye on what's going on.

I've left 'occurrence' until last because it's probably the most complex, and most subtle, of the filters. As with recurrence, an event plays, or not, according to some calculations made about the state of the clip. An event can be made to play once only when a clip is triggered. (This means the first occurrence of the event: jump into the middle of an unrolled loop in the arrangement and the event won't play.) Or an event can play every time except the first. An event can be made to play only when the previous event did, or only when it did not. For MIDI clips, the playback can be dependent on being the same (or different) note pitch, or same or different MIDI channel. (For some reason, these conditional events don't get highlighted dynamically during playback according to whether they trigger.)

Occurrence can also be established by the state of a global 'fill' toggle, presumably intended to be mapped to a MIDI footswitch to add fill parts to a drum pattern. There's only one fill control in the entire session; I'd quite like to have had one per clip (or at least per track), but you can at least automate it in the project's master track.

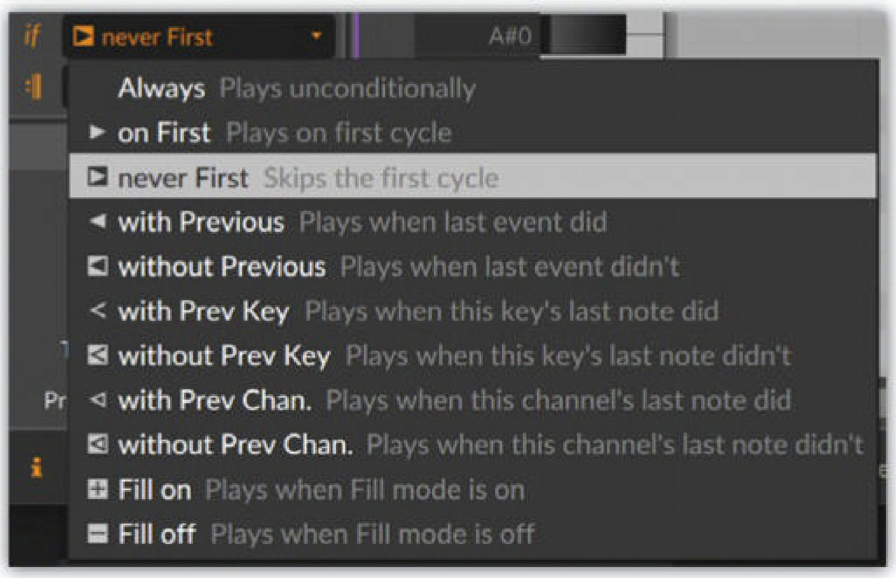
The subtlety of occurrence involves simultaneous notes in a chord: if the notes are all set to not play after a previous event, will they interfere with one another, and which notes will actually trigger? Bitwig very neatly sidesteps this problem by treating all notes at the same time location as comprising the same event, so every quantised chord is 'atomic': it either registers as a played event for following notes or it doesn't.

There's a lot to digest in the operator machinery, given how individual events can influence those downstream in a sequence according to potentially complex combinations of rules. The Bitwig press release states that the operators work "in parallel", though to my mind it's more 'in series': any one of them can remove an event from playback, and that can then cause alternate events to trigger instead.

I've long been a fan of randomness in composition, and it's one of the first things I jump to in any device or environment I use, but with Bitwig's operators it's best used sparingly. The ability to combine occurrence and recurrence to 'unroll' simple melodic loops into more sophisticated forms is something I found really engaging as a creative process. It's a bit of a shame that note repeats can't be made conditional in some way; you can hack the effect by placing two identical notes in the same place, if they're on different MIDI channels, and alternating between them, but it's a bit of an effort. On the whole, though, operators form a sophisticated toolbox, letting you transform the base metal of a simple step sequence or drum pattern into the gold of a longer, more varied and engaging musical idea.

Other Enhancements

There are other low-profile — though non-trivial — enhancements that have arrived with Bitwig 4. On the Mac, there is now native support for the M1, Apple's new ARM processor architecture. There's lots to be said about the M1 in relation to music software, although as the owner of a seven-year-old Intel Mac, waiting for the next generation of ARM machines to arrive, my thoughts would be somewhat speculative. One thing that Bitwig have achieved, however, is to port version 4 to the M1 processor whilst maintaining support for legacy Intel plug-ins (which, at this stage, is still most of them). This is to some extent an accident of fortune, because Bitwig has always had the option to host plug-ins in a separate process from the program proper. This allowed the roll-out of an M1-based DAW using Intel-based plug-ins, which is at the moment the best of both worlds, and should allow for a smooth transition from Intel to M1 over the coming months and years. Any Intel code still needs to be



The different occurrence operators.

passed through Apple's Rosetta 2 translator, and it's not clear what the performance gains are for Bitwig on M1 given that most of a DAW's CPU load is from the plug-ins themselves, unless you stick to the built-in devices. But this does mean that Bitwig can steam ahead with M1-based development and not leave users with their legacy instruments and effects behind — at least as long as Apple support Rosetta.

Bitwig 4 also claims to be able to import projects from other DAWs, quoting FL Studio and Ableton Live as supported formats. I was rather sceptical about this, and found my scepticism pretty much confirmed. Bitwig loaded projects from Ableton Suite 11 without much complaint, but they were, to say the least, incomplete. Some track names and instrument names were missing. A drum rack imported with a complaint about missing samples, which is odd because they were in the standard place in the Live project; importing them manually didn't fix things. An attempt to import Live's native Wavetable synthesizer, obviously doomed to failure, resulted in a rather arbitrary instance of Bitwig's Polymer synth with a completely different sound, although to be fair it did have the correct preset name. An Ableton Live flanger did miraculously make it into Bitwig sounding pretty much unfazed (as it were) by the ordeal.

VST presets seemed unharmed, and MIDI clips seemed to transfer OK (even those with MPE gestures!), although the story with audio was a bit more mixed, as a syncopated warped drum loop in Live transformed, not entirely disastrously, into a pattern containing fragments in triplet time. Track groups and routings were intact, as were the positions and MIDI data of arrangement clips, but any kind of automation was basically dropped on the floor.

The question to ask yourself, faced with a copy of Bitwig Studio 4 and an Ableton Live set, is what you want to achieve and how much work you are willing to do to get there. Porting track setups, clips (launcher or arranger) and third-party plug-ins seems pretty robust. Anything more complex or more Ableton-specific and you're in uncharted, and probably unchartable, territory.

Conclusion

I've been growing more fond of Bitwig Studio through the years as it has matured from DAW into a well-rounded compositional system, with sophisticated modulation features, an internal modular synth and considerable versatility in some of its built-in instruments. The comping feature in version 4 is surprisingly powerful by virtue of being implemented at the clip level, allowing for some creative applications, while the operators open the door to new algorithmic compositional techniques, complementing the modulators at the device level.

I can't think of any real down sides to the new release, but at the risk of pointing out the obvious: this impressive music environment is, at the end of the day, all screen-based. Do explore and exploit all that it can do, but don't forget to get out into the real world with a physical controller and actually make some music!

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COMPRESSION

What Do All Those Knobs Do?

Dynamics processing is a core mixing technique, but it's often misunderstood. We explain how to make a compressor do what you want it to!

SAM INGLIS

Sound is dynamic. It varies in level on a large timescale, such as when a loud finale follows a reflective slow movement. It also varies on a much smaller scale. For example, each time we hit a piano key, the note starts loud and then dies away.

When we record or mix sound, we sometimes want to alter this dynamic behaviour. We might wish to amplify the quiet passages so they can be heard more easily. We might seek to make a performance appear more consistent. We might even want to change the dynamic character of individual notes, for instance by making them sound more or less staccato.

The simplest way to alter dynamics is to adjust the level using a fader. A human listener can follow the large-scale dynamic variation in a piece of music and make it louder or quieter as appropriate. However, small-scale dynamic variation — sometimes known as micro-dynamics — happens too fast for a human operator to respond reliably. Enter the compressor: a device that 'listens' to the input signal, analyses its dynamic variation, and adjusts the level automatically in response.

Some compressors have almost no controls at all. Others bristle with an intimidating array of obscurely named knobs. The aim of this article is to demystify these parameters and help you to use compressors with confidence.

Insert Or Send?

Conceptually, a compressor is like an automated fader. It's a transformative rather than an additive process: it modifies, rather than augments, the sound. For this reason, a compressor plug-in or hardware unit would usually be used as an insert, not an auxiliary effect. If we want to compress multiple sources at once, we either need to use a separate compressor for each of them, or bus them to a single channel and compress that channel. Both are valid approaches, but they have very different effects and applications.

Watching The Detectors

Most of the key controls on a compressor adjust the circuit or algorithm that decides whether the input signal should be turned down or left alone. This is known as the detector, and it usually receives its own, dedicated copy of the input signal, on a separate path known as the **side-chain**.

A basic detector could simply compare the peak level of this side-chain signal with a chosen fixed value, and trigger gain reduction in the audio path whenever the one exceeds the other. The lower we set this value, the more likely the signal level will be higher at »

The stock Compressor plug-in bundled with Steinberg's Cubase has an Analysis control that sets balance between peak and RMS detection.

What About Stereo?

The much-loved AMS Neve 33609/N is designed as a stereo compressor, but allows the two channels to be unlinked. In this mode they can be used as independent mono compressors for different sources, or on 'stereo' material where preservation of image isn't an issue.

By definition, the two channels of a stereo recording contain different signals. These signals won't necessarily peak at the same time or the same level; so what should happen when we place a compressor across a stereo channel?

Most stereo compressors default to acting identically on both channels. Gain reduction is applied equally to both sides whenever a peak occurs in either. This is the safest option, because applying gain reduction unequally risks an audible sideways shift in the stereo image. It also makes the design of the compressor simpler, because the side-chain signal need only be a mono sum or average of the left and right channels.

However, some compressors do have a true stereo side-chain, and give you the option to partially or wholly unlink the two sides. Fully unlinking the sides makes a stereo compressor behave as two separate mono devices that happen to share the same settings, and can be useful on group channels such as backing vocal auxes which aren't carrying true stereo signals, just lots of related mono sources. (The same can be achieved in some DAWs by using a dual mono rather than a stereo plug-in instance.)

Quite a few compressors also have the option to operate in **Mid-Sides mode**, whereby the left and right channels are matrixed into Mid and Sides channels, compression is applied, and then the matrixing process is repeated to restore left-right stereo. In this case, unlinking the two channels in a stereo side-chain causes the stereo width to vary, which is much less objectionable than left-right shifts in stereo image, and it can in fact be very useful, especially at mastering. For example, if we're sent a mix to master where the vocal is too loud, we can often improve matters by compressing only the Mid component.

» any given moment, and the more often compression will happen. The fixed value in this scenario is known as the **threshold**, and this is one of the key parameters in nearly all compressor designs.

If the aim is to recreate the way our ears respond to loudness, however, this sort of detection will be a long way wide of the mark. Using peak level as a surrogate for loudness ignores the fact that two signals with the same peak level can sound wildly different to the human ear. It makes no distinction between sudden and sustained sounds, or between low and high-frequency sounds.

A more sophisticated detector circuit thus measures the rolling average of the side-chain signal level. This average is known as the 'root mean square' or RMS value, and although this still isn't a perfect substitute for loudness as we perceive it, it's usually better than the peak value. Some compressors offer the option to switch between RMS and peak detection; usually, RMS will give a smoother and more natural response, but peak will react faster and may be useful on very percussive sources, or when the aim is to ensure that a particular peak level can't be exceeded.

Another core parameter called **ratio** tells the compressor by how much a signal that does exceed the threshold should be turned down. A ratio of 2:1 will see the signal turned down by 1dB for every 2dB that the side-chain signal is above the threshold. Increase that to 4:1 and the signal will be turned down by 3dB for every 4dB above the threshold. Or to put it another way, in order for the output signal to rise by 1dB above the threshold, the input signal must be 2dB above for a 2:1 ratio and 4dB for a 4:1 ratio.

Attacking Instincts

Whether it uses peak or RMS detection, a compressor is of limited value if its reactions are always instantaneous. A device that turns the signal down immediately it exceeds the threshold can't respond to dynamic variation on a larger scale. It can also cause distortion on low-frequency sounds, by kicking in on every cycle and thus changing the waveform. Even if we can't hear distortion, we may well be able to hear other artifacts.

For this reason, nearly all compressors have two additional parameters, known as their 'time constants'. The **attack** control allows us to slow down the initial action of the compressor. Let's say that the detector has determined that 4dB gain reduction is appropriate on a given peak. If we set the attack time to zero, this 4dB level drop will be applied straight away. If we set it to 100 milliseconds, gain reduction will be applied gradually, reaching 4dB only after that time. In human terms, it's the difference between having a fader instantly jump to the -4dB point, and having it moved smoothly to that point.

The **release** time constant has a similar role at the other end of the process. With an immediate release, the compressor will stop acting the instant the input level drops below the threshold. A longer release time causes it to relax its attenuation more smoothly.

Target Practice

Together, these four key parameters provide a lot of flexibility for shaping the response of the compressor. Lowering the threshold will mean gain reduction takes place more often. It also means the signal is more likely to exceed the threshold by a greater margin, and will thus be turned down further. Increasing the ratio likewise means that signals exceeding the threshold will be turned down more. Longer time constants, meanwhile, tend to make the action of the compressor smoother, and more focused on long-term dynamic variation rather than micro-dynamics.

One important thing to note is that detection is a moving target. For example, let's say we set the attack time to 100ms,

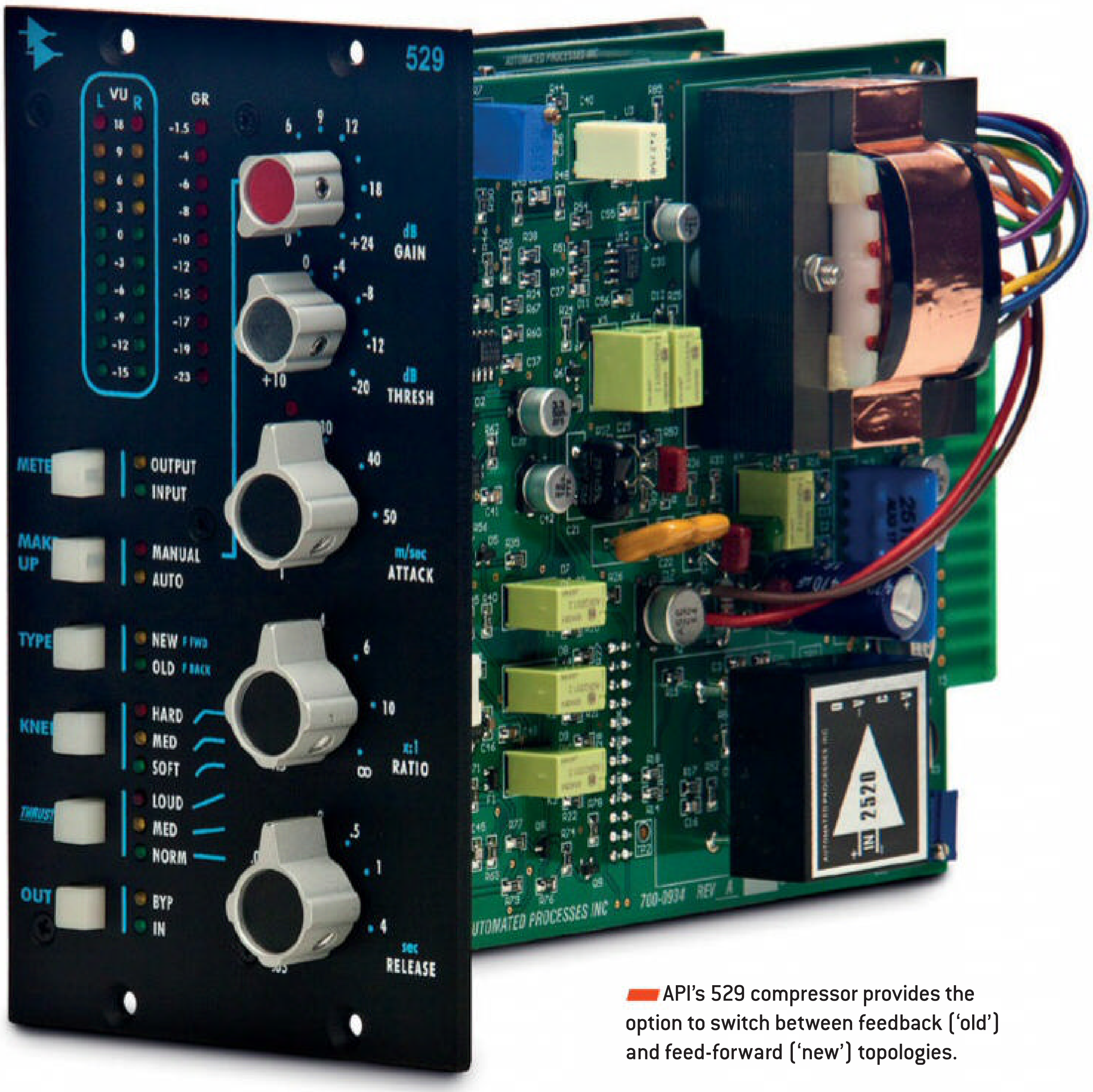
and the ratio to 4:1. A peak that exceeds the threshold by 8dB will, in theory, trigger gain reduction that steadily increases to 6dB after 100ms. But if that peak is only 50ms in duration, the signal will fall below the threshold again before the attack phase is over. The detector will quickly 'change its mind' and switch the compressor into its release phase, so 6dB gain reduction will never be reached.

Another factor affecting the position of this moving target is the **topology** of the compressor: whether the side-chain signal that feeds the detector is tapped before or after any gain reduction takes place. A feed-forward compressor taps the side-chain from the raw input signal, but in a feeback architecture, the side-chain is derived from the processed signal. There are a few compressors that can be switched between the two modes; in general, a feed-forward compressor has a snappier and more obvious action, but offers less precise control of level, while a feedback design is smoother and more accurate.

Time And Motion

If a compressor has the four key parameters described so far — and most do — that's »

The famous SSL compressor has the standard Threshold, Ratio, Attack and Release controls and is a hugely versatile device.



API's 529 compressor provides the option to switch between feedback ('old') and feed-forward ('new') topologies.

A peak limiter such as FabFilter's Pro-L 2 is a specialised kind of dynamics processor optimised to work with very fast attack and release times and very high ratios, in order to prevent any peaks exceeding a fixed 'ceiling' level.

» enough to make it a very flexible tool. Before we move on to consider some additional controls, let's consider how these parameters might be used to achieve different results.

On a snare or kick drum track, we could choose a fast attack and a medium release, a high ratio and a fairly high threshold. Then, only the loudest hits would trigger compression, and would do so immediately: useful for levelling out an unevenly played part. By contrast, if we increase the attack time and lower the threshold, we ensure that all hits trigger compression, but only after the initial transient has passed. This has audibly different results. The sound of the drum will be changed, emphasising the attack of each hit and reducing the sustain. We might take this approach if, for example, the drum is ringing more than we want.

Now let's think about the different ways a compressor placed across the entire mix bus might respond. By setting a very high threshold, the shortest possible attack and the highest possible ratio, we can approximate a special type of compressor called a limiter. In effect, we're specifying a 'ceiling', a level above which no signal is allowed to creep. Any peaks that dare to show themselves above the threshold instantly trigger gain reduction, ensuring that this ceiling level is never breached.

We can achieve a completely different effect by setting a very low threshold, a very low ratio (perhaps as low as 1.1:1) and more moderate time constants. In this scenario, the detector is triggering compression most of the time, and the 'moving target' effect described above is continuously in action. The end result is a series of complex but subtle dynamic changes that we hear as a gentle smoothing-out of the mix.

Filtering Down

Because the side-chain signal that feeds the detector circuit is not actually in the audio path, processing can be applied to the side-chain signal without directly affecting the main signal. This possibility is often exploited to apply equalisation or filtering to the side-chain signal, weighting the response of the detector towards one area of the frequency spectrum. Probably the simplest and the most common application of this idea is to incorporate an optional **high-pass filter** into the side-chain.

The effect of this is to make the detector less sensitive to low frequencies, which often helps to align its action more closely with what we are hearing. Our ears are much more sensitive in the midrange than in the bass; so there are times when we want a compressor to even out the midrange, but it 'hears' the low end as being dominant and responds to that instead. That's especially the case on complex material such as full mixes.

High-pass filters are found in many compressors. Some also feature additional side-chain EQ controls, such as the Thrust options on some API compressors. One common type of de-esser is simply »

Warm Audio's Bus Comp is based on the classic SSL design, and incorporates a variable high-pass filter for situations where you want the processing to respond to midrange rather than low-frequency energy.

a compressor with an exaggerated treble boost in the side chain, emphasising the bursts of high-frequency noise that make up sibilance and triggering compression. Taking this further, some hardware compressors have an insert point that allows you to patch in the EQ of your choice.

Taking Sides

Most DAWs don't allow one plug-in to host another as an insert, so you can't actually patch a software EQ into a software compressor's side-chain. However, many plug-in compressors do have an external **side-chain input** or **key input** which allows a signal of your choice to be fed to the side-chain and used to trigger compression. If you want to experiment with side-chain EQ, you can duplicate the source to a second track, equalise it and route it to this input. But it's also possible to use the side-chain input for other purposes. In particular, you can route a completely different signal to the side-chain input, and this is the basis of numerous compression techniques.

This type of side-chain compression can be used to create an effect that has been a staple of electronic music ever since French house became a thing. It's achieved by placing a compressor across the master bus, with the side-chain input fed from the kick drum track. Every time the kick drum hits, the entire mix is compressed, creating a 'pumping' motion. Other well-known applications for side-chain compression include ducking, whereby instrumental parts or reverbs and delays are 'pushed' into the background by a vocal, then allowed to swell again in the gaps.

Up Up And Away

You'll sometimes see compression described as a means of making things louder. On the face of it, this seems confusing: as we've seen, the fundamental job of a compressor is to turn things down when they get too loud! In fact, there's no contradiction. What most compression settings do is to reduce the **crest factor** of the signal: by turning down the loudest peaks, they reduce the ratio between the peak level and the average level. If we then turn the compressed signal back up again so that it peaks at the same level as the uncompressed signal, the average level will now be greater, and hence it will sound louder. All compressors have the means of doing this built in, usually through a control labelled **make-up gain**.

Knees Up

As obscurely named controls in music technology go, **soft knee** and **hard knee** are right up there. They take their name from a common graphical representation of compression, and relate to the behaviour of signals around the threshold level. In a hard-knee compressor, the transition from doing nothing to full-ratio compression takes place as soon as a peak reaches threshold level. In a soft-knee design, the process is more gradual: compression at a low ratio begins to take place as the signal approaches threshold level, and the ratio steadily increases as it goes further above the threshold. (Dbx coined the alternative term 'over easy' to describe soft-knee compression.)

In most cases, the action of a hard-knee compression is more assertive and less subtle than that of a soft-knee model; soft-knee compression is often preferable when we want to retain the illusion that no processing is taking place.

Parallel Lines

Newer compressor designs often now feature a **wet/dry mix** control. This allows you to implement a technique known as parallel or New York-style compression, whereby a heavily compressed version of the source is blended with an uncompressed version. This modifies the sound in complex ways, with results that are audibly different from straightforward compression. For more detail, I'd suggest consulting Hugh Robjohns' in-depth article on the subject: www.soundonsound.com/techniques/parallel-compression. Alternatively, dive in and experiment! Be aware, though, that there is no universal standard for how the 'wet' and 'dry' sides of the signal should be balanced, so you may find that a setting which works for one compressor is completely different on another.

A wet/dry control isn't essential for implementing parallel compression, however. You can achieve the same effect

Softube's FET Compressor is one of many plug-in compressors that can accept an external side-chain signal.

Based on the legendary Urei 1176, Black Lion Audio's Seventeen compressor is an example of a 'fixed-threshold' design.

Meanwhile, Back In The Real World

In this article, I've described threshold, ratio, attack and release as being the most fundamental compression parameters. But there are some classic hardware compressors — and plug-in emulations of them — that don't have even these controls. This is because analogue electronics can only ever approximate the behaviour of an ideal compressor, and some older designs are very limited in comparison.

Real-world electrical components can react to input signal levels in complex ways. The classic Teletronix LA-2A, for example, runs its side-chain signal into an illuminating panel which, in turn, triggers a light-sensitive component that applies more gain reduction as the light gets brighter. This pair of components exhibits very complex behaviour, with fixed but relatively long attack and release times, a very soft knee and a highly variable ratio. Another classic design, the Urei 1176, does have time-constant and ratio settings, but it too lacks a threshold control.

Both are examples of **fixed-threshold** designs. With no user control over the threshold value, we vary the amount of compression by raising or lowering the input level. Instead of adjusting the threshold level relative to the input signal, these are designed so that we adjust the input level relative to a fixed threshold. A higher input level equates to a lower threshold and therefore more compression (and vice versa).

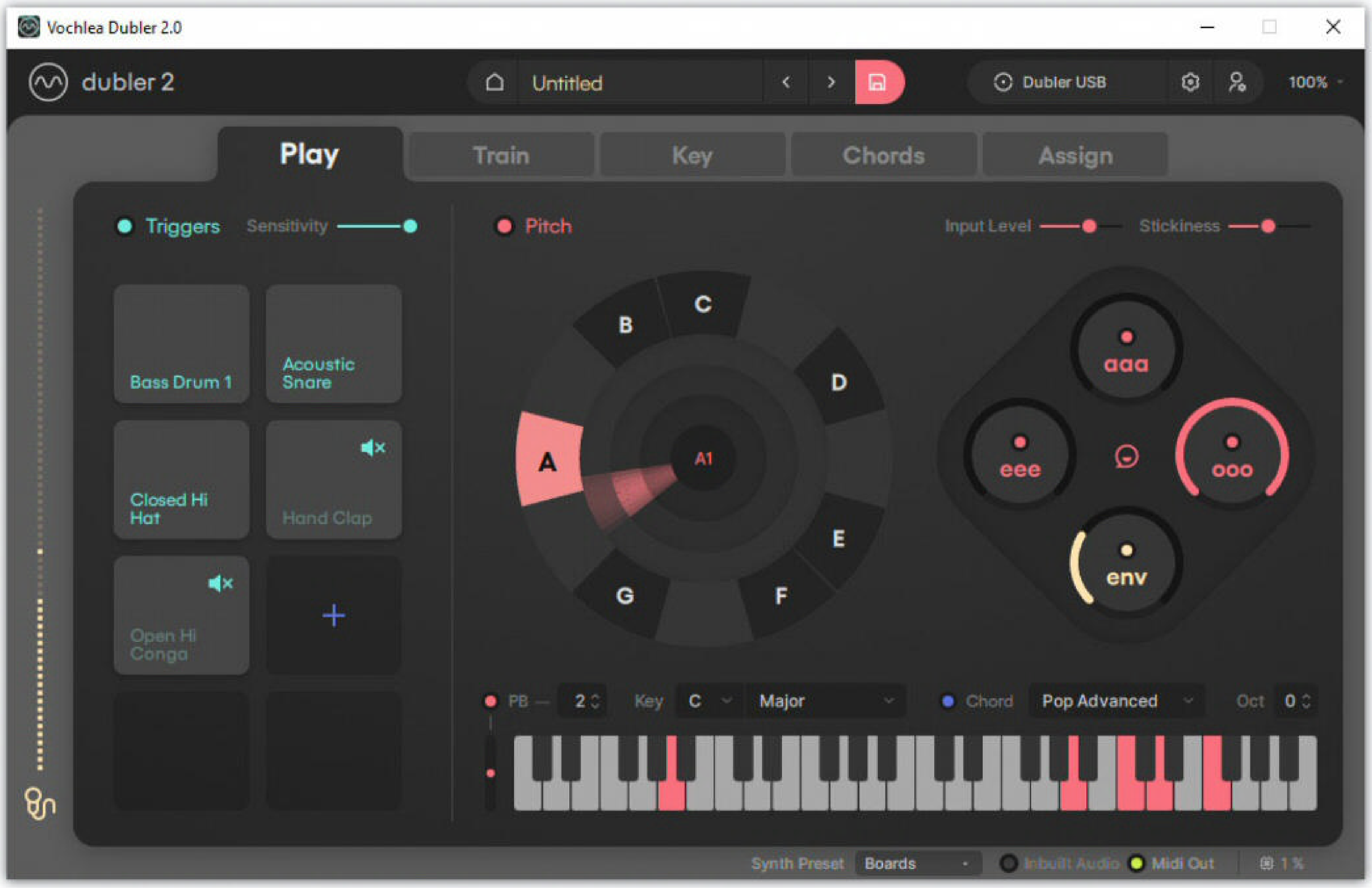
In these two shots of the Compressor plug-in bundled with PreSonus Studio One, the effect of different Knee settings is visualised. With a hard knee setting (top) compression occurs immediately at full ratio once the threshold is reached, but not before; with a softer knee (above) the onset of compression is more gradual and begins before the threshold is reached.

by placing a conventional compressor on an auxiliary channel and sending to it from the dry source channel. The relative levels of the faders on the source and auxiliary channel will then determine the balance of dry and compressed signal.

Compulsive Compression

Like other common processors such as reverb or equalisation, compressors sometimes offer an endless variety of additional controls on top of their core parameters. Sometimes you'll also find familiar knobs bearing unfamiliar names. Most of this, however, is icing on the cake. Once you understand threshold, ratio, attack and release, and how they interact, you'll be a long way towards being able to use compression to get the results you want.

So, my advice to anyone starting out is to focus on these core controls first — or rather, second. The first thing should always be to ask yourself why you're using a compressor at all, and what result you're aiming for. When we have unlimited plug-in power at our disposal, it's easy to get into the habit of applying compression purely for the sake of it. But, like any process, compression is a means to an end, not an end in itself. It's much easier to choose the right settings when you know what you're trying to achieve with them!



The Play view, which is where you'll spend most of your time once you've set everything up to taste.

Vochlea Dubler 2

MATT HOUGHTON

Voice-to-MIDI Software

Not everyone can play keys or guitar, so how about contolling virtual instruments with nothing but your voice?

Making music with computers can be frustrating if, like me, you're not a great keyboard player. There are various alternative controller options, not least for guitarists like me, but I've long been interested in the idea of using my voice to control instruments and effects. It's not purely laziness, either: the human voice is a uniquely expressive 'instrument' and most of us have a decent degree of control over it. I've tried various audio-to-MIDI systems over the years — both real-time and offline types — but, as much as I've enjoyed exploring them, I'd not hit upon a single program that made it easy until very recently.

Several months ago, I found myself contemplating Vochlea's Dubler, which promised to turn my voice into expressive MIDI performances, in real time, with low latency. Better still, it claimed to track not only the pitch and level, but to include a dedicated beat-boxing facility, to trigger chords and, best of all, to create MIDI data from different vowel sounds. So I contacted Vochlea, who sent me their Studio Kit 1 for evaluation, while asking if I might review their more ambitious Dubler 2, the release of which was near.

Dubling Up

It's worth starting with a quick summary of my experiences with Dubler 1, which was available only in the Studio Kit, comprising the Dubler 1 software and Vochlea's USB dynamic mic. Compared with most real-time pitch-to-MIDI systems I've used, it was impressive in many ways, and it was undeniably fun to play with. But it didn't really meet my needs. The beat-boxing was particularly reliable, but the voice tracking was, if anything, too precise; I'm sure skilled vocalists coaxed better results from it, but I tended to find that my own, slightly pitchy efforts too often triggered bum notes or unwanted bends, and there was only limited scope in the single-screen interface to hone its response to compensate for my lack of vocal control.

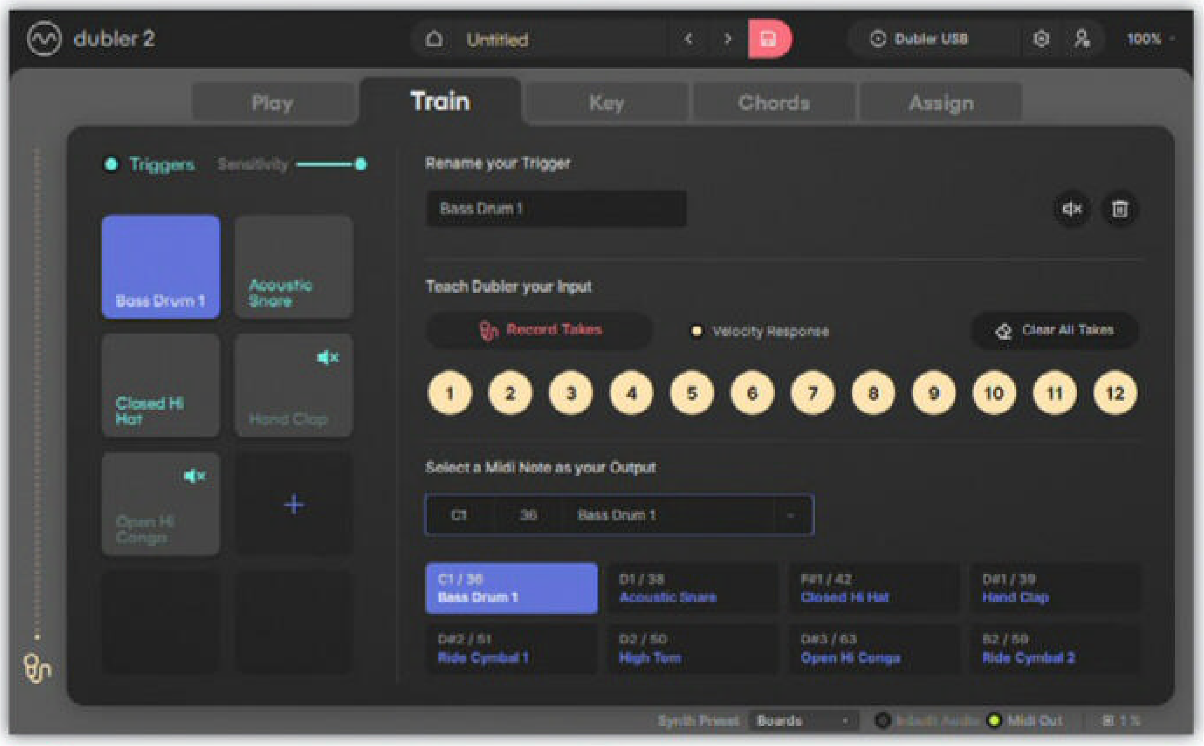
Available for Mac OS and Windows, Dubler 2 is a significant step forward for people like me. Significant enough, in fact, that I'm now using it to 'sing in' ideas quite often, not least if I want to experiment with adding strings and the like to my rock/pop arrangements — I can just sing, hum or whistle simple parts very easily. It's available in the Studio Kit 2 or, new in version 2, as software only. Vochlea recommend their own mic, which was used when refining their algorithms and is automatically recognised by Dubler

2 when connected, but any dynamic mic with a 'professional' audio interface (for Windows, that means one with an ASIO driver) will work. If Vochlea's mic isn't hooked up, an idiot-proof setup wizard configures Dubler for use with your mic.

Trigger Happy

Dubler 2's GUI is very different from that of Dubler 1, with the single screen replaced by five tabbed pages. It never feels busy or daunting, though, and once set up you'll spend the vast majority of your time in the Play tab. Accordingly, Dubler 2 opens in this tab which, as the name implies, is the main performance view. It's divided into two sections: a pane on the left for beat-boxing, and a bigger one to its right dedicated to the pitch and vowel tracking.

Click a Trigger pad on the left and you're taken to the Train tab, where you can 'train' the software to recognise the beat-box sound you want to map to the pad. Training is a really intuitive process with good visual feedback, whereby you record up to 12 versions of the sound in one pass. You can name each pad ('kick', 'snare' and so on), assign it different sounds from the onboard 808-style



You can teach Dubler 2 to map beat-box sounds to its Trigger pads, or you could use it with other sources too, for example to generate snare rimshots to trigger electronic sounds.

drum kit or a different MIDI note (Dubler presents itself to your DAW software as a MIDI device), or both, and can specify whether the output should be dynamic or play back at a single velocity. Up to eight pads can be created per profile (you can save and load different Profiles, which is helpful if you want to use different mics, or want a regular collaborator to use Dubler), and then it's back to Play view to start your beat-boxing performance.

The beat-boxing experience is nice and snappy; I never felt distracted by latency. Triggering a three-piece kit with, say, kick, snare and hi-hat is trivially easy, and that'll be enough to create a groove on stage (more than enough if you team it up with a looper), while in the studio, you can easily go back and overdub more parts. As you start to add more pads, though, you'll need to give thought to the sounds you're 'singing' because, unless they're sufficiently differentiated, you'll trigger 'false' hits, and you might therefore need to adapt your natural beat-boxing style. Personally, I got consistent results with four pads pretty easily, but with more, I found it trickier to generate parts that didn't need tidying in the DAW. On the whole, I'd say the beat-boxing is a real strength of Dubler 2.

Pitch Perfect

The (monophonic) pitch-tracking is accurate, with just a hint more latency than with the beat-boxing but not so much as to be distracting. That said, sometimes I found it reacted more quickly when I sang an octave up (you can transpose the output up/down in octaves).

What really sets Dubler 2 apart from the pitch-tracking crowd are the ways it allows you to fine-tune its response to the incoming signal, and the visual feedback. The latter takes the form of a large 'note wheel', which displays both the current pitch being played and how far you are from hitting that pitch centre. Triggering can be restricted to keys/scales, and there are many more options here than in Dubler 1. There's no support for such exotica as quarter-tone intervals, but in online demos Vochlea »

Vochlea Dubler 2

£189

PROS

- The basics are very easy, and beginners can enjoy instant results.
- Beat-boxing triggers work very well.
- Responsive to three different vowel sounds, as well as signal level.
- Can trigger melody and chord parts simultaneously, on two different MIDI channels.
- Multitouch-friendly.
- Now works with any mic.

CONS

- Can be tricky to hear yourself 'sing'.
- While the basics are easy, getting the most expressive performances out of this system requires skill and patient practice.

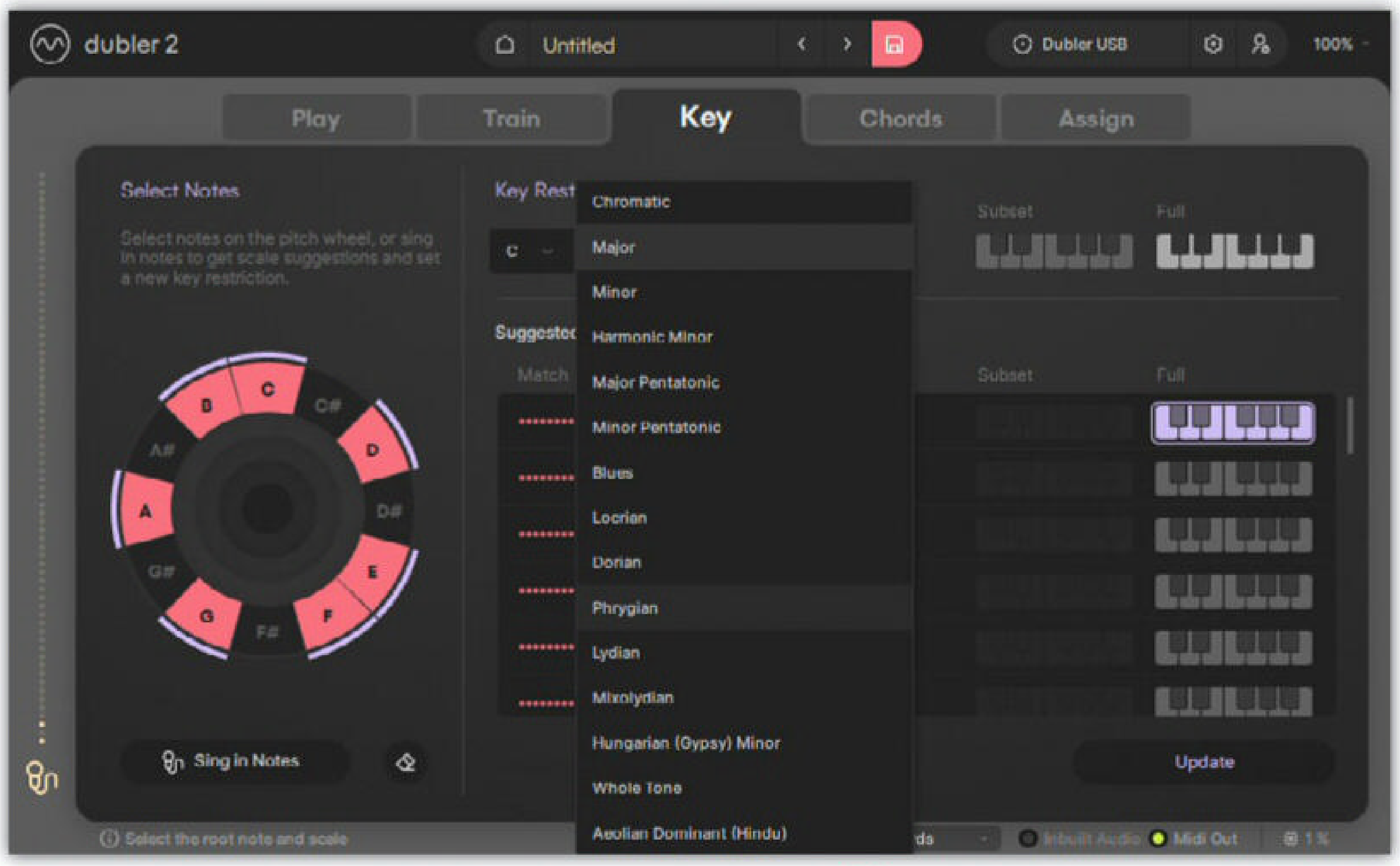
SUMMARY

The first system for control virtual instruments using nothing but my voice that I've found to be genuinely usable, Dubler 2 is also immense fun. It's easy for anyone to get up and running, but perfecting control of it requires skill and practice.

have hinted that they could add more options in the future. You can also 'mute' detection of individual notes, on the note wheel or the keyboard below it, and this can make it less challenging to trigger only the intended notes. I was pleased to discover that the GUI supports multitouch: if you have a multitouch screen it's incredibly easy to mute and unmute notes during a performance. The GUI can also be scaled up to 200 percent of the default size.

The Play tab's Pitch section has an input level control, and getting this in the optimum place does seem to help the detection. More interesting is the Stickiness slider, a sort of sustain/release control that smooths the transition between detected notes, reducing the likelihood of errors. The optimum setting varies according to the part you're playing and the way you're singing, but it's a really helpful feature.

What all this adds up to is that, with only a little practice, pretty much anyone who can vaguely hold a note should be able to use Dubler to play uncomplicated melodies, or, say, to trigger simple string backing parts to embellish a rock song. And while it obviously helps if you know the difference between minor and major keys and so forth, you don't need to: sing notes in, and Dubler can suggest a key.



The pitch detection can be limited to notes in various keys or scales.

Onboard Sounds

A big improvement in Dubler 2 is that it doesn't just spew out MIDI: there are onboard sounds too. They're very usable on stage but, just as importantly, they allow you to configure and fine-tune Dubler without having to map the MIDI output other software. The 808-style drum kit is perfect for beat-boxing, and there are a handful of lead, bass and pad-style synth presets. Most have modulation and delay in the patches, which novices will love — the movement and delay tails can help to cover a multitude of pitching sins! What's here is thus very welcome, even if most people will soon decide that they want to hook Dubler up to their favourite soft synths and sample-based instruments.

You can be much more ambitious, of course, though note that the more you ask of Dubler, the better a 'singer' you'll need to be. I use inverted commas around 'singer' because I mean specifically that you must have or develop very good control over your own pitch; you won't need a 'golden voice'! First and foremost, it's about hitting the right pitch, but you might find that you need to adapt any naturally bad singing habits. For instance, I have a tendency to bend up into the pitch at the start of words, and Dubler can interpret that as two different notes. The aforementioned key and note-muting facilities help, but I got good results without that by deliberately trying to sing as if I'd already been Auto-Tuned! Changing the sounds you're using can help, too. For example, if I sing a sustained 'aaah', I seem to bend into the note more than when I sing 'oooo', and if I add a 'w', to make it 'waah', this seems to improve matters. Dubler's response also drew my attention to my tendency to let the pitch drift on sustained notes, as I 'run out of wind'. The more you practice and adapt your singing technique, the easier it is to get Dubler to do what you want.

Dubler can not only play melodies, but trigger chords too, and can do so alongside the melody. You can output the two parts on different MIDI channels, and, better still, you can activate pitch-bend for either, neither or both parts. Speaking of pitch-bend, you can restrict the bend range in semitones from 1-12, and a new Intellibend mode 'anchors' to a note until you deliberately bend away from it, it's precise behaviour apparently relating to the pitch Stickiness setting.

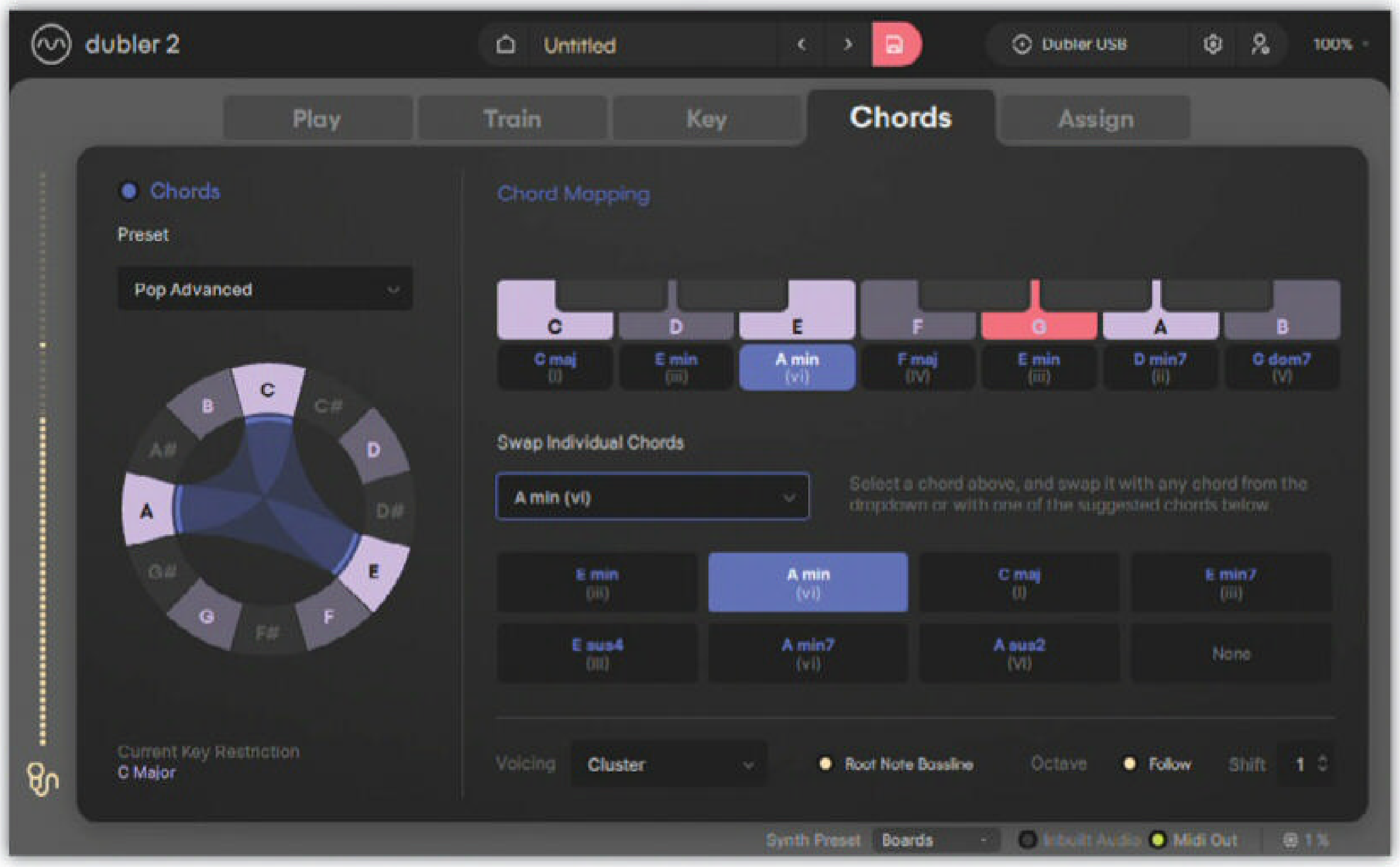
By default Dubler 2 maps notes to chords that work in your chosen key, but you can assign chords manually. Between this and a handful of other options there's scope to create a convincing auto-accompaniment for your main melody. I'd personally be less inclined to play two parts together like this in the studio, where I can easily overdub, but it opens up tantalising options for live performance. If you're stuck for inspiration, you could also use Dubler to suggest chords that will fit your melody.

Oohs & Aahs

In the Play tab, a rotated 'squircle' is populated by four circular meters, one each to track the level of three vowel components and another marked 'env', which tracks the amplitude. The vowel trackers, labelled phonetically ('aaa', 'ooo' and 'eee'), can be activated/bypassed by clicking on the meter, and you can fine-tune their response in the Assign tab. Crucially, you can set minimum and maximum thresholds to suit your vocal character, and you can set the output MIDI CC range in a similar fashion. This new facility makes Dubler 2 so much more expressive than its predecessor.

In the same tab, you can also change the MIDI CC each tracker outputs, though you can't assign them to a different MIDI

Dubler 2's Chords tab.

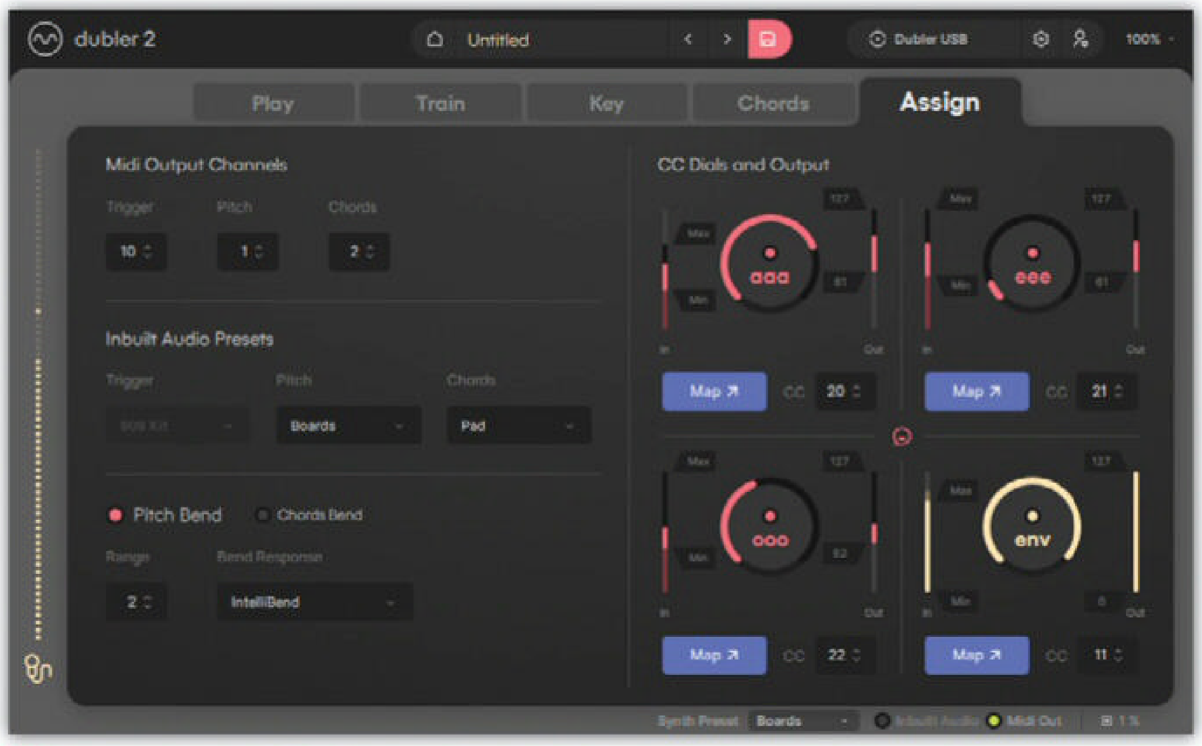


channel from the Pitch tracker, which seems an obvious option to include in an update.

Again, there's a certain amount you can do easily using the vowel trackers, but more becomes possible with practice. It's not hard, for instance, to 'play' (sing) in a synth part and use two vowels to control the frequency and resonance of the synth's filter. Controlling more than that in one go will require more control over your voice. Essentially, Dubler is like any other instrument: the more precise the control you require, the more you must practice. But it's not always about precision. For example, I mapped all four of Dubler's vowel/amplitude trackers to different parameters of a fairly static synth pad and was rewarded with a lovely sense of movement.

Hearing The Voice Inside

One quirk inherent in any real-time voice-to-MIDI system is that it can be tricky to hear your voice when listening to the part you're playing — rather like when singing over a loud band without enough 'me' in the foldback. Sometimes this makes it harder to hit the right pitch or hear the changes in timbre, both things that Dubler can react to. Practice undoubtedly helps, but there are other things you can do to improve matters. For example, you obviously want to hear what you're playing, but does it need to be loud? In the studio, I found it easy to monitor on open-backed headphones at a modest level. This made mic placement less critical, as my vocal wasn't fighting with the sound over the speakers, and I could still hear my voice acoustically in the room; closed-back headphones work too, though I heard my own voice less clearly, and that tended to have an impact on my singing. I wonder whether it might help if, in the future, Dubler were to allow you to blend some of your vocal signal into your monitor mix. All this was much less of an issue when beat-boxing, by the way, which worked just fine, even with the speakers cranked. On stage, in-ears might work, but if you must have sound coming over speakers, the trick is to make sure you use the mic up close, to keep spill from other sounds low relative to your vocal, and calibrate Dubler accordingly; presumably this is one reason Vochlea recommend dynamic mics.



In the Assign tab you can, amongst other things, refine the behaviour of the vowel and amplitude trackers and assign them different MIDI CC numbers.

Lasting Impressions

I wasn't really sure what to expect when I first watched the promo videos of Dubler, and my reactions to Dubler 1 were mixed: it was a fun experience, but ultimately something I wouldn't have bought. Dubler 2 changes that, and has genuinely impressed me. OK, so it's not the 'brain-to-USB interface' that my heart truly desires — no voice-to-MIDI system can be — but, while there's some scope for further development, Vochlea have delivered a genuine option for anyone who can't play keys and wants to sing, hum, beat-box or even whistle some parts down. Even those who do play keys might use it for real-time control of effects. Put in enough vocal practice, and it could open up new creative vistas on stage or in the studio. It's definitely worth checking out the demo.

£ Software only £189. Dubler Studio Kit 2 (includes USB mic) £249. Upgrade from Dubler 1 £59. Prices include VAT.

W https://vochlea.com

How I Got THAT SOUND

Andrew Maury

Lewis Del Mar 'Loud(y)'

JOE MATERA

New York-based engineer and producer Andrew Maury spent years mixing front of house on tour before making the transition into studio mixing. Since then he has worked with a diverse group of major-label and independent artists including Lewis Del Mar, Shawn Mendes, Youngr, Post Malone, Kimbra and the Kooks. He also mixed the 2020 Grammy-winning song 'Jerome' by Lizzo. Asked to trace the story of his favourite sound, he nominates the track 'Loud(y)' by pop duo Lewis Del Mar.

"This track is both a favourite production of mine and something that caught a lot of people's attention. It was the band's first released single and it also garnered a music industry frenzy and a lot of press. The song is about how noisy and verbose the world has become — particularly on the internet. The band's original self-recorded demo was quite calm in its production style. It had drums with the snare wires off and everything was acoustic and played quite delicately. There was an interesting irony in that approach, but we ended up injecting a ton of distortion and power into the song to more overtly elevate its message."

Different Drums

"The original demo drums can be heard for two measures in the beginning of the song. After the first lyric 'Can you please sit the fuck down...' the re-recorded drums drop in for an explosive moment. Those new drums were recorded with four mics on a basic kit with no toms in a small, dry room. We used an AKG D112 inside the kick, an NS10 speaker as a 'sub kick', an SM57 on snare top, and an old RCA 74B ribbon mic aimed at the side of the snare drum and hi-hat, probably somewhere under the crash cymbal. On their own, the recorded mics sound nothing like the final mix.

"A majority of the drum sound comes from the RCA, which has a very metallic and colourful midrange. That mic is smashed up with the Waves TG12345 Channel. All of the mics then feed a drum bus with a pretty aggressive chain. I used some basic EQ, FabFilter Saturn to control/gate the low frequencies, Kush Audio Pusher to create a unique distortion texture which sounds alien and transient-rich, and an old CBS Labs 4450 broadcast limiter as a final saturation stage in the analogue domain.

"This drum mix is all about the interaction of a finely tuned chain of processing. Changing the volume of one mic or tweaking the compression timing could totally throw off the energy. I probably tweaked the nuances of the chain for an hour before finding the chemistry that gave it groove and compatibility with the vocal."

Hear The Sound

W https://open.spotify.com/track/5DOadSIDaKFwUPcbcbDv5J?si=be4e626d9b4e49fc

W www.youtube.com/watch?v=vVFwuq44UTQ

Back & Forth

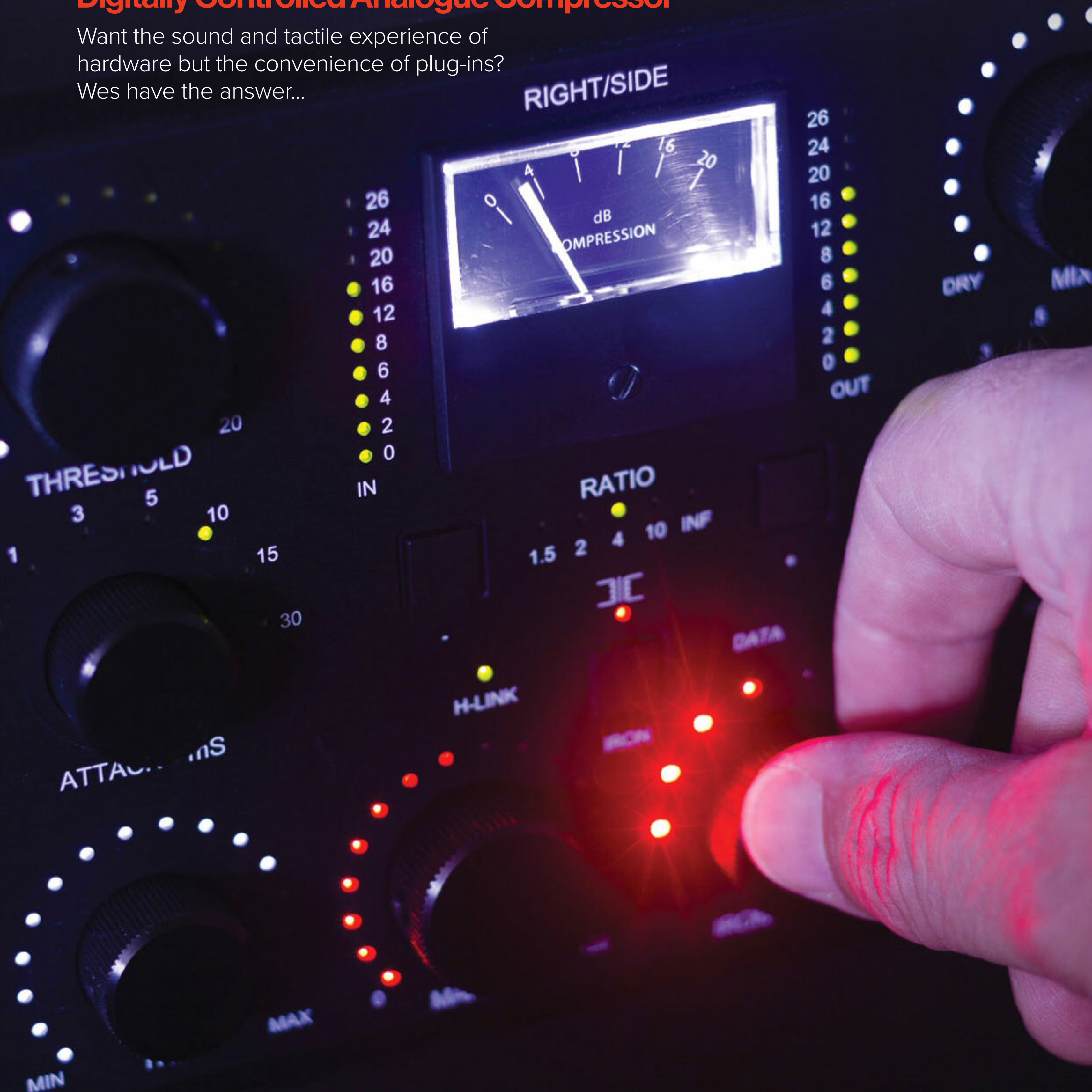
"Another staple in the soundscape of 'Loud(y)' is the contrast between the distorted guitar stabs and acoustic guitar slides in the chorus. It's a pretty intense call-and-response dynamic that helps the slow tempo of the song feel like it has a lot of momentum. The guitar stabs are single notes played on the low 'E' string of a Strat into a Boss OC3 octave pedal, to generate an octave down, into a Zvex Fuzz Factory distortion pedal, to both gate the signal and bring out a really fizzy distortion that matches the character of the drums. The acoustic guitars are roughed up with SoundToys Decapitator and a lot of compression.

"It was fun to push sonic boundaries and create unique textures on this one. The result is a song that cut through the noise — as loud as it is out there."

Wes Audio ngBusComp

Digitally Controlled Analogue Compressor

Want the sound and tactile experience of hardware but the convenience of plug-ins? Wes have the answer...



MATT HOUGHTON

Regular readers will know that I've been smitten with Wes Audio's digitally controlled analogue gear and that I've evaluated various devices inspired by the SSL 4k bus compressor. The ngBusComp brings these worlds together in the most impressive reimagining of the SSL design I've yet encountered. As with the other ng ('next generation') products, this analogue compressor can be controlled using front-panel buttons and encoders, a dedicated DAW plug-in, or both. The communication between them is bidirectional and they never feel out of sync.

Overview

The build quality is flawless, inside and out. A black, 3U chassis is adorned with a substantial, brushed-finish, metal panel which sports many controls, all a pleasing size and generously spaced. The encoders' knurled knobs have just enough 'stiffness' to provide useful tactile feedback and several are touch-sensitive, their LED indicator

Wes Audio ngBusComp
£2899

PROS
- The hardware looks, feels and sounds great.
- Immensely versatile, with roles in tracking, mixing and mastering.
- Best-in-class DAW integration.
- USB and Ethernet options as standard.
- Can be ultra-clean or as colourful as you want.

CONS
- Only that I can't afford it!

SUMMARY
An incredibly versatile VCA compressor, the ngBusComp has bags of character on tap and applications that go way beyond bus compression. It also boasts Wes Audio's DAW plug-in remote control system!

The ngBusComp has two separate analogue channels, which can be configured as L-R, M-S or dual mono, and the digital controls can be linked and unlinked at the push of a button.

rings becoming brighter the instant you lay fingers on them. That's a nice touch, which means the rest of the front-panel 'light show' can be left tastefully dulled. Speaking of light shows, the metering sections comprise not only the expected moving-coil gain-reduction meter but, either side, 10-LED input and output level meters too, each ranging from 0 to the maximum 26dBu.

On the back, male and female XLR connectors cater for the balanced line-level I/O and two TRS jack sockets for the external side-chain inputs. An IEC inlet is switchable for 115V or 230V mains, and USB Type B and RJ45 Ethernet ports allow communication with the plug-in.

Lifting the lid requires reveals neat PCBs, populated by a combination of SMDs and through-hole components. The business end of the compression circuitry is classic SSL, with THAT Corporation's THAT2181 chips used for the gain reduction and in the detector circuit. These chips aren't inexpensive, and some 'SSLish' designs cut costs here: the better designs, including ngBusComp, use four in parallel in the signal path, helping to keep the noise floor low.

The front-panel's biggest giveaway that this is more than a clone is not the additional functions, but that there's a separate control set for each channel, with each laid out identically from left to right. Between the two sit some backlit buttons, some of which have a secondary press-and-hold action. Uppermost is the Parameter Link button, with three more beneath to configure the compressor as a Dual (mono), (L-R) Stereo or M-S (stereo) device.

When Parameter Link is engaged, each control operates on both channels, LEDs lighting on both accordingly. Linked changes are relative to the existing settings, rather than one channel forcing the other to its position. You can also copy one channel's settings to the other: press and hold the Dual button, wait for it to blink, and press the M-S button to copy from channel 1 to 2, and vice versa for 2 to 1. Holding a touch-sensitive encoder for both channels simultaneously temporarily disables parameter linking. It's a neat system, and the sort of feature that makes reading the detailed PDF manual worthwhile.

Each channel's Bypass button doubles as a channel mute: press to toggle the compression circuit in and out (hard-wired bypass) or press and hold to mute the signal. Since there are only two channels, it's possible to use the mute to solo the Left, Right, Mid or Sides channels.

Three more buttons, labelled simply A, B and C, allow saving and recall of hardware presets to compare settings: press and hold the preset you want to copy, then press to overwrite the destination. Then press to switch between them. Simple. You can store many more presets using the DAW plug-in, of course.

It Takes Two

Each channel's control panel sports eight rotary encoders and three buttons. The Threshold knob does what you'd expect, and on the other side of the meter section sits a post makeup-gain wet/dry mix control. The in/out meter LEDs show increments of »

» 2 or 4 dB, depending on the level, and the coil meter can indicate gain reduction up to 20dB. Beneath the meter, two buttons increment the Ratio between five settings, each with status LED: 1.5, 2, 4, 10 and INF:1, the last being hard limiting. Another button engages Iron mode, of which more shortly, and either side are knobs for the attack (0.1, 0.3, 1, 3, 10, 30ms) and release (0.1, 0.3, 0.6, 0.9 and 1.2 seconds and Auto).

A side-chain filter can be set to Off, 60, 90 or 150 Hz, or T1 or T2, the last two combining tilt EQs with high-pass filters. That's plenty of scope for fine-tuning the compressor's response, but should you need more, the last position engages the external side-chain.

It's not uncommon for SSL-inspired compressors to offer ways to inject colour, but the ngBusComp takes it further than most, with two different approaches, each beautifully controllable. First, a THD control introduces harmonic distortion, from 0 to 8 percent. Wes say it's a proprietary circuit and, while I'm not sure what it does differently from other such circuits, it sounds nicely musical and offers good, precise control, making it easy to introduce anything from subtle enhancement to obvious attitude. Yet more colour is available if you engage Iron mode. A red LED indicates when you've done so, and LED rings around the two associated knobs turn from the usual green-tinged yellow to red. Iron mode puts the makeup gain stage before the Carnhill output transformer, and introduces a pad (15dB in 1dB steps) after it, so you can drive the transformer into saturation without the levels getting silly.

It's worth noting that the makeup gain operates only on the wet signal and the pad only applies in Iron mode. So if Iron isn't engaged and you want to change the wet/dry mix without affecting the overall level, you'll need to juggle this with the makeup gain (instant comparisons are still possible using the presets). Another slight limitation is highlighted in the manual: in M-S mode, you can't use different Iron settings on the M and S channels, as the signal is converted to L-R before the output. You could use a separate M-S encoder/decoder and configure the compressor in dual mono, though.

Plug-in Power!

The ngBusComp's stand-out feature is that it can be controlled by a plug-in: you can save and recall settings with your projects and even use automation. And the execution is excellent — it's easily the best, slickest integration of plug-in and analogue hardware that I've seen. Whether you control the plug-in from a mouse, touchscreen or using automation, it just gets on with the job with a minimum of fuss.

The plug-in communicates bidirectionally with the hardware controls, and you have the choice of USB or Ethernet connection.

The installer was a 350MB download for my Windows 10 system. You can specify which plug-in formats you want installed (Mac/Windows; VST2/3, AU and AAX plug-ins; 32- or 64-bit, where applicable) but, slightly frustratingly, can't specify which individual plug-ins are installed; you get one for every device in the ng range. The VST3 64-bit plug-in (curiously, categorised by Cubase as an EQ!) and USB driver occupied just under 170MB. Installing the driver necessitates a reboot, after which the software loads automatically and prompts you to check for firmware updates.

Communication with the computer can be via USB or Ethernet, as you prefer. USB is marginally easier to set up but that realistically means placing the ngBusComp within 5m of your computer. With Ethernet, that limitation is removed, and gives you the option of connecting to a router, for a wired or wireless link with your computer. (The steps for setting up the Ethernet link are described clearly in the manual.) Once connected, you fire up your DAW session, instantiate the plug-in, and select the hardware in a drop-down menu (you're prompted to do so). This may seem unnecessary but it allows multiple instances to control different units. A nice touch is that there are both stereo and mono plug-ins, and when using the ngBusComp as a dual-mono device, you can place mono instances on different channels. Another cool feature is that when you put your finger on one of the touch-sensitive hardware controls, even without changing it, the current value will be displayed on the GUI.

This being analogue outboard, of course, you have to hook its analogue connections up, if that's to your interface rather than an analogue chain, and to set up the routing separately in your software, for example, using your DAW's external I/O plug-in. (I'd love to see USB or Thunderbolt A-D/D-A conversion on-board in the future, and routing taken care of by the control plug-in...)

The one small limitation compared with a software-only experience is that there are inevitably some low-level clicks if you automate certain parameters or switch between presets with radically different settings. I don't regard this as a problem as it's in the nature of the technology, and it's not the sort of device you'd typically want to use in that way. But prospective users should be aware of it, nonetheless.

Verdict

The excellent plug-in integration, though, is merely the icing on this most tasty of cakes. In my decade and a half at *SOS*, this has to be one of the best compressors I've had the pleasure of trying out. It sounds great, has great technical specs, and lets you inject as much character as you desire. Along with the different channel modes, the parameter linking, the side-chain filters and range of attack/release times, the 'colour' facilities make the ngBusComp incredibly versatile — it might be called a bus compressor, but it most certainly isn't suited only to such duties, and could work beautifully in a vast range of tracking, mixing or mastering roles, with mono or stereo sources. For many prospective users, that should go a long way to justifying its price. I can't personally justify the investment at present, but I am sad to be sending it back and it now has a place right near the top of my gear wish list!

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RME Fireface UCX II

USB Audio & MIDI Interface

The UCX II packs 25 years' interface design experience into half a rack space.

SAM INGLIS

Throughout their 25-year history, RME have championed a consistent design philosophy. They have maintained a relentless focus on driver quality, reliability and product longevity. They've also been quick to embrace new technologies and protocols, but only when there's a clear benefit to the user. If you bought your RME interface in the early years of this century, there's a good chance that it is supported today. And if you buy a new RME interface now, you're benefiting from two decades of development that have produced an extremely mature product ecosystem.

The latest fruit of this development is the Fireface UCX II. This supersedes the long-established Fireface UCX, which was reviewed back in *SOS* February 2012. However, it's perhaps better understood as the little brother of the Fireface UFX II and UFX+ than as a direct descendant of the UCX. Like the UFX II, the UCX II is not only a USB audio interface for Mac OS, Windows and iOS. It's also an extremely well-featured digital mixer that can operate with no computer attached, with full control possible from the front panel. And it's a multitrack recorder, writing interleaved multichannel WAV files to an attached USB drive using RME's established DURec system.

Now UC It

Likewise, although the UCX II retains the compact 1U half-rack form factor of the original UCX, its layout owes more to the UFX II. Happily, this means that in place of the UCX's primitive two-character LED, there's now a small but colourful and detailed TFT panel that can display detailed meters for all I/O simultaneously, as well as text-based menus. This is a big improvement that, among other things, makes possible the UCX II's full standalone operation capability.

Front-panel socketry comprises two Neutrik Combo XLR/jack sockets, accommodating mic- and line-level signals, plus two further quarter-inch inputs that can accept line-level signals or act as high-impedance inputs for electric guitars. There's also a single quarter-inch headphone output. On the rear panel, you'll find four further quarter-inch line inputs and six line outputs, plus a locking socket for the external PSU and a comprehensive array of digital I/O. There are 5-pin DIN sockets for MIDI in and out, plus a pair of optical sockets for ADAT Lightpipe, supporting eight inputs and outputs at base sample rates (and switchable to carry stereo S/PDIF if needed). A single BNC connector can act as a word-clock input or output, and a supplied flying cable allows stereo XLR AES3 and phono S/PDIF to be accessed via a nine-pin D-Sub. When RME describe the UCX II as offering "40 channels",

therefore, they mean that it has a total of 20 inputs and 20 outputs.

The UCX II's rear panel also has two USB sockets, in the older A and B formats rather than the current Type C. RME's Matthias Carstens explained that there's no technical advantage to using the newer socket, and that in his experience it is less reliable. The Type B socket is used to make the connection to the host computer; cables are included to connect to either a Type A or C socket. The UCX's own Type A socket has two purposes. It can either be used to connect RME's ARC USB remote control, or to attach a USB drive for direct recording.

The original UCX was a dual-format interface that could connect either via USB or FireWire. The latter format is now effectively superseded by Thunderbolt, but RME haven't bestowed a Thunderbolt socket on the UCX II. This seems to me a sensible enough decision: building in the necessary chipsets and connectors would have raised the price of the unit considerably, and Thunderbolt is still not universally available on desktop or laptop computers. Most importantly, although there is often a clear performance gap between the two formats with other manufacturers' interfaces, RME's USB drivers are so efficient that there is little to be gained by switching. At base sample rates, the UCX II was happy to operate on my 2019 Mac Mini at a 16-sample buffer size, giving an outstandingly low round-trip latency of 2.3ms.

Free Standing

At the heart of the UCX II is an FPGA chip running the latest version of RME's TotalMix FX. This has been discussed and described many times in previous *SOS* reviews, and it would probably take a book to list all its features; it's no exaggeration to say that TotalMix FX is as capable as many dedicated digital mixers. Each physical output can receive its own dedicated balance of the signals from all the inputs, DAW returns and the built-in reverb and delay. Each physical input and output also has its own EQ and dynamics, as well as controls such as mic preamp gain where appropriate to specific input types. There's also comprehensive monitor control, metering, preset save and recall, grouping, support for remote control and more.

All of this has been part of the RME package for a long time now. What's new compared with the original UCX is full standalone operation. Eagle-eyed readers will notice that the UCX II is not endowed with faders, gain pots, mute and solo buttons and the othe accoutrements of a conventional mixer. When a computer is connected, or when a tablet running RME's TotalMix Remote software can see the UCX II on a Wi-Fi network, this isn't a problem, because everything can be controlled on screen. However, in standalone mode, hands-on control is rather more limited.

To the left of the UCX II's small but detailed meter panel, you'll find four small buttons: to its right, a rotary encoder which also has a push action. Under normal circumstances, the encoder serves as a level control, with the push action switching its focus from your designated monitor outputs to the headphone output. The four buttons each cycle through two menu pages before returning you to the metering view. The first pair of menus enables quick and easy control over preamp gain, while the second activates DURec recording and playback, and provides control over related parameters. The last pair brings up various setup and routing options, as well as control over the TotalMix reverb and echo.

Mix Mastery

However, the vast majority of TotalMix FX functions are found under the third button, labelled Chan/Mix. On the first press, this brings up a scrolling menu presenting all the global parameters for the selected input channel; settings such as mute, stereo link, polarity, M-S decoding and so on are accessed here. Initially, I thought important options such as phantom power and line/instrument switching weren't available from the UCX II's front panel, because I'd expected to find them among the Mic/Gain settings accessed using the first button. Instead they lurk at the bottom of this menu, and only hove into view when you scroll down.

Pressing Chan/Mix for a second time lets you select a TotalMix mixer page, within which you can then set fader and pan or balance levels for each channel. So, for example, if you have Mix to Line Out 1/2 and Mic 2 selected at the top of this screen, the Fader and Bal/Pan controls further down concern only the level and position of the second mic/line input within the particular mixer that is feeding this output pair.

Needless to say, you wouldn't want to set up a busy mix entirely from the UCX II's front panel, but there are certainly circumstances where this standalone functionality might be useful. For example, if you have a two-track source such as a CD player or master recorder in the studio, you might not want to switch the computer on every time you want to play something back; and in other circumstances, it's not unknown for Wi-Fi to take a dive and rob you of your iPad controller. Standalone operation would rarely be the preferred way to interact with the UCX II, but the full colour display and sensible menu layouts mean there's never any confusion about what you're adjusting at any given time.

At this point you might be thinking: this standalone capability is all very well, but I'm unlikely to need it often, and meanwhile, a single rotary encoder doesn't offer much by way of monitor control. Wouldn't these buttons be more useful if they could be used to activate mute, mono, dim, and so on? Those were my thoughts too, until RME drew my attention to a feature called Key Remap, which is somewhat buried in the UCX's extensive PDF manual. Accessed from the nether reaches of the Setup/Rev menu, it allows you to assign your choice of 20 or so different TotalMix settings »

RME Fireface UCX II

£1149

PROS

- Extremely powerful built-in mixer with remote control and standalone operation.
- Driver performance that rivals Thunderbolt and PCIe interfaces.
- Good audio specifications and subjective sound quality.
- Small but clear and comprehensive colour display.
- Built-in multitrack recording to an attached USB drive.
- Key Remap allows buttons to be reassigned in useful ways.

CONS

- Comparatively expensive.
- Only two preamps and one headphone socket.
- Talkback and some monitor-control features can't currently be controlled from the front panel.

SUMMARY

The latest version of RME's half-rack interface packs in all of their latest technologies, offering full standalone operation whist retaining the core virtues for which the company are well known.

The TotalMix Remote app is compatible with all iPads going back to the second generation.

» to each of the buttons. Once you've done so, a momentary press toggles this setting, and a sustained press brings up the menu as usual. This works well for functions such as mute, dim and low cut, but at present it can't be assigned to TotalMix features that aren't available in standalone mode, such as talkback and speaker switching. If I was to use the UCX II long-term, I think I'd want to invest in an ARC USB or other hardware controller to access these options.

Future Proof

Like many of RME's interfaces, the UCX II is designed partly as a platform for expansion. Hence, for example, it has only two built-in mic preamps, but offers six line inputs and plenty of digital I/O in case you want to add more. Perhaps uniquely, it permits AES3, coaxial and optical digital I/O to be used simultaneously — whether the latter is set to S/PDIF or ADAT — so there are no caveats to its stated channel count. The on-board preamps use RME's latest design, which offers a 75dB gain range digitally controlled in 1dB steps and sounds very good indeed. The headphone amp is likewise a cut above what you'd find in many interfaces, and the line-level I/O both specifies well and sounds clean, clear and quiet. The line outputs are DC-coupled, so they can be used to output control voltages to drive modular synths.

Given that you need additional hardware such as the ARC, headphone amps and mic preamps to really exploit all the UCX II's features, though, it's reasonable to ask whether it offers value for money. After all, MOTU's Ultralite Mk5 offers very similar connectivity, with equally impressive audio specifications and an additional headphone output, for little more than half the price; and although it lacks the UCX II's audio recording features, it's functionally similar in other respects. Vary the feature set or form factor a little more, and there's no shortage of competition from other manufacturers, too.

I think RME themselves would be the first to acknowledge that in a race to offer the most ins and outs for the lowest cost, they're simply not participating. Nor does a product like the UCX II have a single, eye-catching headline feature that makes it unique. But there are nevertheless plenty of reasons why people buy RME interfaces rather than apparently cheaper alternatives. One is the incredible flexibility of a product like the UCX. From built-in M-S decoding to OSC support, from software-switchable word-clock termination to full ASIO Direct Monitoring implementation, it positively bristles with niche features of the 'you don't need this... until you do' variety.

But probably the UCX's biggest selling points are qualities you won't find on feature lists or spec sheets. Product longevity is an oft-neglected but important virtue, and RME have no peers when it comes to supporting older products. Spending a little extra on the UCX might well mean saving much more in the long term. Likewise, although other manufacturers have upped their game in recent years, RME remain the standard by which driver quality is judged. If you want to be certain of getting an interface that's reliable, offers exceptionally low latency and works well with the widest range of host machines, you won't be disappointed.

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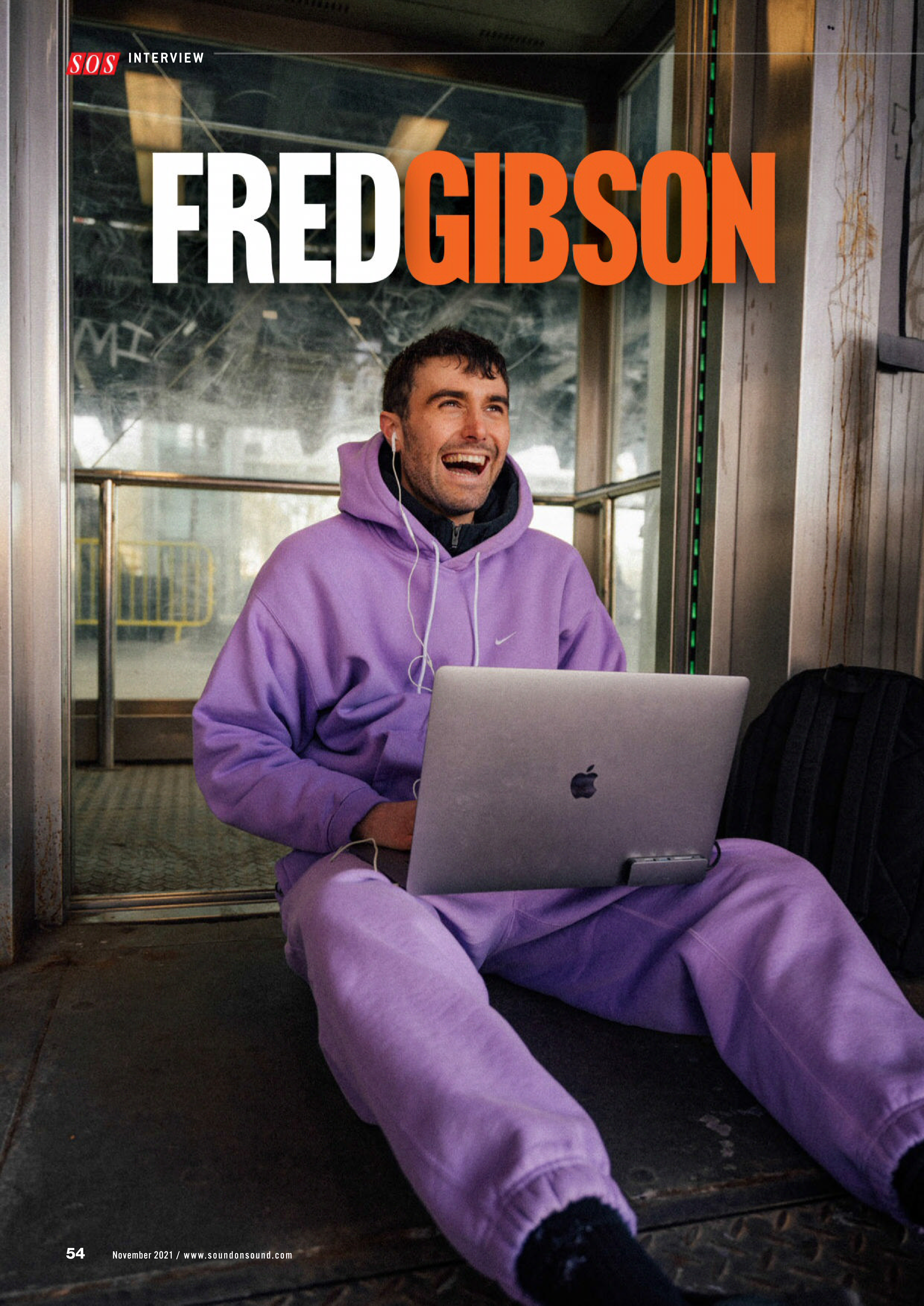
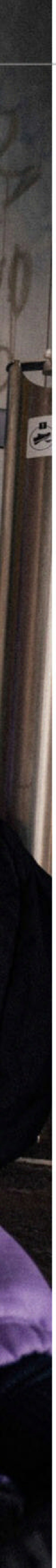




TOM DOYLE

Producer Fred Gibson, aka Fred again.., studied under Brian Eno and has worked with some of the biggest artists in the world. And his favourite microphone is the one built into his iPhone.

A dozen years into a music career that has thus far involved an apprenticeship with Brian Eno, writing and production credits for Ed Sheeran, Roots Manuva and Stormzy, a Brit Award for Producer Of The Year in 2020, plus his own solo career under the name of Fred again.., Fred Gibson is someone who very much likes to go his own way. As often working out and about and on the move on his MacBook as rooted in either of his studios — a home setup in Waterloo, central London, or his room at Promised Land Studios in Maida Vale, West London — Gibson's emphasis is very much on productivity over location.

"I'm like an anti-studio guy," he laughs. "I personally don't love studios. For what I do, I feel so empowered by the fact that I can do it wherever I want in the world with just my laptop, and my phone. I like the freedom of just being able to go anywhere and load up any song and make music really freely."

Gibson discovered Logic as a young teenager and hasn't been tempted away from it by any other DAW. "I tried, when I was about 18, to retrain myself in Ableton," he says. "Because I was watching people use it and there were so many things they could do so easily that were trickier in Logic. But I realised that I was fighting what had become my primary language. And it just wasn't a constructive use of my time.

"Logic over the last few years has added to its toolkit most of the stuff that I was wanting that Ableton had," he adds. "But, really, I think software is just a vehicle. You want it to get out of the way, so you can do your ideas. I've found no link between great music and software."

Logic is also Brian Eno's preferred DAW and so it was the key to Fred Gibson, only 16 at the time, being taken on by the former Roxy Music man, ambient music pioneer, and U2 and Coldplay producer to work alongside him as his assistant in his West London studio. As a kid, Gibson had studied classical piano, but it was his interest in recording that fired his young imagination. Initially, he began experimenting at home using a Boss BR1600CD 16-track digital recorder, before his music teacher at the time introduced him to Logic.

Nonetheless, even though he immersed himself in Logic every day for a couple of years, it took a friend to point out that Gibson hadn't yet even started using the program's software instruments. "I just thought it was an audio recorder, like the Boss thing," he grins. "I'd never really heard of MIDI. So, yeah, Logic repeatedly blew my mind and it's been a love affair that's lasted."

Inspired, Gibson created what he calls an "electronic symphony" — 90 minutes long and involving rappers, singers and a 50-piece school orchestra — which was passed on to Eno via a mutual friend. It bagged Gibson the assistant job. "I was 16 and I was the classical guy who was doing hip-hop. So that was kind of my USP as well as being the guy who could do things in Logic quite quickly.

"Brian would be the first to say he's not classically trained. That isn't how his brain works... to his advantage. But when I first got there, I remember being kind of confused. Brian would play this thing and be like, 'Oh my God, listen to that... What's that?' I'd be like, 'Oh, it's just a C major chord.' He'd say, 'Yeah, but listen to it.' It's that childlike naivety that everyone is so desperate to keep, but that musical theory can sometimes threaten to tarnish.

"So, it was a beautiful, long learning curve to sort of really try and understand the extent to which the theory can be a tool, but nothing more than that. The real goal is to listen with the innocence of any person who's not thinking about anything other than how that thing sounds. And that's a thing that I think every musician is endeavouring to keep."

Gibson experienced Eno's unorthodox approaches and methods when working on two albums the seasoned producer made in collaboration with Karl Hyde of Underworld in 2014: *Someday World* and *High Life*.

"I remember Brian holding this massage ball and creeping up behind Karl as he was doing a guitar take. I was holding the mic next to Karl's guitar. And Brian comes up behind him and starts smashing the guitar with this thing. I'm just holding the mic, like, 'Everybody stay calm.' There were lots of funny moments like that."

Jam Hot

By 2015, Fred Gibson was fast progressing as a producer in his own right. His first major credit came with Roots Manuva's ninth album, *Bleeds*, after he met the manager of the London rapper (real name Rodney Smith). "I was like, 'Please let me have a go. Send me one a cappella. Let me just try like one thing.' He sent me an a cappella of the song 'Cargo' and I made 10 versions of the song from that. The same thing with the song 'Hard Bastards'.

"Then they put me in a session with Rodney. Literally within an hour of our first session we wrote 'Fighting For?'. That was just like a jam on the piano with him. The best way I find working with rappers is to try and thrive off of the jam spirit, because most of them have come up from some school of freestyle. They're quite comfortable being put on the spot. So, there's no, like, investment, no egos getting hurt. You just jam.

"'Fighting For?' was jamming piano and him chatting around on the mic. And then at like, 35 minutes, a riff comes, and he starts singing that. And then it's a very pure creation moment. As opposed to sitting there, pausing, playing, pausing, playing the tune, and having to be like, 'Oh, no, that's not a good idea.' You don't have to say anything. You just speak through music. So, it's a much more pure type of creation, I find."

When working on hip-hop beats, Gibson tends to exclusively use NI's Battery plug-in and his own sample library, which he's constantly updating. "Every two months or so I'll just make a new set of custom kits and instruments for myself," he says. "On a lazy Sunday where you don't feel inspired, you don't really have to have much in the tank to do that. And it fuels you behind the scenes. When you are then really motivated, you got these 10 kits that have all the sounds.

"I like Battery as the host, because it's the quickest way you can easily pitch them and add compression, and there's a good master compression bus for them all. It's got all the stuff I like in terms of being able to pitch a sound down like 20 semitones really quickly and mash it through tape saturation. And the ADSR is also really important. Because 90 percent of drum manipulation, I find I just do with the ADSR [*envelope*]."

»

Synth bass-wise, Gibson tends to stay in the box, using Logic's ES2 software instrument. "My main feeling with bass is there is nothing purer and louder and bassier than a sine wave. Maybe I'll overdrive it a little bit, using the Logic Overdrive.

"All of the best bass things to me are where the song is not getting in the way of the bass. If a track doesn't feel bassy, it's not that you need to add stuff. You need to take away the things that are getting in the way of that absolute bass heaven [*laughs*]."

Studio Tools

When working in his home studio or at Promised Land, Fred Gibson simply plugs his laptop into a screen and uses his Focusrite 4i4 interface, mainly because it's portable. "I've got an Apogee Duet, and I've got an RME somewhere at the studio as well. But it's more like engineers I've worked with have bought them. For projects, it's not a thing I feel. I have tried doing blind tests on converters and things like that, and it's never resonated with me."

In terms of monitors, at Promised Land, he switches between his ATC SCM25As and his Mackie HR824s. "The 824s I've used for 10 years and I love them," he says. "I think you just need to know the speakers well. I haven't found the link between great-sounding music and absolute perfect monitoring. Like, I've had lots of people tell me that they think 824s have got a really bad 2-4 kHz response. But I kind of know that when I listen to them. I know that if I make something that sounds harsh on the 824s, it's gonna sound doubly harsh elsewhere."

Meanwhile, in the hardware synths and controllers department, Gibson is a fan of both NI's Maschine and Teenage Engineering's OP-1. "I love the OP-1, 'cause I'm just always trying to get into a state of play. So, anything that you can fuck around with and just not really totally understand... which is what the OP-1 is beautiful at."

Gibson often carries these experimental methods over into software. "[*Granular effects plug-in*] Portal by Output is good at that," he says. "You can get to really interesting places without really understanding it. I use the Softube Modular rack plug-in quite a lot. Because I don't know what I'm doing and I love it [*laughs*].

"I have modular expert friends I'll get around to the studio and we'll do a day of that. But sometimes I'll just try to hack it through by myself on the software and just put things in and not really know what's going on. Just playing and seeing what sticks out. The Sugar Bytes Effectrix is a fun one for fucking with samples. iZotope Iris can do really creative things.

"In the box, I mainly like to put things to sample-based synths," he adds. "Whether it's the Logic sampler that you can do mad stuff with now, or the granular side of Alchemy. I like styling with something that's really intricate and complex, sound-wise. Like dragging a whole finished song into a granular synth. So, you're starting from a point of real intricacy and trying to find order in it, as opposed to coming from a pure sine wave and trying to add intricacies. I like going the other way around more."

Elsewhere, Gibson maintains a soft spot for his Korg Triton workstation synth, which he's used for many years. "It's just nostalgic for me," he says. "I love recording it in mono and forcing these very stereo sounds into being very mono. It feels like you're sampling, weirdly, when you do that. Whenever you take something really stereo and modify it, it has this immediate nostalgia to it, I find."

Fred Gibson and rapper Headie One recording their 2019 release *GANG* at Fred's home studio in Waterloo.

Lost Stems

Away from the conventional studio environment, Fred Gibson worked with Ed Sheeran on the singer's 2019 album *No.6 Collaborations Project* at a rented house in Nashville. "Ed can do it all himself as it is," Gibson points out. "But we just really, really truly got along. We see eye-to-eye in the studio in terms of we both like to write quickly and instinctively and not overthink. Do more and edit later. It was a very quick and easy fit.

"We did some of it at Promised Land, but at mostly the house in Nashville. I just rented some 824s, and a [*Shure*] SM7[*B*]. Ed was on tour out there in America. When you do those tours, they often set up in the middle and then they'll fly out to each show. So, I flew out for a few different weeks in that stint, and we made the record out there. Then we'd fly to like Atlanta to do the Travis Scott verse or go to LA for a couple of days. But most of the »

writing was all just done in the living room of that gaff with just my laptop."

Another key collaborator on the album (and visitor to the Nashville house) was uber-successful songwriter/producer Max Martin (Britney Spears, Taylor Swift, the Weeknd). Gibson says that the work with Martin involved frenetic levels of creativity that sometimes meant the participants occasionally lost track of exactly what they'd done. A case in point being the song 'I Don't Care', featuring Justin Bieber.

"We wrote a lot of songs very instinctively and just followed our guts. For 'I Don't Care', it was on the fourth day of really intense writing, and everyone was absolutely frazzled. For some reason I'd had a really good night's sleep the night before, so I had loads of energy and everyone else was actually beat.

"Later, I called Max and said, 'Can you send me the stems of the last one we did? It sounded so good.' He was like, 'I don't remember it.' So, I sang it to him down the phone. He was like, 'I have no recollection of what that is.' We'd made so many songs in four days that it was just absolutely a blur. Then luckily, we did get the stems."

Phone Home

Although he sometimes uses Neumanns for recording vocals, Gibson typically reaches for the SM7B. "For pop records and rap records, I'll chuck up an SM7. Some people find it surprising that, like, for Stormzy, Ed Sheeran, it's never anything posher than an SM7. I mean, it's the best mic ever. You don't need to go posher than an SM7, I don't think."

More surprising still is the fact that even above the SM7B, Gibson prefers the sound of the iPhone microphone. "It's the in-built compression of the iPhone I adore," he explains. "Like, whenever I use an SM7, I have a plug-in chain I use to try and recreate the compression of the iPhone."

In fact, the iPhone is central to the music Fred Gibson makes. "I record everything on my iPhone, or everything apart from lead vocals," he says. "I record guitars, I record all the backing vocals, I record drums. I love being able to just capture something and then chuck it into my laptop and warp or manipulate it and turn it into something totally different.

"My mate, Johnny McDaid [*songwriter/producer/Snow Patrol member*], he always has this joke when we're working together. He'll play a part on the guitar and I just get my iPhone out and I'll be like, 'Go go go'. The phrase he always says is, 'It doesn't matter what you put in,' which I find really empowering."

One of Gibson's favourite manipulation tools within his iPhone is elf audio's Koala app. "It's this sick kind of MPC-style sampler that I got recently for the phone. We've been using it in the studio the whole time — recording random bits and manipulating them in the phone, then recording them back into the computer. These things are just so expressive as instruments, I find, and you can really quickly get really cool results. I'm most interested in those things that can just be taken everywhere and really fuck with the sound as opposed to something that gives it a little glaze or whatever."

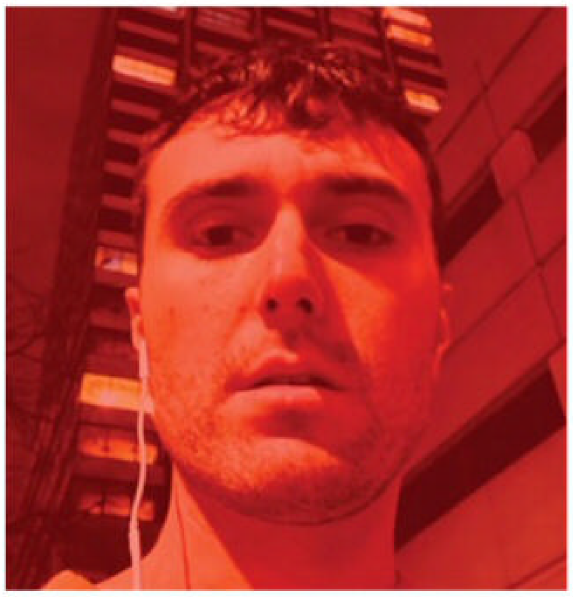
Another of Gibson's — and Eno's — current favourite tools for more extreme sonic treatments is Expert Sleepers' Spectral Conquest plug-in. "Brian and I are obsessed with it," Gibson enthuses. "It's basically a spectral EQ, but to ridiculous extremes. You can delete any single frequency, like absolutely take it out. And you can do really creative things with that if you put in a whole song. Suddenly, you get this whole new world and you can make granular synths out of it."

Found Sounds

Gibson's various sonic techniques are perhaps best showcased by his electronic solo project Fred again.., his first release being the *GANG* mixtape with Tottenham rapper Headie One in 2020, which also featured guest vocals from FKA Twigs and Sampha. "It was really effortless," says Gibson. "It was just instinct again. I'm always obsessed with capturing the moments when people aren't overthinking. I was really inspired by how sonically fearless Headie was."

Meanwhile on the 2021 debut Fred again.. album, *Actual Life (April 14 — December 17 2020)*, Gibson fully explored a snatched field recordings approach in what he describes as a "collaborative diary". The springboard into the project was a chance meeting in a bar in Atlanta with a construction worker named Carlos, whose enthusiastic phrases Gibson recorded into his phone.

"I was just having a beer and he came up to me and was like, 'What you sayin', partner?' We were just having jokes all the time and I was sometimes hitting record on my phone, like I do on nights out, capturing moments. And he was all like, 'We gonna make it through!' I was laughing and just being like, 'This is an amazing man.'



Released in April this year, Fred again..'s debut solo album, *Actual Life (April 14 — December 17 2020)* is based around recordings made on Fred's mobile phone.

"Then I woke up, hungover, in the hotel room, with all these videos on my phone, and I just found myself dragging them into Logic and messing with them. Really quickly I loved that feeling that it was bringing to life and showing all the glory in these seemingly mundane moments."

Gibson then began to run with the idea for the *Actual Life...* album, going further and sampling songs and snippets from social media to build up top lines and sonic motifs. A clip of Minneapolis poet/rapper Kyle Tran Myhre aka Guante performing at an open mic night featured in-house track 'Kyle (I Found You)'. An Instagram video of Australian singer-songwriter Angie McMahon resulted in the melancholic dance track 'Angie (I've Been Lost)'.

"She's just sat on a bed and she's playing this song," Gibson remembers. "It gets to the bit where she goes, 'I've been lost, I've been lost', and it was just absolutely hypnotic to me. I've been sort of obsessed with her and that moment ever since.

"The key thing to me was just trying to find moments of what felt like real-world honesty. I would stumble across them very naturally. It would just be in an aimless scroll, as we're all very prone to doing. Or like, on a random night out, as I'm often the person filming bits and bobs on my phone."

Although, as he said earlier, he rarely records lead vocals using his iPhone mic, for Gibson's own vocal parts on *Actual Life...* he matched the sonic qualities of the samples by reaching for his phone.

"I just do it in a silent room a lot of the time," he explains. "If I played the song five seconds ago, I'll still be singing in the right key. I'll pause the song and sing some

Fred Gibson's live show combines technology with live vocals and piano.

ideas into my phone and just Airdrop them over and drag them in and see how it feels. It usually lines up."

Inevitably, of course, sometimes pitch correction or editing is involved. "Yeah, but what I like about it is usually that leads you to creative solutions," Gibson says. "Like you often end up doing things you'd never have done. You'll manipulate it into a rhythm that you'd have never sung on, or you'll pitch a note suddenly up a whole fifth that you couldn't physically have sung. It forces your hand to play cards you don't expect to play."

In his vocal processing, Gibson is often trying to exaggerate the natural reverb compression of the iPhone microphone. "If you speak into an iPhone, the reverb is totally overpowered by your close voice," he stresses. "But as soon as you stop speaking, obviously it fills up the gap with the room noise.

"Aside from quite extreme compression, I love putting the reverb before the compressor. You can do it with a massive reverb or with a really small room reverb and it means you still get that clarity and closeness. But each time the vocal gives you room to breathe, the reverb fills that space. So, it really hyper-emphasises the space of the singer, which I find gives it a real emotional context."

Lasting Legacy

Unusually, perhaps, Gibson favours legacy Logic plug-ins such as Silver Compressor and AVerb. "I love Silver Compressor, which you have to hit Alt now to get. You have to load up the secret legacy menu [*laughs*]. And ChromaVerb is I guess what Logic thinks replaced AVerb. And it's good, but it's good at different things to me. AVerb has this kind of hollow longing to it. And it's so simple. That and Silver Compressor, they're both, like, two-dial plug-ins. And that really speaks to me because it's not worrying too much about this niche thing that's gonna make one percent of difference. I want to be making big differences quickly and easily."

All of which underlines the fact that Fred Gibson is anything but a picky audiophile — something he shares with Eno. He remembers a funny moment from the mastering of the second Eno/Hyde album, *High Life* in 2014.

"We had a song that we were trying to get mastered and we couldn't find the session," Gibson recalls. "The way we'd been working, everything was very loose and free and jammy. And we couldn't find the session that had that exact version. We just had this MP3 of it. And the masterer was like, 'We need a WAV, you can't send us the MP3.' So we just dragged the MP3 into Logic, bounced it as a WAV and sent it off to be mastered [*laughs*]."

Ultimately, then, Fred Gibson is someone who feels that traditional audio engineering rules are only there to be bent or broken. "A lot of the time I think those rules that are set up were for people who were trying to record like a Stravinsky quartet in Abbey Road back in the day," he states. "They're not for how music is made now. Like, I get how you want to keep your signal chain clear if you're trying to get the drum sound of [*D'Angelo's 2000 album*] *Voodoo*, or these types of things. But for loads of what I do, all of these audio rules that we were taught do not apply. Because if the first thing you're sampling is a YouTube video or something you recorded from your iPhone, then forget everything else. And if you can use the plug-ins to take the harshness out of this or that low-grade thing, then do.

"But, I mean, Skrillex, who is a friend who I've worked with a bunch and who to me is a genius and one of the greatest living sound designers... he is the ultimate practitioner of this. I mean, this guy works through anything. There's absolute recklessness, as there should be."

Looking to the future, Fred Gibson has many projects on the go, including a new collaborative album with Eno and a second, upcoming Fred again.. album — a taster of which arrived in August with the Baxter Dury-featuring track, 'Baxter (These Are My Friends)'. Additionally, he's doing more of his live Fred again... performances, in which Gibson combines technology with live vocal and upright piano, and appearances from his various sampled collaborators on screen.

"I want to make a lot of music now," he stresses. "I've spent a long time, three years or so, really honing what I wanted to do. And I'm still totally working that out, of course, and I will be for years. But I've reached a point where I'm now like, 'OK, I understand enough that I can now start showing my work and putting it out and going, going, going.' So now it's like, it's 'go' time.

"I feel motivated by the moment where you are just absolutely obsessed with the song you made five minutes ago," he concludes. "It's going around your head, and you're going to sleep thinking, 'Yes.' That is all I'm ever chasing every day."

Antares Auto-Tune Vocodist

Vocoder Plug-in

PAUL WHITE

Combining the classic vocoder with Antares' popular pitch processor, this plug-in is capable of some pretty out-there effects.

Antares are famously the inventors of Auto-Tune, the world's first real-time vocal pitch-correction system, and they now produce a range of different plug-ins that incorporate that technology. Most recently, they announced the release of Auto-Tune Vocodist, a seriously comprehensive vocoder plug-in that incorporates Auto-Tune, to ensure that the vocal input is optimised for pitch, whether for subtle correction or to use Auto-Tune as an overt effect. Vocodist is also compatible with the Auto-Key plug-in for detecting key and scale.

Vocoders are able to produce an instantly recognisable 'talking instrument' effect by using a voice input to modulate a series of filters which process a musical input. The result is the sort of robotic effect you've doubtless heard on recordings by artists including Daft Punk, Kraftwerk and ELO, not to mention the voice of the Cylons in the original *Battlestar Galactica* TV series!

Overview

Vocodist includes filter models derived from 20 different vintage hardware vocoders, as well as offering a useful degree of largely intuitive user adjustment. It has an internal, MIDI-playable synth section that can be used as the musical component or carrier signal, or it can accept external sounds via its side-chain input. The internal synth can also track the incoming voice pitch, if desired. Over 125 artist presets by the likes of P-Thugg and producer Buddy Ross are included, and there are plans to add more over the next few months. Currently, Auto-Tune Vocodist is only available as part of Auto-Tune Unlimited, a subscription service that covers all current versions of Auto-Tune plus all 11 AVOX vocal effects. (Auto-Tune Unlimited costs $24.99 monthly, or an annual subscription works out at $18.74 a month, but a free 14-day trial is available.)

Vocodist goes beyond the classic vocoder sounds — actually, it's capable of some very radical transformations and the only competing product that I've tried which offers similarly 'out there' vocal treatments is Waves Ovox, though the two are pretty different. The vocoder section allows the user to change the number of bands, to set the minimum and maximum frequencies of the bands, to choose from one of four filter shapes and to adjust the Q of the filter response with further controls for Stretch and Slide.

The Envelope section is where the voice is analysed and turned into control signals for the filter bank, and here you can adjust the level of each band, shift the bands up or down and change the attack and release time of the envelope response. A Smear control is also provided, along with a random dice button.

At the bottom of the GUI is the synth, which can have up to eight voices and two oscillators per voice. The waveform for each can be either sawtooth or a variable mark/space-ratio pulse wave, with controls for Glide, Octave, Interval and Detuning. The synth control source

Antares Auto-Tune Vocodist

PROS

- Excellent sound quality.
- Can emulate traditional vocoders as well as producing special vocal effects.
- Auto-Tune is built-in.

CONS

- Subscription-only package won't appeal to everyone.

SUMMARY

A hugely flexible vocoder that doesn't let its enhanced capabilities get in the way of intuitive operation.

can be set to MIDI, to follow the pitch of the voice input, or you can use a side-chain input with the option of adjusting compression and gain. Furthermore, the synth includes an oscillator scale-transpose function for creating three-part harmonies — very unexpected in a vocoder, but a pleasant surprise and a useful facility.

A noise section with a choice of white or pink noise and controls for Sensitivity, Balance, Level and Color allows noise to be added to the carrier sound, which can sometimes improve the intelligibility of unpitched fricatives. The voice input, meanwhile, can be compressed, its emphasis can be adjusted and the familiar Auto-Tune parameters adjusted to change the Retune Speed, Humanize amount, voice type and to select the musical key.

There are also adjustments and effects that can be applied to the vocoder section's output, affecting the envelope, tonality and mix of the signal as well as adding in some chorus. Tools include saturation and ring modulation. An automatable X-Y pad can also be assigned to various functions such as Filter Slide or Vocal Envelope, to add some in-flight variation to the sound. If you select MIDI as the pitch-track source, you can assign any pair of MIDI CC numbers to the X and Y axes. While any of the parameters can be automated within your DAW, Vocodist's X-Y mapping capability provides a quick and easy alternative.

In Use

When you first launch Vocodist, you're offered a walk-through of its three modes of operation: synth controlled over MIDI, synth controlled from a voice, or external synth source; once you've grown familiar with Vocodist you can switch this feature off, but it does help when starting out, as it takes you through each step of setting up. The user manual is also very detailed and clearly written. Note that Logic users will have to play the now-familiar MIDI Controlled Effect game via an Instrument track and side-chain input if using the MIDI input facility, but otherwise Vocodist can be instantiated on an audio track.

It doesn't take long to get to grips with the controls, but you can get an idea of the breadth of sounds available just by exploring the presets. The conventional vocoder sounds range from blurry vintage models with relatively few bands to high-definition contemporary effects using the maximum number of bands, all of which capture the essence of a traditional analogue vocoder but each with a unique character. The change in sound that can be achieved simply by changing the Q of the filter bands is surprising too. Once you get into special-effect territory, all bets are off and you can go as far down the rabbit hole of the alien, robotic, mangled and strangled as you like. Vocodist is a fabulously creative tool, but don't take my word for it — I highly recommend downloading the demo and checking it out, along with Antares' other apps.

£ Available only in Antares subscription service, from $24.99 pcm or $224.90 per year. (Free 14-day trial.)
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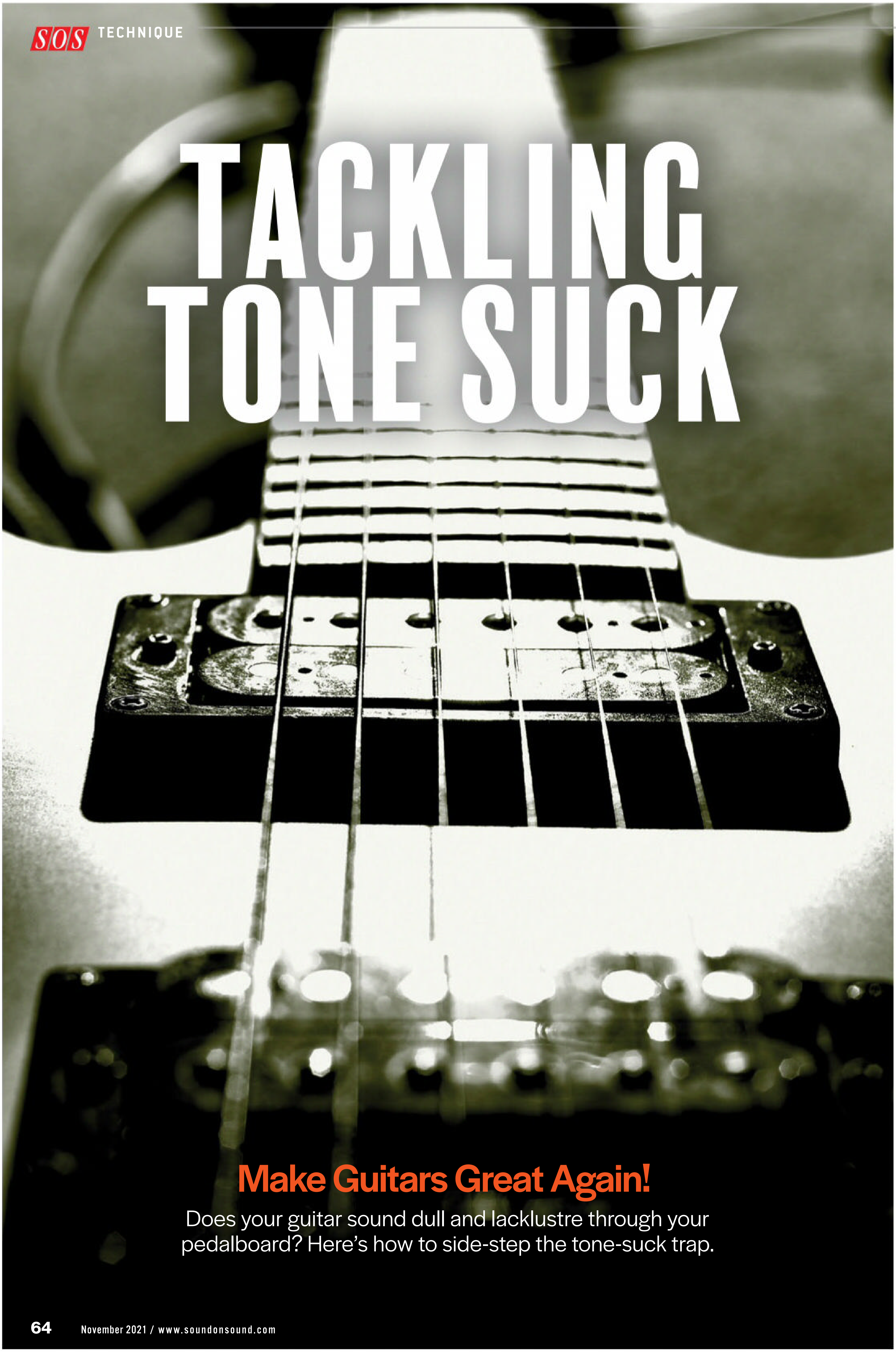


TACKLING TONE SUCK

Make Guitars Great Again!

Does your guitar sound dull and lacklustre through your pedalboard? Here's how to side-step the tone-suck trap.

PAUL WHITE

Most electric guitars are still based around primitive magnetic pickups and use passive, WW2-era radio circuitry to control the volume and tone, so it's little wonder that the sound they produce varies according to what you plug them into! If you want to get control over your sound, then, you need to understand a little about how your guitar and amp — and anything you put between them — interact. It's largely a question of the impedances involved, but it's important to note that impedance is more complex than resistance; the combination of resistance, inductance and capacitance makes the impedance change with frequency.

A guitar cable's capacitance is determined both by its construction and its length. The longer the cable, the greater its capacitance — and the more likely it is to dull your tone.

Impedant Swine!

The higher the input impedance of the device you plug you guitar into, the lower the current required to drive it. If an input impedance is resistive and too low, it 'loads' the circuit, reducing the level. As it is in parallel with the output circuit of the guitar, it may also affect the tone. The output impedance of an electric guitar is usually a few kΩ — the specific value is determined mainly by the impedance of the pickup coils — so most guitar amps present a much higher input impedance than that, typically from 500kΩ to over 1MΩ. This ensures that the amp doesn't unduly load the signal produced by the guitar, and it's why plugging you guitar directly into a nice amp generally makes it pretty easy to set up a pleasing sound, with plenty of life and responsiveness.

But your guitar and amp are never the only things in the signal chain: at the very least, there'll be a cable and you might use pedals too. Both change this relationship between guitar and amp, and it's not at all uncommon for people to notice that their guitar tone becomes considerably duller and less 'vital' when using long cables or lots of pedals. This unwanted effect is often described as 'tone suck'.

Cable Capacitance

While it's probably a healthy thing to approach cable companies' marketing claims with a good dollop of skepticism, all guitar cables do have a significant effect on the sound, even when plugging directly into an amp. That's because the close proximity of the 'hot' inner conductor and the outer screen makes the cable function as a capacitor. If the cable's capacitance is too high it will dull the sound, since high frequencies pass through capacitors to ground more easily than low ones.

Different cable types can have different capacitances; as with a conventional capacitor, the cable's capacitance is determined by the 'dielectric' and the space between the centre conductor and the screen. But just as important is that, all other things being equal, the longer the cable is, the higher its capacitance will be, and the more noticeable the high-end roll-off.

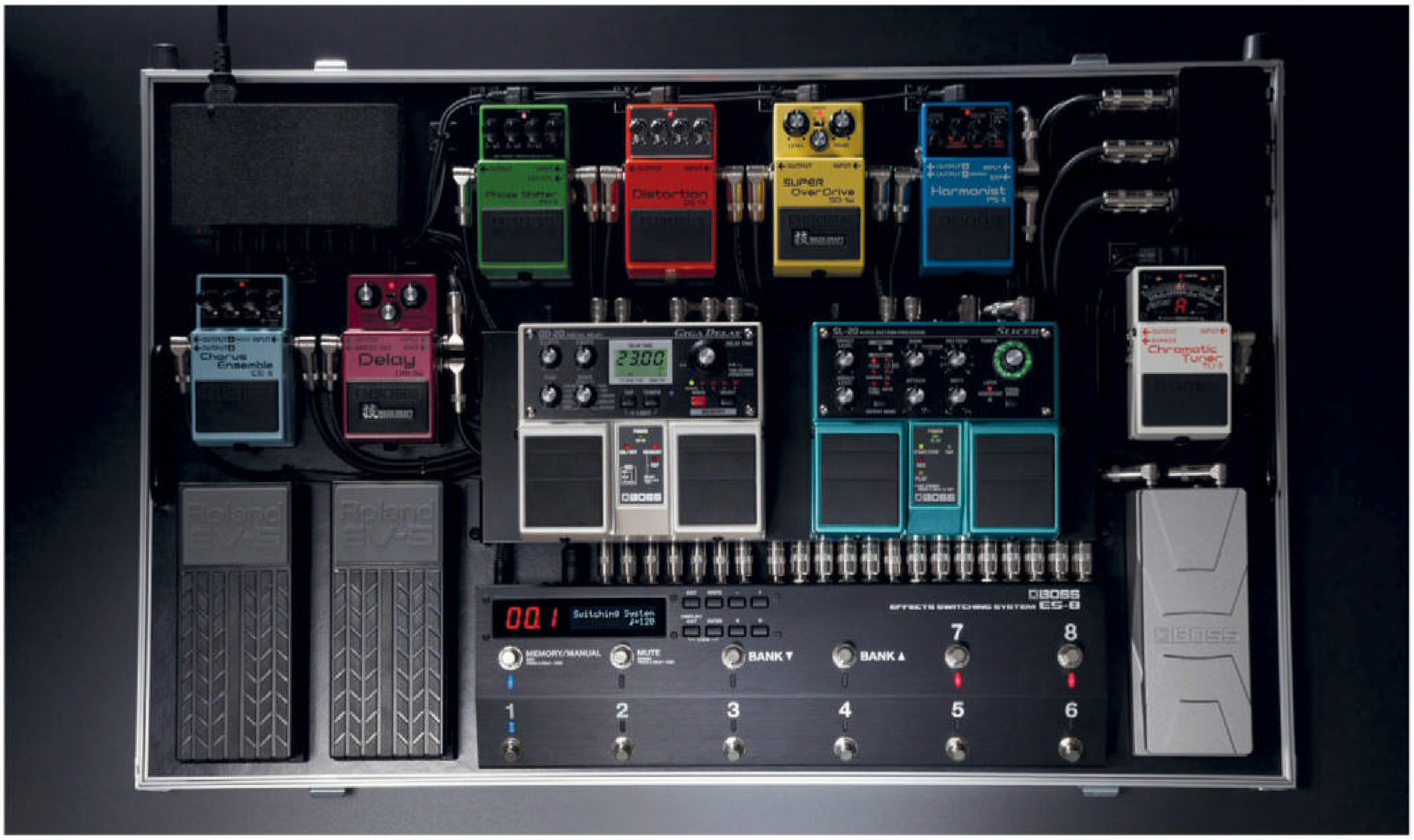
There's more going on here than low-pass filtering, though. The inductance of the pickup coils in parallel with the capacitance of the cable and the self-capacitance of the pickup forms what is, in effect, a tuned circuit with a resonant peak. It's not a really strong resonant peak such as you'd hear from a wah-wah pedal, but it is strong enough that it imparts a distinct tonality. Most high-end pickups are designed so that the coil inductance and self-capacitance of the pickup, when loaded by the volume and tone controls, produces a pleasing sound. But plugging in your 'cable capacitor' moves the resonant frequency down.

So is there an optimum cable capacitance? Not really! It's all about what sort of sound you prefer. Sometimes, a modest amount of cable capacitance can be pleasing, especially if a particular guitar-amp combination is too bright and 'glassy' for your tastes. If the cable capacitance is too high, though, that carefully calculated resonant peak will move low enough that you perceive a distinct loss of treble: the dreaded 'tone suck'. With all this in mind, it's worth trying out low, medium and higher-capacitance cables with your own guitars and amps and listening for the differences. If you plug into an amp directly, simply choose what you feel is the best-sounding cable for your guitar and that amp, and you're done. If you have a range of guitars and amps to play with, it might be worth investigating products such as Undertone Audio's VariCap, in essence a low-capacitance cable that incorporates a switch to increase the capacitance to taste.

Taking The Bypass

Once you get into the murky worlds of stompboxes and radio transmitters (for more on the latter, see the box), the rules change. Before we take a closer look at the effect of the pedals and patch cables, though, I'll stress that you should ensure you're using a good-quality pedalboard power supply, with isolated outputs capable of delivering at least as much current as each pedal requires. Some of the cheaper PSUs that claim to have isolated outputs actually don't, and if you hear hums, buzzes or digital whining noises, even with your guitar volume turned right down, the PSU is probably to blame.

More pertinent to 'tone suck' is what happens when you bypass your pedals. Today, almost all pedals (there are exceptions, as I'll explain later) use one of two main approaches: buffered (active) or true (hard-wired) bypass, and a handful now allow you to choose which type you »



» prefer. You might suspect that a hard-wired bypass would give you the truest tone: what harm could a short stretch of wire do to your signal? Well, if you have only one or two pedals in the signal chain, you might be right. But once you put together a moderately busy pedalboard you'll have lots of short jack cables in series, lots of plug/socket spring contacts, and the cumulative effect of any stray capacitance that occurs from the wiring inside the pedals. On top of that, you'll be adding a second, longer cable to connect the pedalboard to the amp. If all your pedals are hard-bypass types, the capacitance added by all that extra cabling and internal wiring, plus the contact resistance of all those plugs and sockets, can easily compromise your tone.

Something you can do to mitigate this is always to buy (or make) good-quality, low-capacitance patch cables and keep them as short as possible; the cheaper moulded types often sound significantly worse than decent cables.

Buffer: The Amp Ire Slayer?

OK, so if hard-wired bypass pedals can cause problems, presumably buffered pedals are better? Erm, not necessarily! A buffer is intended to do two things: (a) present your guitar with a high enough input impedance to keep it sweet; and (b) provide a very low output impedance, so that long cables or other pedals don't cause too much treble loss. A good buffer circuit should be tonally transparent, though some dedicated buffer pedals are deliberately designed to inject a little preamp-like mojo too. A further benefit of buffered bypass pedals is that they can be designed to provide click-free switching; many hard bypass pedals, especially vintage examples, suffer from switching clicks.

If all your pedals have a hard-wired bypass, placing a dedicated buffer pedal at the start of the chain is definitely a good idea. It's OK to mix buffered and hard-wired types too, but it will help preserve your tone if you place either a good-quality buffered bypass pedal or a dedicated buffer at the start of the chain. Any buffer with a very wide audio bandwidth, low noise and low distortion should sound suitably transparent. If you already put a tuner pedal at the start of your signal chain, you may find this has a perfectly serviceable buffered bypass (check its manual to find out), in which case you probably won't need a separate buffer.

The main problem when using multiple buffered-bypass pedals is that they're not all created equal; not all are as transparent-sounding in bypass mode as we'd like them to be. None should sound completely terrible but, most likely, you'll hear at least some kind of tonal change and maybe some added noise too, and cascading multiple buffers of the type you get in most pedals typically leads to both low and and high-frequency roll-off, and a general muddying of the sound.

To complicate things further, some pedals work better when connected

If you use lots of pedals, a dedicated switcher can help to ensure the effect of bypassed pedals on your tone is minimised. And always be sure to use short, good-quality patch cables, to keep capacitance to a minimum.

Vintage fuzz pedals such as the Dallas Arbiter Fuzz Face generally sound much better placed immediately after the guitar, before any buffer stages.

directly to the guitar, without any buffer in front of them. Vintage fuzz boxes based on germanium transistors (such as the DA Fuzz Face) are notorious for this, so make sure you place these right at the front of the chain, before any buffer pedal, and before any tuner with a buffered output. If you want to put that sort of effect later in the signal chain, after a buffer, you might be better off using a more modern fuzz/distortion.

Another quirky device is the passive volume pedal, whose pot resistance is placed in parallel with the impedance of the rest of chain, effectively lowering it. If that pot value is below 1MΩ the loss of top end can be very noticeable, so a passive volume pedal should ideally be placed after either a buffered bypass pedal or a dedicated buffer pedal.

Finally, it's worth noting that while the two main approaches to bypass discussed above are by far the most common, some old-school pedals adopt a third approach: they still use mechanical switching to connect the input directly to the output but leave part of the circuit connected, and »

Not all buffer pedals are created equal but before you rush out and buy one to place at the start of your signal chain, you could check to see if your tuner pedal is already up to the job.

Photo: Starman1984, CC BY-SA 3.0 via Wikimedia Commons

this has the effect of loading the bypassed signal. Such pedals usually benefit from being placed after a buffered device.

The Shortest Path

Given all this, you might expect that you'd obtain the cleanest results by first chaining only hard-wired bypass pedals, and then placing high-quality dedicated buffer pedals at both the start and end of the chain. You don't choose pedals for their bypass characteristics alone, though, and most practical pedalboards, mine included, will comprise a mixture of both types of pedal. So, other than putting a good-quality buffer at the start of the chain — which won't solve everything — what can you do?

When recording in the studio, you can probably get the best tone by patching together only the pedals you need to get a particular sound directly. If you already have a separate buffer pedal, try that at the start of the chain too (unless using one of those antique germanium fuzzes or a clone); it might help support your tone.

The miinimalist approach is less likely to be viable on stage, where you'll often need to bypass different pedals for different tracks. A more serious option, for stage and studio, is a dedicated pedal switcher. Many of these use reed relays to give you the cleanest possible path through a chosen set of pedals, regardless of whether they have a buffered or a hard-wired bypass. Most switchers allow you to save patches, too, which not only remember your active pedals but also allow you to change the order in which they're connected. Many have buffers built in, both at the input and the output, which you can place in or out of circuit as required. Having a good buffer on the final pedal's output will often help if you have a long (so high-capacitance) cable running from your pedalboard to your amp. The only real downside of switchers, other than their cost, is that they take up space on your pedalboard.

A less costly alternative is to add one more pedal to your board: a passive bypass switch that connects the input of your pedal chain (or just the offending section of it) directly to the output, bypassing everything in between. If you're handy with a soldering iron you can make your own using a double-pole, double-throw (DPDT) foot-switch, or a three-pole switch if you want to add an LED. This will ensure that when you just want the sound of your guitar plugged directly into your amp you'll get it.

Purism aside, it may still be worth putting an input buffer before the bypass switch, as it will prevent your pedalboard-to-amp cable dragging down your tone when the pedals are bypassed. In theory, you shouldn't need an output buffer at the end of the pedal chain: whenever one or more pedals are active the pedal's own circuity should be capable of driving a length of guitar cable, and when all are bypassed the input buffer will drive the cable.

Some pedals have oddly designed output stages that can be affected by the load they run into, so let your ears decide whether you also need an output buffer. To do that, keep comparing the sound of your bypassed pedals with that of your guitar plugged straight into the amp. The aim is to get the two sounding as close to each other as possible. If a crucial pedal has a particularly nasty bypassed sound, hook it up to the type of hard-wired bypass footswitch I described just now; leave the pedal active all the time and use the footswitch to bring it in and out of the chain. Sometimes a hard bypass switch will introduce small switching clicks, but that's preferable to a permanently mangled tone!

Radio Transmitters

Since a cable's length influences its capacitance and thus the guitar tone, it's worth noting that guitarists who use radio systems will generally plug into the transmitter using a very short cable. Such a cable will inevitably have much less capacitance than a typical full-length guitar cable so it can serve to brighten the sound, and sometimes excessively. Some radio systems include what they call cable capacitance emulation, but even where it is implemented I've only seen it in the receiver and, if you think about it, while you can shave off a little high end that way, you can't change the resonant peak of the pickups. To do that, you need to add capacitance at the transmitter end.

Using your usual, longer cable would probably defeat the point of using a wireless system but if you prefer that sound you can replicate the effect by putting a small capacitor across the jack plug contacts of your short radio-link cable; if you choose a roomy jack plug you should be able to get the capacitor to fit inside. A typical traditional guitar cable (there are specialist low-capacitance types available too) has a capacitance of around 100 picofarads per metre — so if you usually use a standard 5m cable you'd need to add around 500pf to make it sound similar. If the resulting sound is too dull, reduce the capacitor value and try again. My own 'radio cable' is fitted with a Neutrik TimbrePlug, a TS jack plug that has a set of switchable capacitors built in — it's a very neat solution and it's not too expensive!

Tonal Takeaways?

Tackling tone suck can seem challenging because multiple factors can contribute to it. But you can break the task down to more manageable stages. A good start is to ensure you use good-quality cables and a decent pedalboard PSU. Next, audition each pedal individually, to judge whether its bypassed sound matches the sound of your guitar plugged directly into the amp. Then patch all your buffered-bypass pedals together in series and see if you can hear the cumulative effect when all are bypassed. If you can, a physical switcher may be the only truly effective solution. But you can also see if connecting your pedalboard to your amp using a very short cable sounds better than with your usual cable. If it does, that indicates that an output buffer as the last stage in your chain might work well, along with a better-quality, lower-capacitance cable. Nobody said it would be easy, but an afternoon's detective work could improve you guitar sound forever.

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Genelec 8351B & W371A

PHIL WARD

Active Monitors

This impressive setup combines point-source technology, advanced loudspeaker management DSP and a powerful new bass extension system. Is this the state of the art in studio monitoring?

The 8351A was the first of the technologically advanced Genelec The Ones series of three-way, dual-coincident, 'point-source' monitors to be introduced, and Bob Thomas gave it a glowing review in this magazine back in August 2015. Bob has also more recently (in *SOS* September 2021) written, again in glowing terms, about the larger 8361A. Since the launch of the 8351A Genelec have expanded the series to include both smaller and larger models, and some of the know-how and newer technology developed for those products has been fed into the first subject of this review; the 8351B.

There's more than just a revised and updated monitor to describe here, so the plan is to cover quite a big parcel of Genelec land along with the 8351B. That's because Genelec have significantly refreshed and upgrared their GLM (Genelec Loudspeaker Manager) optimisation and SAM (Smart Active Monitor) control technology, and also introduced the W371A Woofer System. There are, by the way, some very good and technically fascinating reasons why I didn't, and Genelec don't, describe the W371A as a subwoofer. I'll get around to explaining that later on. First I'll offer a few paragraphs on the electro-acoustic fundamentals of the 8351B.

To The Point

As with a few other dual-coincident monitor designs of recent years, Genelec's opportunity to design and introduce dual-coincident technology came about with the lapse of KEF Electronics' original patents, which effectively protected the idea of a small tweeter mounted concentrically at the apex of a substantially conical bass/midrange diaphragm. With The Ones series, however, Genelec chose not just to borrow the KEF idea and apply it to an otherwise conventional two-way monitor design, but to develop it much further and extend the idea to a three-way 'point source' (meaning that low, mid and tweeter elements all effectively radiate from the same location in space) monitor format, and to integrate the mid/tweeter compound driver with the curved form of the enclosure front panel. At a quick glance it's not really obvious where the midrange driver diaphragm ends and the enclosure front panel begins. And as I'm mentioning the front panel, it's probably a good time to say that the whole 8351B enclosure is created from two immensely rigid aluminium die-cast components.

Also cleverly incorporated within the enclosure design is Genelec's unique three-way 'point-source' solution to integrating low-frequency reproduction with dual-coincident mid and high frequencies. A fundamental issue with low frequencies and dual-coincident drivers is that the output of a tweeter mounted at the apex of a bass/midrange diaphragm will suffer significant frequency modulation distortion if said diaphragm is required to move beyond a millimeter or two. This is because the bass/midrange diaphragm effectively has a secondary role as a tweeter waveguide, and the last thing you need in that scenario is the waveguide moving significantly when the bass player hits a low B, just as the drummer's playing some delicate brushed ride cymbal. The obvious solution is to take the responsibility for bass reproduction away from the midrange driver so that its diaphragm is not required to move significantly — but then, where do you locate the bass driver and still retain the point-source principles?

Genelec's solution is to go to a three-way format and split responsibility for low frequencies between a pair of bass drivers located symmetrically about the compound mid/tweeter axis and loaded by a single rear-panel reflex port. The bass drivers are oval in form and all but hidden behind the profiled front panel. They radiate through slots formed between the front panel and the top and bottom surfaces of the enclosure. With the bass-to-midrange crossover frequency at 320Hz, the physical spacing from the bass driver slots to the mid/tweeter driver is small enough a proportion of the one-metre wavelength for the point-source principle to remain valid.

The Drivers

While I'm on the subject of the bass drivers, not only is their architecture and location unusual in the 8351B, they also display an interesting constructional feature in that their rubber roll surround is corrugated around the curved section of the oval. I asked Genelec's R&D Director Aki Mäkivirta about the surround and he explained that its corrugations and curved sections work together to optimise the force distribution around the perimeter of the glass-fibre-reinforced paper driver diaphragm. Diaphragm stability is important in enabling the driver to retain linearity at high excursions, and to achieve that, the surround force needs to be equal around the perimeter of the driver. Genelec optimise the surround in this respect by using variation in its height as a factor to adjust its flexibility, hence the corrugations. Finally on the bass drivers, they are loaded by a shared reflex port located around the back of the enclosure that's very

Genelec 8351B & W371A

PROS

- Extraordinary levels of detail, imaging and tonal accuracy over the full audio bandwidth.
- GLM 4.1 immensely capable yet easy to use.
- System able to self-optimise for a wide range of installation environments.
- 8351B with GLM can work well in very small spaces.

CONS

- Other than the price, none.

SUMMARY

The 8351B and W371A, combined with GLM 4.1, joins a very select group of products that can recalibrate expectations of what's possible from a monitoring system.

generously flared at its exit. The flare will help keep the airflow laminar and so delay the onset of obvious chuffing noises or more insidious port compression effects.

I wrote earlier that it's not really obvious where the midrange driver diaphragm ends and the enclosure front panel starts, and that's because, relieved of the need to play any bass, the midrange driver of the 8351B (and the other The Ones-series monitors) can be fitted with an almost flat surround. It doesn't need the generous roll that would otherwise enable the diaphragm to move more significantly. Designing a surround that needs only to deal with midrange energy also means that its material properties can be optimised to work over a much narrower band of frequencies. Dual-coincident or not, this is one of the big wins that can come with the decision to go with a three-way monitor format: the midrange driver can be designed specifically for midrange and doesn't suffer compromise by also having to play bass. With a dual-coincident driver, however, a flat midrange surround brings a second benefit: it doesn't result in a significant high-frequency diffraction feature.

Before I leave my basic description of the 8351B, I'll just mention that the mid driver diaphragm not only features an unusual two-piece aluminium construction, it also has an inner surround along with the outer surround that I've already described. The inner surround joins the apex of the midrange diaphragm to the outer part of the central tweeter housing, and in doing so cleans up a few potential sources of diffraction. And again, it's the three-way format and the fact that the midrange diaphragm isn't required to move significantly that makes the inner surround feasible. Finally, the 8351B tweeter, nestled down at the apex of the midrange diaphragm, is a 25mm aluminium dome. It's quite possibly the most conventional-looking element of the 8351B.

From A To B

A simple change of codicil letter from A to B, and the fact that the fundamental architecture and external enclosure design of the 8351B remains the same, perhaps suggests that not too much has changed since the 8351A. The reality is that there are numerous revisions. One of the most significant is that Genelec have moved from a conventional, linear internal power supply to a switched-mode device. This not only results in a monitor that's a significant 4.7kg lighter, but enables a substantial increase in amplifier power. Where the 8351A was rated at 150W, 120W and 90W for bass, midrange and tweeter respectively, the 8351B is rated at 250W, 150W and 150W. This increase in amplifier power helps the 8351B achieve a 2dB increase in maximum output level. However, it's not just more power that results in increased maximum volume; all the 8351B drivers are slightly larger than their 8351A predecessors. The slightly larger midrange driver has also made possible a drop in the bass-to-midrange crossover frequency: from 470Hz to 320Hz. This ought to help a little with vertical off-axis consistency at large angles.

A further refinement on the 8351B over the 8351A is that its DSP incorporates phase equalisation from around 500Hz upwards. Phase equalisation corrects for the frequency-dependent delay effects that are fundamental to the processes of electro-acoustics, and although it results in slightly increased overall in/out latency, Genelec say it can provide a subtle but worthwhile gain in subjective performance.

One perhaps even more significant DSP revision in the 8351B is that it incorporates significantly more filter 'horsepower' than the 8351A in terms of the number of individual notch and shelf filters than can be accessed and deployed by Genelec's GLM room optimisation package. Where the 8351A offered a total of 10 filters (six notch and four shelf), the 8351B increases the total to 20 (16 notch and four shelf). The notch and shelf filters are accessible for manual selection through banks of DIP switches on the 8351B's rear panel, adjacent to its balanced XLR

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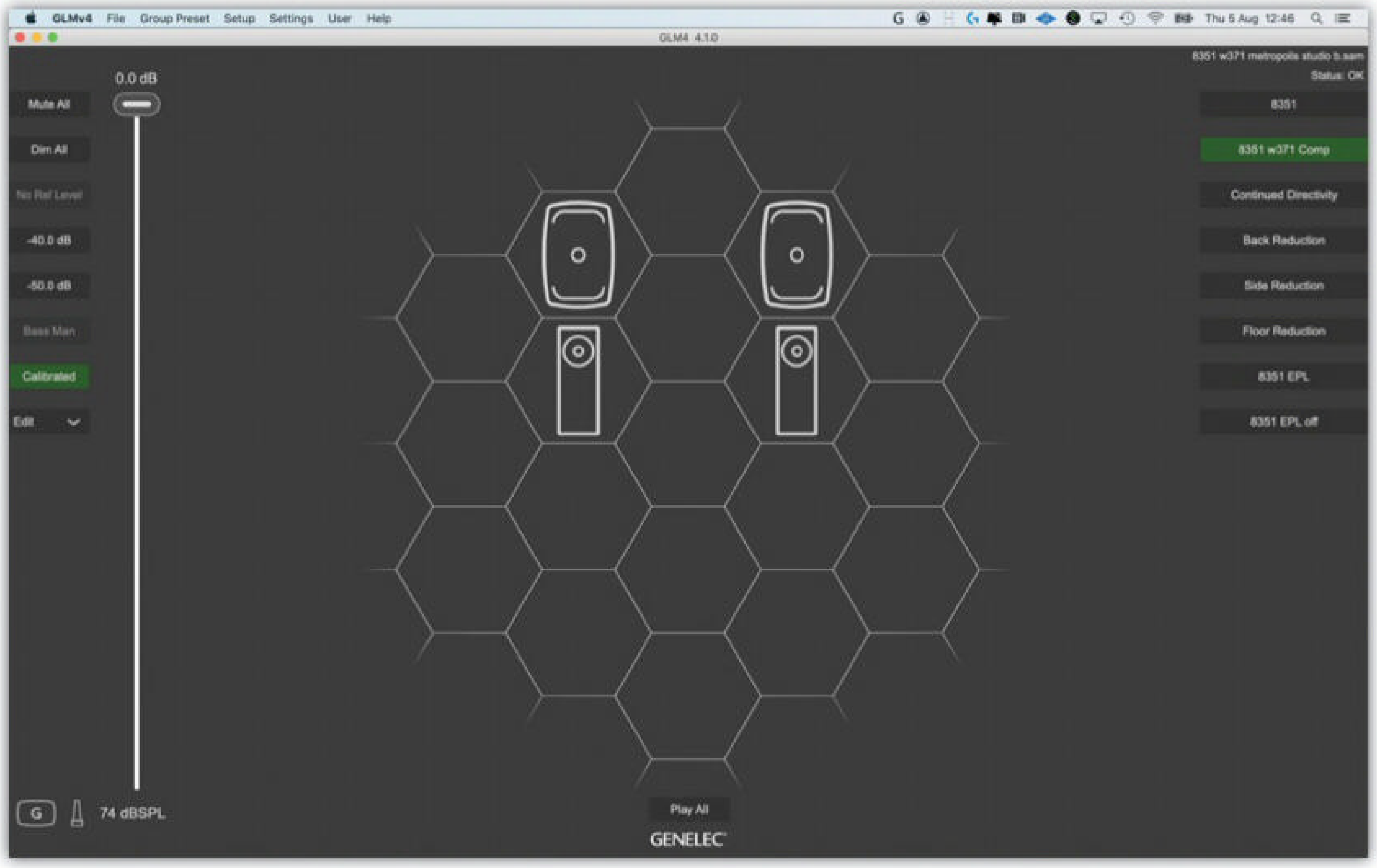


Screen 1: The GLM 4.1 launch page.

analogue and AES3 digital input sockets, but to my mind, manually selecting EQ options for the 8351B borders on the daft when the power of Genelec's GLM app is so easily and, in the context of the cost of the 8351B, relatively inexpensively, available.

GLM

We've written about GLM a few times since its launch in 2006 so I'll not spend too many words covering old ground, but to recap a little, there are two fundamental elements to it: the hardware, comprising a measuring microphone, a compact USB interface and a bunch of RJ45 cables to connect the monitors to the interface; and the software, comprising a Mac OS or Windows app that handles both monitor and room optimisation and monitor system configuration and management. Once a monitoring system is up and running, GLM can also operate as a very handy on-screen monitor controller too.

The recently launched 64-bit architecture, 4.1 version of GLM marks the 15th anniversary of Genelec's room optimisation technology, and incorporates a re-engineered AutoCal 2 routine that, Genelec say, results in quicker and more accurate optimisation data. In particular, AutoCal 2 is claimed to produce better optimisation results in rooms that are more reverberant in character, and also in setups that employ extreme nearfield monitoring (as close as, say, 50cm), either necessarily due to an ultra-compact listening space, or through a desire to emulate a more headphone-like feel to monitoring. GLM 4.1 also adds further to the 500Hz and above phase compensation inherent to the 8351B that I mentioned a few paragraphs ago. The new Extended Phase Linearity option reduces low-frequency group delay (at the cost of an additional 7.5ms overall system latency) such that, for example, an 8351B displays group delay that's comparable to that typical of closed-box monitors: less than 5ms down to 70Hz.

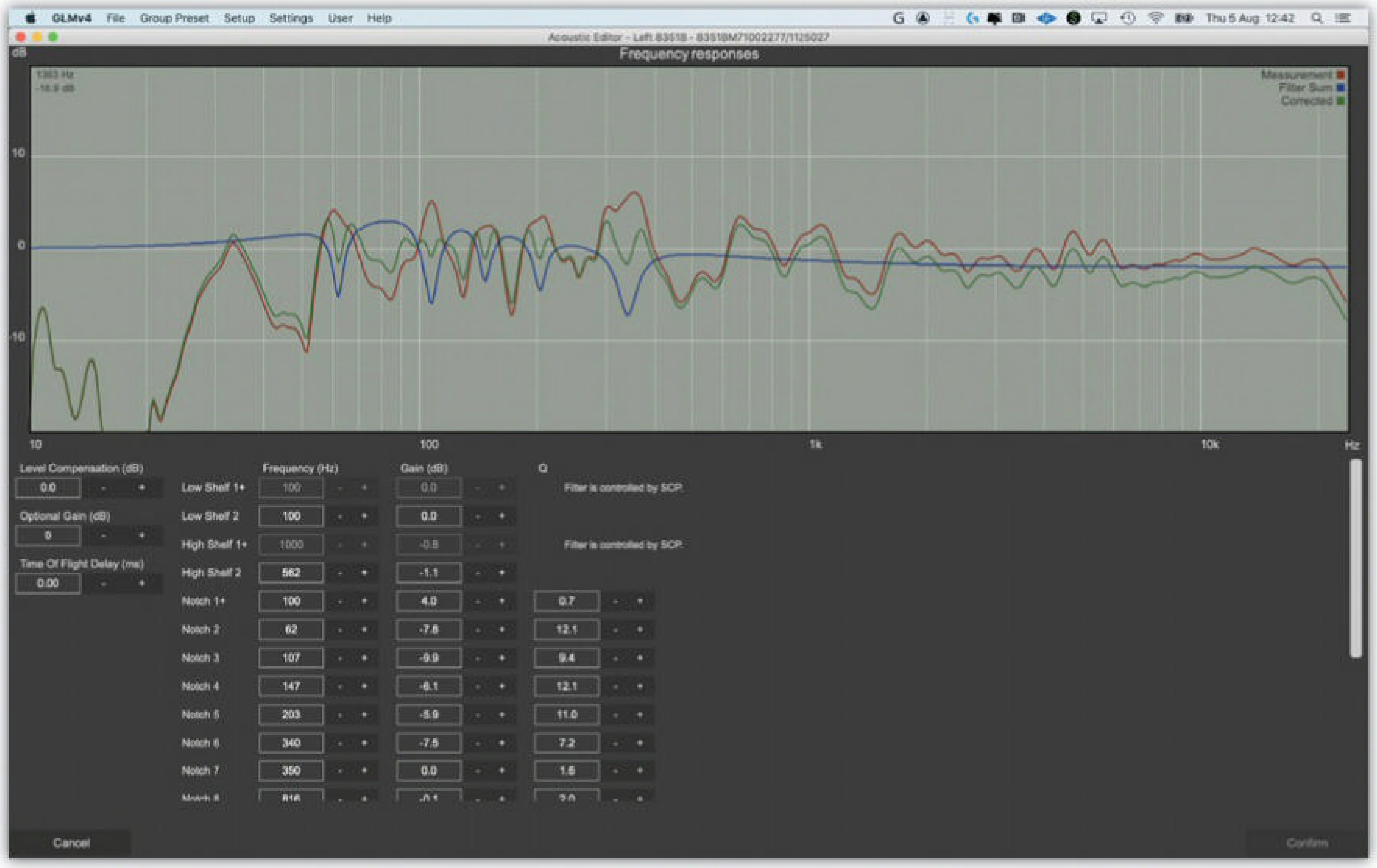
Another new development in GLM 4.1 is that it allows for localised regions of positive gain EQ to be applied where a significant notch in the in-room response can be feasibly equalised without the monitor running into headroom problems. Equalising notches by adding localised gain is something that previous versions of GLM were not designed to do, for two reasons. Firstly, any notch in a room response that's primarily caused by destructive interference between two signals (direct and back wall reflected sound for example) fundamentally cannot be filled in by adding localised gain. The destructive interference between two signals will always occur, never mind how much extra local gain is applied. The second reason for avoiding extra localised gain is simply that it can potentially result in a monitor running out of amplifier headroom. Every 3dB of gain demands twice the amplifier power, so even relatively modest positive-gain EQ can quickly ask significant questions of an amp. The decision to implement positive-gain EQ in GLM 4.1 was partly, say Genelec, a result of the data collected since GLM 3 began to employ the Genelec cloud server for AutoCal processing. The resulting dataset of 20,000 examples of monitor optimisation curves gave Genelec the confidence to know in which frequency bands, and to what degree, positive EQ could be both safely and effectively implemented.

In use, GLM 4.1 is fundamentally straightforward and intuitive, but its ability to manage and optimise systems comprising very large numbers of monitors and channels, and to create optimisation profiles for multiple listening positions and multiple monitoring setups, means things can quickly become complex. To provide a flavour of just how complex, GLM 4.1 can manage systems comprising over 80 monitors and subwoofers, creating an individual response optimisation profile for each one while also enabling monitors to be grouped in different format arrangements (stereo, 5.1 or Dolby Atmos, for example) and also offering multiple optimised listening positions for each group. GLM 4.1 imposes very few limitations in terms of the practical application of even the largest monitoring systems. The decision I think to endow GLM 4.1 with the capability to optimise and manage so many monitor and subwoofer channels reflects Genelec's increasing success in equipping very high-channel-count immersive performance and production spaces.

A significant element of GLM 4.1's ability to manage such complex systems is the combined architecture of the software and

ALTERNATIVES

If I were in the fortunate position of considering a full 8351B, W371A and GLM 4.1 system, I would probably also want to hear the **Kii Three BXT System**, the **Ex Machina Quasar**, the **Barefoot Sound MiniMain 12**, and perhaps something more conventional like an **ATC SCM100A** or **PMC IB2S**.



Screen 2: The correction EQ graph for a single 8351B.

monitors, which puts the DSP hardware needed to handle the room compensation EQ in the monitor. The GLM software processes no audio itself, so in terms of doing the maths it cares little how many monitors are involved — it 'simply' manages the optimisation process and uploads the response optimisation data to the appropriate monitor. Similarly, the remote cloud server location of the AutoCal 2 optimisation calculations helps GLM manage extremely complex systems without imposing any significant processing load on the local computer (which quite possibly already has its hands full running a massive, multi-output DAW session).

Calibration

Regardless of how many monitor or subwoofer channels a GLM system comprises, the process for creating room optimisation data for upload to the appropriate monitors is the same. The GLM 4.1 launch screen displays a hexagonal grid graphic that represents the monitoring space, as illustrated in Screen 1. Monitors connected (via the proprietary Genelec Ethernet-based network) to the Genelec GLM USB interface appear in a list at the side of the hexagonal graphic and can be dragged into one of the boxes to represent its monitoring role (left, right, centre, surround, and so on).

When the monitors are positioned, optimising each one is simply a case of placing the GLM measuring mic at the preferred listening position and pressing the AutoCal button. GLM will route a sine sweep test signal to each monitor or subwoofer in turn and upload the results from the measuring mic to the GLM cloud server, which intelligently decides on the appropriate EQ settings. GLM then uploads the EQ settings to each monitor taking into account the number of DSP-based filters available in each. Screens 2 and 3 respectively illustrate the GLM correction EQ for the left-channel 8351B alone and then the composite 8351B/W371A system. Being a room-mode effect, the low-frequency suck-out between 40Hz and 50Hz on the 8351B alone wouldn't be fixable by adding targeted gain, but is corrected perfectly when the W371A is added.

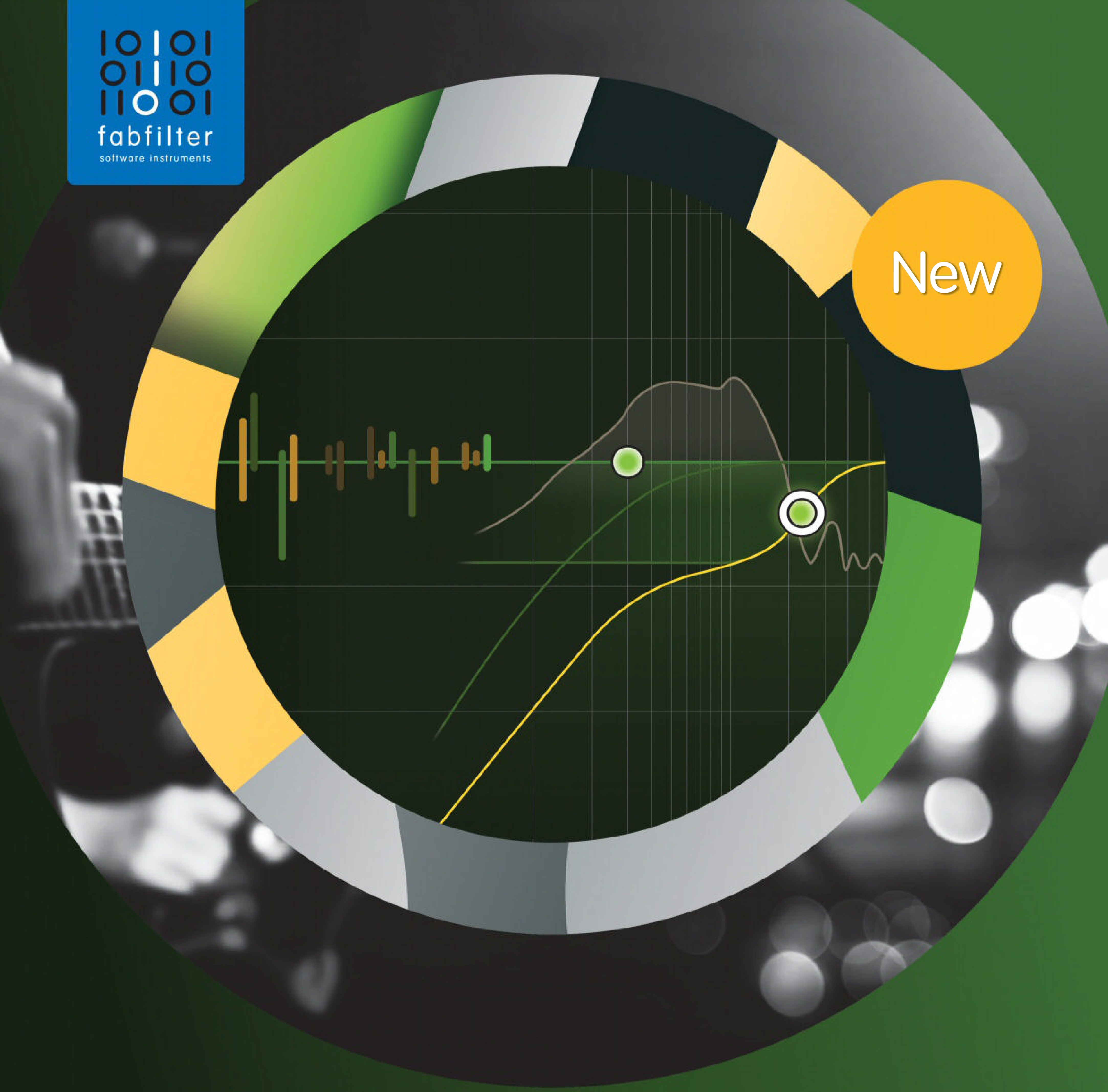
If the studio space incorporates multiple listening positions (mix engineer, client sofa, producer desk...) then the measuring mic can be moved to a new position, the AutoCal process repeated and the new optimisation profiles saved. GLM 4.1 then makes it very easy to switch between listening positions. Similarly, in complex multichannel systems, monitors can be saved as groups so that if, for example, the same audio programme is being mixed for multiple formats (maybe stereo, 5.1 and Dolby Atmos), it's a simple matter to create a group of monitors for each and switch between them. Once calibration is complete, GLM can then take on the role of a software-based monitor controller where it offers pretty much all the functionality you'd expect of a hardware device.

The W371A Woofer System

Moving away from GLM 4.1 for a moment, I'll spend a few paragraphs describing the new W371A Woofer System. While, thanks to its sub-25Hz low-frequency roll-off, it's so very tempting to describe the W371A as a subwoofer, it's really a very different animal. To begin with, rather than just fill in the lowest octave, the W371A can actually operate all the way up to 500Hz — well over two octaves higher than you'd normally expect the job of a subwoofer to be done and dusted. And furthermore, the W371A, thanks to its two widely spaced drivers, brings a significant element of low-frequency dispersion control to the party. Conventional subwoofers are almost always exclusively omnidirectional devices.

Having mentioned the widely spaced drivers of the W371A I'll now add a little flesh to those bones. The W371A is not small: it's a 0.4m deep and wide rectangular black box that stands just over 1.1m tall and weighs a not insignificant 61kg. Use of the terms 'stands' is appropriate because in many installations the W371A is an appropriate height on which to place the 8351B (or the equally compatible 8341A/8361A). In being plain, black, and obstinately rectilinear in form, the W371A has little of the lavish industrial design style of the 8351B, but part of the justification for that I suspect is that maximising internal volume for electro-acoustic needs trumps any softening of appearance.

Facing forward at the top of the W371A cabinet is a 365mm (14-inch) diameter bass driver, and facing backwards at the bottom of the cabinet is a 305mm (12-inch) bass driver accompanied by an adjacent flared slot reflex port. The two drivers operate »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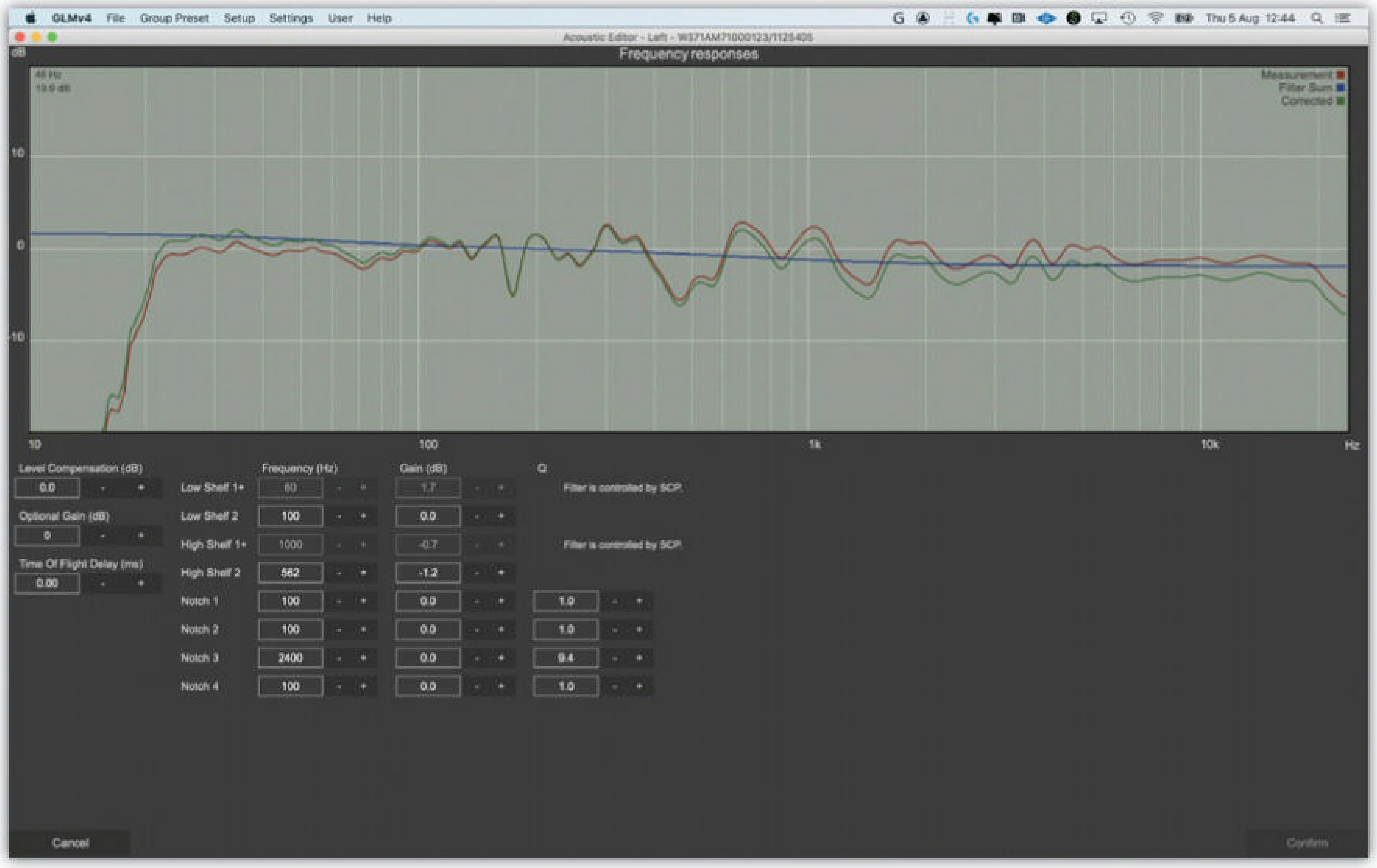

Screen 3: Correction graph for a full setup comprising two 8351Bs and a single W371A.

» in separate enclosures and are driven by 400W Class-D amplifiers. Their combined role is to turn the 8351B (or again, 8341A/8361A) into a four-way system, with their specific responsibilities, in terms of low bass to upper bass crossover integration and room optimisation, managed intelligently by GLM depending on the needs of the room optimisation and dispersion targets at each optimised listening position — hence the W371A being able to operate up to 500Hz. The result is that systems comprising the W371A and 8351B, (or 8341A/8361A), in addition to offering a remarkably flat and very wide-bandwidth in-room frequency response at multiple listening positions, can also offer multiple dispersion profiles all the way from around 60Hz up to frequencies where the 8351B itself begins to display cardioid characteristics.

The system dispersion control profiles I mentioned can be chosen to, for example, suppress floor or side wall reflections, or simply to continue, all the way down to low frequencies, the dispersion characteristics of the main speakers. The combination of GLM with the W371A and 8351B makes all this possible across multiple channels, with GLM doing all the difficult sums and making the decisions concerning exactly how, and at what frequency, the crossover between the 8351B and the W371A is best handled. It's pretty much as if the composite speaker system of 8351B and W371A is intelligently designed by GLM on the spot to suit the specific room acoustics and needs of the user.

Home & Away

I mentioned earlier that the 8351B, along with the majority of Genelec's monitoring products, incorporates an enclosure constructed from aluminium die-castings. The die-cast construction undoubtedly helps ensure that the enclosure makes minimal acoustic contribution in terms of panel resonance, but it also represents a pretty eye-watering investment in terms of tooling costs, and that I think has some significant implication in terms of the market sectors and applications Genelec pitch their monitors towards. Once a decision has been made to invest huge sums of cash into tooling, it's no good then aiming the product at a niche — rock & roll or EDM mix monitoring, for example. You need to attract customers from as broad a church as possible; from bedroom indie guitar pop bedroom studios to major, multichannel immersive experience stages. And as if to reinforce this idea, I had two contrasting opportunities to spend time listening to the 8351B and W371A. Firstly, a few weeks at home with just a stereo pair of 8351Bs and GLM, and secondly, a few hours in the control room of Studio B at London's Metropolis Studios with a system comprising a stereo pair of 8351Bs, the W371A, and of course again with GLM.

Home listening first. The 8351B loan period coincided with a house move that left me working temporarily in a pretty small (3 x 3.5m) and minimally treated room, and for some monitors, especially those like the 8351B blessed with significant low-frequency bandwidth, that would likely be a significant problem. Thanks to GLM and the point-source nature of the 8351B, however, the small size of the room turned out to be relatively benign. Firstly, GLM effectively suppressed the low-frequency room modes, and secondly, one of the advantages of point-source monitors is that they don't really care about listening distance so can be used effectively in the extreme nearfield. There comes a point with conventional multi-way monitors where a really close listening distance can begin to render individual drivers subjectively apparent (mainly through the off-axis cancellation nulls in the crossover region becoming more audibly significant), but with all the audio emanating from a single point in space that problem doesn't occur.

My overwhelming subjective response to the 8351B, optimised for the room and primary listening position by GLM, was one of tonal neutrality and bomb-proof professional competence. The 8351B exudes a sense of trustworthiness in that its subjective performance is completely without quirks or any particularly noticeable character — if it were an Instagram post there'd be no filter. Bass, despite the small room, but aided no doubt by GLM, was informative and clear with no obvious ported monitor traits. Midrange and high-frequency performance was similarly satisfying, with great image focus and detail portrayal without any monitor-borne artefacts. Those midrange mix details that we tend to get obsessed with

The 8351B/W371A setup at Metropolis Studio B.

— reverb tails and compression character, for example — were all easily audible and satisfyingly focused in the space between the monitors. The combination of a completely trustworthy and neutral tonal balance and outstanding midrange mix detail is, to my mind, one of the ultimate signs of a great monitor. You can often have one or the other, but both at the same time is much more unusual.

Metropolis Studios

Following my time with the 8351B at home, I decamped to Metropolis Studios to hear the combination of the 8351B and W371A in the Studio B control room. The system had been previously set up by Genelec, with GLM 4.1 configured to provide a few system response and dispersion options. Of course, being dropped into a less familiar environment to listen to an unusual monitoring system makes any definitive conclusion difficult to draw. Having said that, however, the experience left me with little doubt that the 8351B/W371A system is extraordinary. Bass bandwidth and level available from the W371A seemed subjectively unlimited, perfectly integrated with the midrange, and flawless in terms of pitch and dynamics. I listened to a wide variety of old favourites via Tidal, a couple of which I'd both mixed and played bass on, and heard countless elements that I'd simply never previously noticed: everything from new perspectives on how, say, reverbs integrate with the stereo image, to hearing significant instrumental or vocal performance details that I'd simply never been aware of before. It was seriously addictive.

As I described previously, GLM provides the opportunity to select from various pre-configured monitoring response and dispersion presets, and in Screen 3 some can be seen listed on the right. The presets include those that employ the dispersion control capabilities of the 8351B/W371A composite to suppress low/mid-frequency floor, back or side wall reflections, and these in particular made for fascinating listening. The subjective differences between dispersion options weren't apparent so much in tonal terms as they were in the way the system presented stereo focus and imaging. With side reflections suppressed, for example, the image seemed less expansive in terms of width but more explicit in terms of depth. But while the comparison of different presets was fascinating, I had some trouble deciding which one I preferred or which would offer the most value in terms of mix work. I think I'd need longer with the system, and probably an actual mix job to do, before I settled on a preference. I also suspect that 8351B/W371A dispersion control options might become somewhat more relevant in less well-controlled listening spaces than a Metropolis control room.

Back to the more fundamental performance of the 8351B/W371A at Metropolis. The system is undoubtedly a triumph of electro-acoustics and if I were lucky enough to be able to afford such a system I'd probably only hesitate for as long as it took to experience its few equally impressive competitors. I was left with a couple of questions, however. Firstly, if I were working in Metropolis Studio B with the 8351B/W371A, I'm not sure I'd ever feel the need to use the main monitors. And secondly, I wonder, when a monitoring system is so hugely capable in terms of revealing detail over such a wide bandwidth, how I'd ever be able to conclude that a mix is finished? I'm pretty sure I'd never know when to stop fiddling with it.

£ 8351B £3599, W371A £7859. Prices are per speaker, including VAT.
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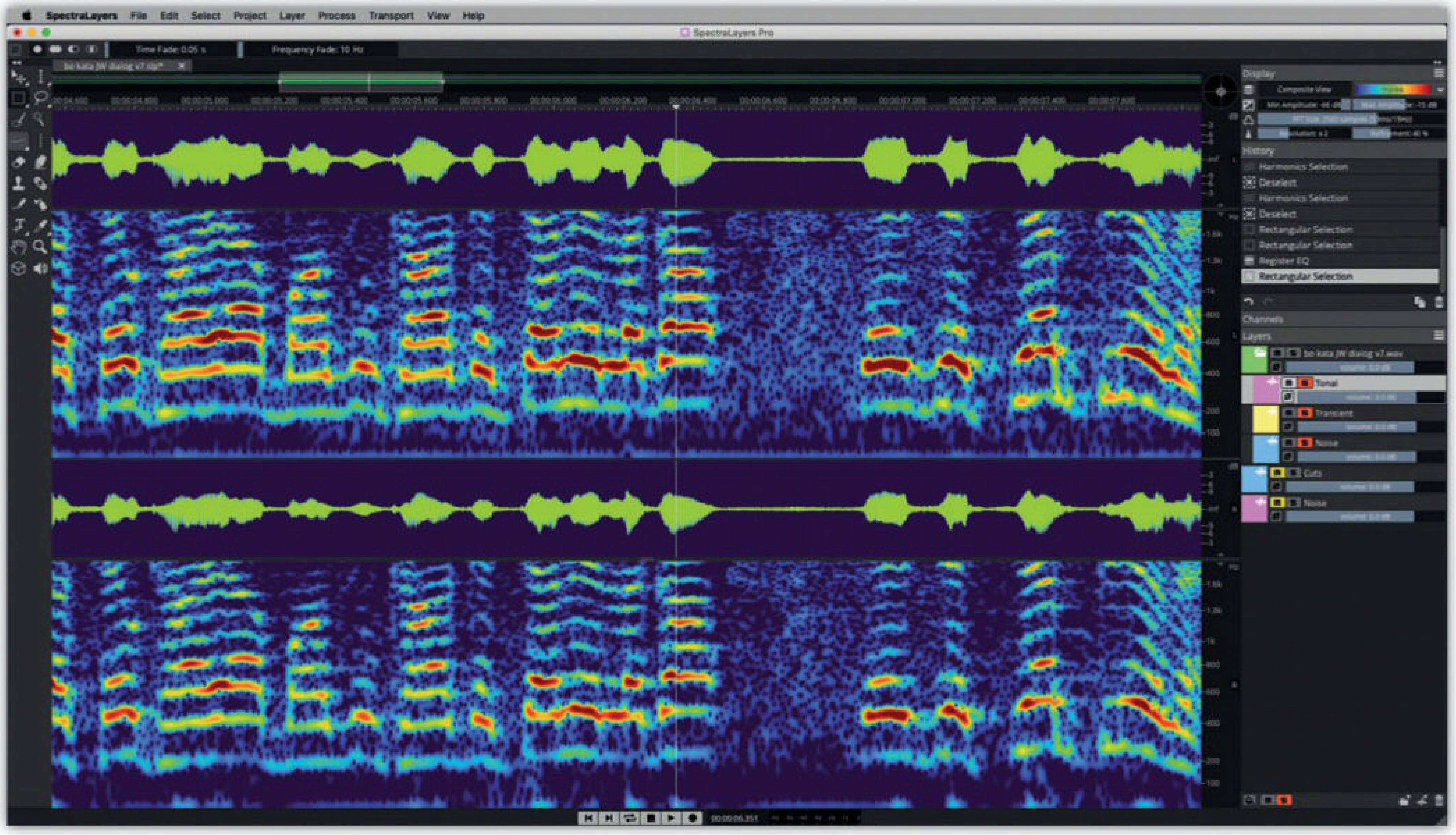


Steinberg SpectraLayers Pro 8

Spectral Editing Software

Steinberg are on a mission with SpectraLayers. So, what's new with Pro 8?

JOHN WALDEN

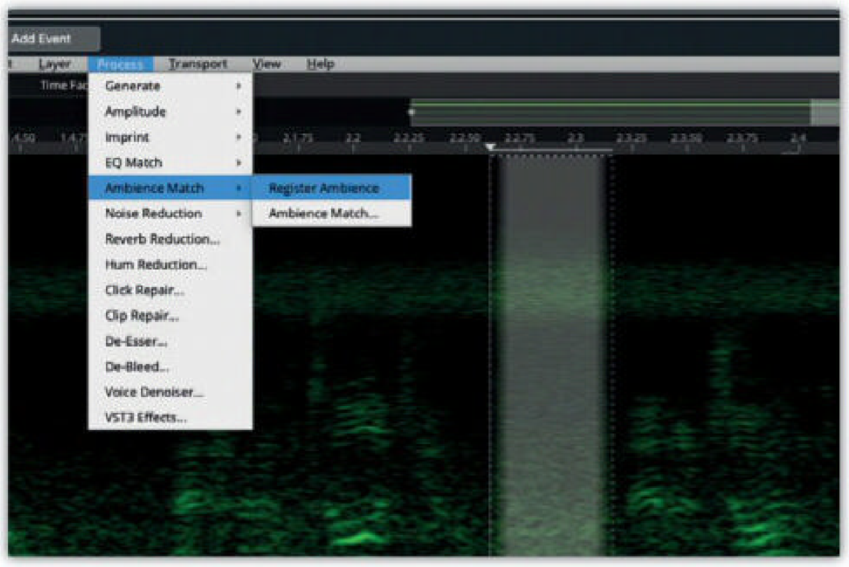
Steinberg acquired SpectraLayers in 2019, and *SOS* have covered both the Pro 6 (December 2019) and Pro 7 (February 2021) releases since then. However, less than 12 months on from the v7 release, Steinberg are back with SpectraLayers Pro 8. Existing users can be reassured that all the established functionality remains and, for those unfamiliar with the product (or spectral editing in general), you can dip into the *SOS* archives for a comprehensive catch-up. In this review, I'll focus on the key additions and improvements that the latest release delivers.

The Big Match

Amongst the new additions, perhaps two of the most eye-catching are the new 'second generation AI'-fuelled Ambience Match and EQ Match features. The most obvious application for Ambience Match would be within the film/TV ADR process. ADR is commonly used when the dialogue recorded on set is not of sufficient quality, but it can be problematic to make the studio-based replacement dialogue sound like it was recorded in the same environment as the on-screen action. Adding some of the background/ambient noise from the original on-set audio is generally required. SLP8's new Ambience Match feature makes this incredibly simple to do, allowing you to capture the background noise from one audio recording and then apply it evenly across any other audio, including the pristine, studio-recorded dialogue replacement.

This task can be performed within the standalone version of SLP8 but, for anyone working in a compatible DAW such as Nuendo or Cubase, it's likely to be more efficient to use SLP8 as an ARA2 plug-in. Once it's inserted as an ARA2 extension on your source track (the original on-set dialogue), you can simply select a time range that contains just background noise (between dialogue phrases, for example) and execute the Process / Ambience Match / Register Ambience command to 'capture' the required ambient sound profile. Once done, you can then select a different audio layer and execute Ambience Match; the captured ambience is then added evenly across the entire length of the audio within the selected layer. You could apply this ambience directly to your ADR track or, for greater flexibility, to an otherwise empty audio clip. The latter option lets you control the balance between ambience and the replacement dialogue track at the final mix stage.

The Ambience Match process is very efficient and generates a very even reproduction of your ambient noise capture, without any of the obvious signs of looping that can occur if you simply snag a bit of the noise and repeat that under your ADR. The process can obviously have applications beyond ADR, potentially letting you make any studio-based audio signal sound like it was recorded in a different environment with very particular ambience characteristics.



The new AI-based Ambience Match algorithm will be a boon for those doing lots of ADR work.

Taking the tonal characteristics of one audio signal (your EQ source) and applying them to another (your EQ target) is, of course, something that other software has made possible for quite some time. In that context, SLP8's new EQ Match might not seem too groundbreaking. However, like Ambience Match, it is remarkably easy to use (particularly via ARA2) and is built upon SLP8's AI spectral editing algorithms. The same simple two-stage process of registering an EQ curve from a reference audio layer, and then applying it to a target audio layer is involved, but you also get a percentage slider (from 100 percent to zero) to control how strongly the 'match' is applied.

There are lots of obvious applications here including in mastering, either to match your own mix's EQ to a reference, or to nudge multiple tracks within an album/EP project towards a similar overall EQ shape so they feel sonically cohesive. Maybe I simply got lucky with the materials I used in testing but, a few graphical gremlins within the ARA2 extension aside, I have to say that this was the best EQ matching process I can recall experiencing. I've no idea what's going on under SpectraLayers' hood, but it's clever stuff.

The EQ-matching process could also be very useful within the recording stage. For example, when you return to a song project to do further overdubs — guitar or vocals, for example — it can often be difficult to exactly match the tone, regardless of how carefully you try to recreate the original signal chain. I can imagine EQ Match usefully nudging your overdubs into sonic alignment with the original tracks.

I'm also sure there is potential in combining these two 'matching' processes. For example, could you capture both the ambience and EQ from a legendary Led Zeppelin drum track and apply it to your own drum mix? Could you match both ambience and EQ from a live performance guitar track and apply it to a studio overdub or two to replace some duff notes, so that the overdub blends in more easily to the rest of the live recording? SLP8's new matching features will undoubtedly provide some interesting experiments to explore.

Knock Out

Two processes for removing or reducing unwanted elements within audio have also received an AI-based enhancement: De-Bleed and Reverb Reduction. De-Bleed allows you to reduce the spill into one microphone based upon a recording from another microphone. The obvious application here is to help clean up a multitrack drum recording, allowing you to take, for example, kick and hi-hat bleed out of the track recorded with your snare mic. Of course, bleed can be one of things that helps give a drum mix a sense of cohesion but, if you do need to clean things up, De-Bleed has the advantage over conventional gating in that it can try to subtract the spill even when the two drums are played at the same time.

Applied to a single drum track the process is pretty slick, with a simple dialogue allowing you to select the appropriate layer to reduce the bleed from. You can also tweak the sensitivity of the algorithm and adjust the percentage of the reduction to find the best compromise between bleed reduction and any possible audio artefacts. The results are impressive, and I was able to clean up a snare track by De-Bleeding both kick and hi-hat elements in two passes. Auditioned in isolation, I could hear the very occasional artefact from the process, but these became inaudible when the snare was blended back into the overall drum mix. The only downside is that, for a complex multitrack recording, if you wanted to clean up everything, it might involve a lot of iterations of the De-Bleed process. That's a somewhat picky criticism, though, given how well it works.

Reverb Reduction has been available in SLP since v3, but this also gets the AI treatment in v8, making it easier to use. »

Steinberg SpectraLayers Pro 8
£257

PROS
- Ambience Match and EQ Match are excellent new features.
- Lots of new workflow improvements.
- Better ARA2 integration.

CONS
- Great to see such rapid progress but occasional users might baulk at the cost of keeping up.

SUMMARY
SpectraLayers Pro 8 brings some excellent new features and further workflow improvements. Steinberg are continuing to transform the power of spectral editing into a much less intimidating format.

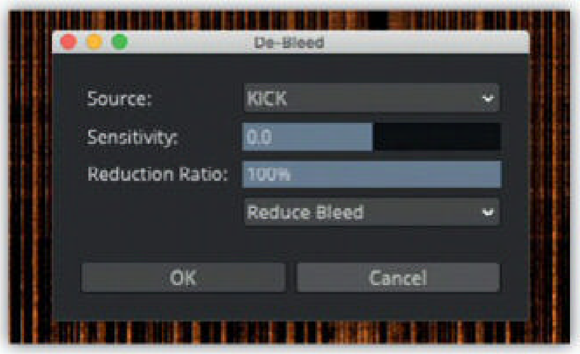
The new EQ Match process is an interesting proposition within a music mastering workflow, and the improved ARA2 integration makes for a streamlined process.

» You can adjust both the sensitivity of the reverb detection and the percentage of reduction. Equally, you can flip between reverb reduction or signal reduction, with the latter allowing you to isolate just the reverb if required. This combination of options provides plenty of both corrective and creative possibilities. Testing with a common application — reducing reverb on dialogue recorded in a room with lots of natural reverberation — proved very effective and could easily be used to attempt a rescue job on an otherwise unusable audio recording.

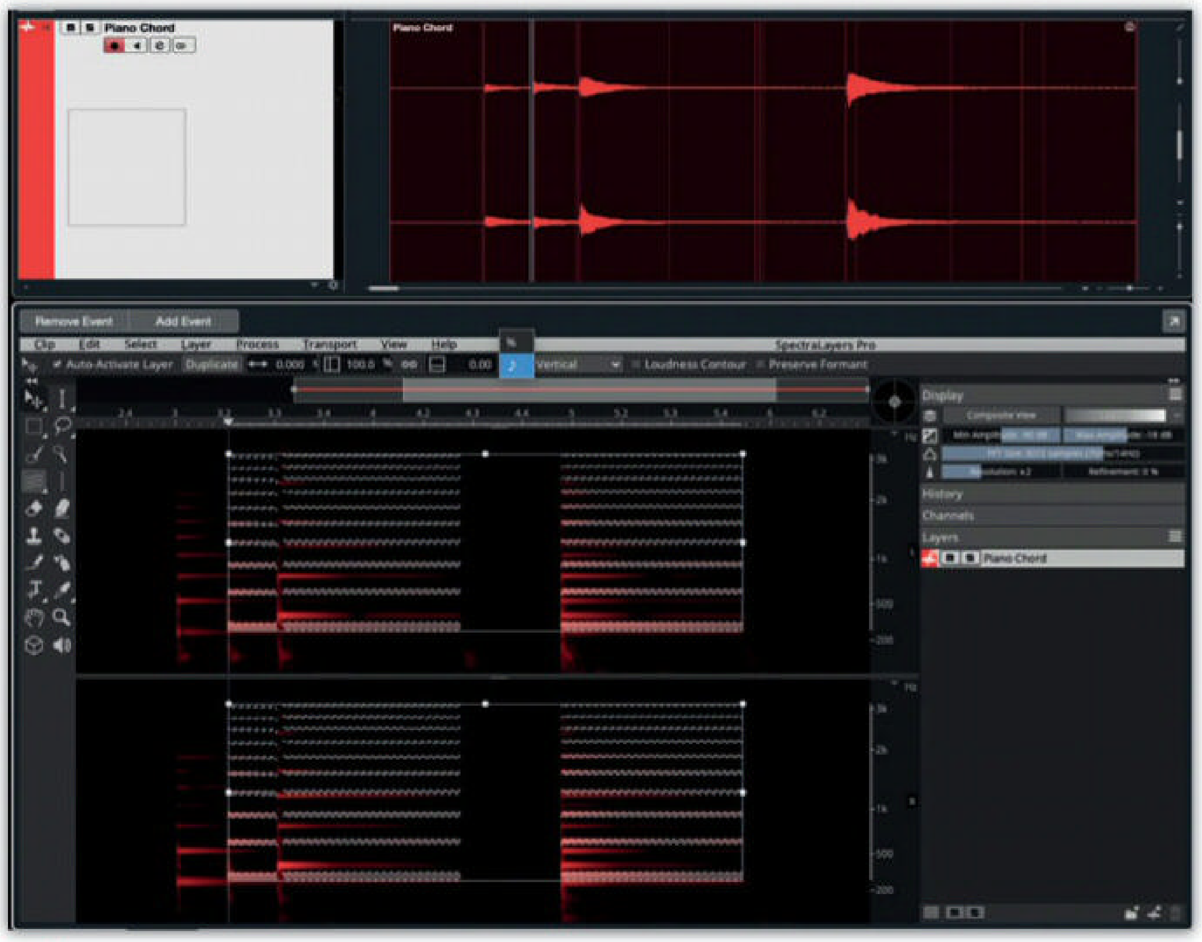
The Best Of The Rest

So, headlines aside, what else is new? Well, the revamped Reverb Reduction process is not the only improvement for cleaning dialogue; the Voice Denoiser has also been revamped. This now offers two different AI algorithms; one for the spoken voice (Noise) and one for sung vocals (Music), both of which try to isolate the voice from the rest of the audio. Presumably, Voice Denoiser's algorithms share technology with the stem unmixing process that was the star addition in the SLP7 release. Whatever is under the hood, though, the results can be very good and some combination of Voice Denoiser, Reverb Reduction and Noise Reduction can deliver dramatic improvements in a noisy, ambient voice recording. If the starting point is particularly bad, don't expect perfection, but 'useable' may still be a win in many circumstances.

Other improvements include the ability to set hard limits for both the Eraser and Amplifier tools; improved multiple selections (including saving multiple selections within a project); better options for selecting harmonics; improved repair for clipping and, if you have suitable audio hardware, support for sample rates up to 384kHz. Used alongside the harmonics selection options, the new ability to view pitch-shifting in semitone-based intervals has some interesting musical applications. For example, in a recording of piano



De-Bleed provides an interesting alternative to conventional gating for tasks such as cleaning up a multitrack drum recording.



Pitch-shifting within SLP8 can be now be done on the basis of semitones, as being used here to convert a major piano chord into a minor one by dropping the third down half a step.

chords, I was able to select the harmonics associated with just the major third of the chord and drop those by a semitone to create a minor chord. While the workflow is not as slick as Melodyne's polyphonic mode in this regard, it can certainly work and, for modest moves, the quality of the shifting is very good indeed.

Perhaps the most notable other improvement is in the integration of SLP8 within a host via ARA2. The main change here is that you can have independent SLP8 instances within the same ARA2 host project; not all clips being edited via SLP8 therefore have to appear as layers within a single SLP8 editor window. This makes it much easier to manage your workflow, allowing you to group related audio clips (for example, your drum tracks for De-Bleeding) into their own SLP8 instance. The SLP8 ARA2 sub-window also now includes Remove Event and Add Event buttons, allowing you to remove or add audio clips to a specific SLP8 extension instance as required. I'm sure there are still refinements to come for the ARA2 integration, but this is a very useful step forward for both music producers and post-production specialists.

Going Up?

Steinberg have undoubtedly delivered some excellent enhancements to SpectraLayers Pro since 2019. With impressive new features (such as stem unmixing), improved accessibility (complex processes that are less intimidating to use) and streamlined workflow (via AI algorithms to do the heavy lifting and ARA2 integration), it would therefore appear that they are investing heavily in the software and its development. This is good news for existing users and potential new users alike.

So, do the new developments provided by v8 scream 'buy me!'? There may be some existing users who might hesitate, particularly if their v7 update is still relatively fresh in the memory (and on the credit card statement). However, if SLP is an essential part of your workflow in dialogue post-production, drum mixing or audio restoration, v8's new features will get you better results and let you get them more efficiently; in a commercial context, the upgrade could pay for itself very quickly in enhanced productivity. For undecided potential new users, Steinberg do have a free trial available for download on their website, so you can try before you buy.

Quietly and confidently, Steinberg seem to be committed to bringing the power and potential of spectral editing into the audio mainstream. They have made great strides since 2019 and SLP8 is a further welcome step in that direction.

£ £257; upgrade from v7 £68.
Prices include VAT.
W www.steinberg.net

Warm Audio WA-8000

NEIL ROGERS

Valve Capacitor Microphone

In perhaps their most ambitious project yet, Warm Audio attempt to recreate the legendary Sony C800G.

I've had some great experiences with Warm Audio mics recently, and while reviewing their latest interpretation of a classic German tube mic for *Sound On Sound* a few months ago, I couldn't help but wonder if there was anything left for them in the fantasy mic locker to tackle. The answer to that was obviously yes, and I was pleasantly surprised when I peeked inside the sleek black mic case that arrived at my studio for review...

The Sony C800G is a curious, perhaps even unique beast in that despite not being a 'vintage' design (it was released in 1992), it holds an almost mythical status in engineering circles — and has an eye-watering price tag to match. This Japanese mic is still technically available to buy new off the shelf, but they don't seem to be in great supply and the price of used models has rocketed in the last few years to well past the £10,000 mark. This tells you how easy (or not) it is to get hold of one! New or secondhand, the price is still well outside what most home recordists or small studio owners can justify spending on a single microphone.

It's debatable if any mic is worth these kinds of sums of course, but I've used a C800G on a few sessions over the years and it is a fantastic-sounding microphone. Known for its bright but smooth sound, it's often associated with pop and rap productions. If you work in those genres and with a certain calibre of artist it can even be a necessity, and is certainly a good asset to have in your studio's mic collection.

Warm Audio aren't the first company to revisit the unusual-looking design of the C800G. I was impressed with the Golden Age Premier GA-800G I reviewed last year (www.soundonsound.com/reviews/golden-age-premier-ga-800g), but despite it feeling and sounding very authentic, its price was still going to be a significant barrier for many. Belgian company ZP Microphones' take on the Sony C800G is also still priced at the high end of the market, though unlike either the Warm Audio or Golden Age Premier models, it's a solid-state microphone and so doesn't have the characteristic heatsink. Hugh Robjohns reviewed the ZP800G in last month's issue of *Sound On Sound*.

As is their way, Warm Audio have attempted to shake things up and bring the sound of this studio classic into the realms of reality for many more engineers, and I was very intrigued to see if it could deliver the sound that I associate with one of the strangest-looking and best-sounding microphones that money can buy.

Canned Heat

The designers of the C800G were given a 'no compromise' brief to create the highest-performance microphone possible. Appearance was not an issue and, apart from the price, the most distinctive feature of the C800G is the large heatsink that protrudes from the rear of the microphone. This is an integral part of the design: the heatsink dissipates the heat generated by the valve in the mic's casing, which ensures a higher signal-to-noise ratio than other tube-based designs, as well as extending the valve's life. The valve is mounted in an aluminium case with a cooling chip attached to the bottom; this allows the heat to be transferred to the large external aluminium-alloy heatsink, and from there into the air surrounding it. It's not a subtle design feature by any means, but is symbolic of the original ethos behind the design. This is a mic to impress the ears and not the eyes!

I've reviewed several Warm Audio releases over the last few years and I've occasionally given them a bit of a hard time in the looks and accessories department. With the WA-8000, however, there is nothing to grumble about: the review mic arrived in an impressive hard case housing the mic itself, a shockmount, power supply and cabling, all of which look and feel very professional. The microphone looks very similar to its inspiration, albeit noticeably smaller. The heatsink is around half the size of the original's, and Warm Audio founder Bryce Young explained to me that they wanted to try to bring the heatsink down to a more practical size whilst still achieving the enhanced signal-to-noise ratio. Apparently, Warm Audio found that most of the heat given off by a C800G was dissipated from only half of its heatsink, and so made the decision to make theirs smaller. They did, however, take the extra precaution of increasing the thickness of the metal alloy material below where the tube is mounted. The result is a slightly less unusual-looking mic, although I did notice that the heatsink got a little hotter to the touch after being left on for a few hours than I recall with the original. It's safe to assume that this has been thoroughly tested, as Warm Audio do production runs in the thousands — which is a big part of how they achieve their prices. These relatively large production runs also help overcome the challenge of sourcing what can sometimes be quite esoteric components, such as valves. The WA-8000 uses a French military NOS 6AU6, while elsewhere the unit employs a Lundahl-made replica of the transformer used in the original C800G, a K67-style capsule (again much like the original), and a Gotham Swiss 7-pin XLR cable.

In Use

My first chance to use the WA-8000 was on a vocal recording session with a female artist I was working with for the first time.

Warm Audio WA-8000
£1159

PROS
- Successfully captures the essence of the original Sony C800G.
- Bright, smooth-sounding valve microphone.
- Tight, consistent proximity effect.
- Good all-round package.
- Less than a tenth of the price of an original!

CONS
- None.

SUMMARY
Warm Audio continue to push the boundaries on price versus performance with this convincing-sounding take on a classic studio vocal microphone.

After making sure she was on board with a little time spent experimenting, I set up a couple of different mic options to find a good fit for her voice. My client immediately liked how the WA-8000 sounded on her, and I was impressed with the familiar bright but smooth-sounding vocal that I was hearing in the control room. We worked on four jazz/funk-style songs during that session, and used the WA-8000 for two of the songs, preferring the fuller sound of a U47-style mic on the other tracks.

The next session was with a male singer I've worked with a lot, and whose voice I know can be a bit sibilant around the 8kHz range. I was curious to hear how the bright-sounding WA-8000 would handle this challenge, and was pleasantly surprised at how well it suited his voice. That element to his voice didn't suddenly disappear, but felt softer and generally more like a natural part of his vocal sound. On another male singer I've worked with a lot, I again felt immediately comfortable using the WA-8000, particularly the way it seemed to 'frame' his voice, which has a lot going on around the 2-3 kHz range. Having extra information higher up the frequency range seemed to make this less prominent — although the restrained proximity effect did little to add any low-frequency authority to his voice. On that last issue, it can feel like this style of mic is lacking in the bottom end at the tracking stage, but its tight, consistent low end is one of the reasons it's so well regarded. Testing the mic outside the pace of a real-world session revealed how the WA-8000 successfully captured this characteristic of the original. I got one of the guys at the studio to talk into the mic at varying distances, and the authoritative low-frequency part of his voice stayed remarkably consistent. You can find audio examples from all of these sessions at https://sosm.ag/warm-wa-8000-media.

Summing Up

In an ideal world, I would be able to report to you how the WA-8000 directly compared to the original but, like most *SOS* readers, I don't have a C800G sitting in my mic collection. I have used the original before, however, and I found that the WA-8000 captured more than enough of that mic's essence. Judging this new release on its own merits I found it to be an excellent-sounding vocal mic that gave a usefully different perspective to other microphones in my collection, including a U87 and a U47-style mic.

The headline feature of both the Sony and the Warm Audio mics is their ability to sound bright whilst still adding the subtle smoothness of a good valve microphone. I've talked exclusively about this new release in the context of recording vocals, but I also found myself using it on acoustic guitar with good results, and you would find plenty of other uses for it when not recording vocals. As a self-confessed gear snob, it's taken a while for Warm Audio to win me over but, like the last two microphones of theirs that I've reviewed, the WA-8000 does an excellent job of bringing the sound of a studio classic within reach of the masses.

£ £1159 including VAT.
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TalkBack

Gemini Major

WILLIAM STOKES

Malawi-born Benn Gilbert Kamoto aka Gemini Major moved to Johannesburg in 2009, where he has been based ever since. The producer behind an array of South African stars such as Nasty C, Sho Madjozi and Cassper Nyovest, he has twice won Producer Of The Year at the South African Hip-Hop Awards. Alongside this he is also an artist, with a formidable catalogue of original material to his name. As he explains, not all of his artistic ambitions fit with the genre expectations currently placed on him. "The scene here is very fast," he says of Johannesburg. "If you want to be an artist then this is where you want to be. Because everyone's here. It's where people come to find their 'American Dream'. I've been producing in Johannesburg and in South African hip-hop since around 2014 or 2015. That's when everything kicked off. Now I'm producing record after record. But, as an artist... I just want to make beautiful African music."

At the moment I can't stop listening to
Oh, man! It's just African music in general right now. My soul is there. I don't know if I actually have a favourite, I'm just in love with so much African music right now. Most recently, it's Tems. She's from Nigeria. She's really dope.

The project I'm most proud of
I have to say, it's the Gemini Major record I'm working on right now. The one I'm about to drop. I've never been so proud of myself. It's called *Island Water*. It's an EP. It's a transition from what people are used to getting from me — a new direction. But it feels so right, because I feel like I've found myself and connected with my soul. I've always been making music as a hip-hop artist, but now with this project I've really connected with myself. As a producer I've just been banging out music, all the time. But now with this project — as an artist and as a producer — I'm really proud that I've found myself, I've found my sound and it feels good. Everything is connecting and it's where I've always wanted the energy to be at. It's African fusion. I'm mixing a lot of sounds, experimenting with a lot of African sounds. I'm from Malawi, that's where I grew up, so it's quite easy for me to go back to where I started.

The first thing I look for in a studio
Studio monitors. I always like to see what I'm working with when I step foot in a studio. That's the first thing I usually notice. "Oh, those are nice monitors! Let me try them out!" They have to bang! Right now I've got a pair of PMCs. They're my favourite. If I had to upgrade, I'd upgrade those.

The person I would consider my mentor
I think it has be my older brother. I've always applied his theories of life to my music journey. He's guided me through everything. So he is definitely my mentor. He taught me responsibility, working hard and just... aiming for better. He showed me that at a very young age. Tried to teach me that as much as he could. I'm a very humble person, simply because of the values my brother taught me. About how to manoeuvre through life, about patience. Most importantly he was God-fearing, you know? I feel like when you're intact spiritually it always brings you back to ground. Keep your faith up because it's something you definitely need in life, especially if you're pursuing music. You need a lot of faith! And patience, and those things that he definitely taught me early on. At some points he wasn't supportive of my music — he had his own dreams for me. I was still going to school and doing my advanced diploma when I was 19. But he told me, "If you're sure you want to do this, then you've really got to do it." So I couldn't fail!

The biggest misconception about the role of a producer
I'm a music producer, right? And then, there are people who make beats. You get it!? Music producers and beat makers... people tend to put both in the same bag, and it's very irritating! If you're going to work with me, let's have a studio session. It's fair enough if you're in a different country — but if you're in the same country I feel disrespected when someone just says to me "OK, send me beats." I'm a music producer. I want to work. I want to have an input. What if I have an idea that could change the whole song? My role isn't just to create sounds. Anyone can make a beat, but to produce a song is a whole thing. You have to see it through.

My go-to reference track or album
Right now it has to be *Made In Lagos* by Wizkid. I just love it sonically. It's beautiful: vocally, the way the instrumentation works, the drums... Everything is just clean, you know!? Every time I'm trying to get a bounce, to create a sound, I always go back to that. When I hear something beautiful I always try to pay attention to it, to learn from it.

My desert island studio item
I'd take my laptop! That's the studio! I use FL Studio and Logic. Right now I mostly use FL Studio. It's very flexible. I've also worked on PreSonus Studio One a lot. Earlier on I was using FL Studio to produce beats and then taking

it into PreSonus to record vocals there. Then I ended up getting into Logic. On the desert island I'd just produce all day. What else could I do!?

The piece of gear I'd bring back into production

I have two: firstly the [*Yamaha*] NS10 studio monitors. They should still be making those. And also the [*Neumann*] U87. The vintage one. Just, the sound! The warmth... It's beautiful. And with the NS10s you're able to hear everything. When you're making the mistakes, you're going to hear the mistakes! I like to have two sets of studio monitors, one to bang and one to come to for reference and expose all the rubbishness in the sounds. The NS10s do that perfectly. If your mix is lacking then you're definitely going to hear it.

The advice I'd give myself of 10 years ago

Stick to the plan. When I started making music, as a producer I wanted to be the best at everything, making all kinds of music. And that's OK — I'm doing fine at that and still exploring, but as an artist I wanted to be an African artist, not a rap or hop-hip artist. But somehow, while coming up, I got caught up in the hype and started making a lot more hip-hop as an artist. Not that it was bad, that worked very well, but most of my success in my career comes from that. I feel like I would be somewhere different if I'd actually have just stuck to the plan and believed in what I was doing. But obviously I lost direction with everything that was happening here, with the people I'm working with, with everyone's hip-hop. The industry that we have here does not accommodate a lot of those other sounds. The plan should have been not just to cater for where I am right now, which is South Africa. I should have thought of the whole of Africa, the whole world. My dreams should have been bigger. They are now!

Gamechanger Audio Light Pedal

Spring Reverb Stompbox

We all love a good spring reverb, and this one offers much more than most.

PAUL WHITE

If you've seem our reviews of Gamechanger Audio's previous devices, you'll know that whenever they unveil a new product you should expect something that's very far from the ordinary.

This splendid tradition continues with their Light Pedal, which is a spring reverb — but not as we know it! The Light Pedal does have three springs that can be operated in the usual configuration, but Gamechanger Audio seem to have taken another bunch of 'what if?' ideas and really run with them. Notably, in addition to the output from the springs being picked up by the usual electromagnetic transducers, this device also has also LEDs and infra-red optical sensors arranged around the springs, detecting the vibrations from various points along their length, to produce different flavours of reverb.

Overview

Springs are very sensitive by nature, but this pedal cleverly features a switchable shock sensor circuit that shuts off the audio signal as soon as a mechanical impact is detected: with the rear-panel sensor switch set to Hard, you won't get a twangfest whenever you stomp on the pedal. Also, the springs are mounted on a separate circuit board that sits in a resilient suspension cradle to mitigate the effects of vibration on the springs.

As you can see from the main image, the springs are visible in the centre of the top panel, with the most conventional controls being on the same panel, to the left. The red and white LED illumination follows the playing dynamics and modulation,

Gamechanger Audio Light Pedal

£319

PROS
- Wide range of spring reverb sounds, both traditional and radically effected.
- Expression pedal support.
- Easy to use.

CONS
- A stereo output would have been nice!

SUMMARY
The Light Pedal upholds the Gamechanger Audio Devices 'be different' ethos, by incorporating optical sensors and some unusual modulation effects into a spring reverb.

where relevant. On the left you can set the Spring and Optical wet levels, the Dry level and the overall Tone. You can also select, using a small toggle switch, whether you want the reverb tails to continue when you bypass the pedal. Both the input and output are on unbalanced jacks, providing mono-only operation.

To the right is where the fun happens, and there's also a mini toggle switch at the top of that section to set the footswitch to latching or non-latching behaviour, enabling effects to be brought into play briefly when needed. A rotary switch selects an operating mode with a further rotary Ctrl knob below to adjust the key parameter for the current selection. If you fancy varying the control function during performance when you have no free hands, you can also hook up an expression pedal.

In Optics mode, you can move between different optical sensor pairs using the Control knob, to change which sensors are employed. In Sweep mode, on the other hand, Control introduces a gentle modulation to the optical component, which I suspect is produced by modulating the contribution of the various optical sensors. Trem applies amplitude modulation to the reverb, Control adjusting the rate. The modulation depth of both Sweep and Trim is affected by picking intensity, such that it gets deeper as the note decays.

Reflect adds in some warm delay via a feedback path, sending the delayed reverb back to the input in a very murky, lo-fi kind of way that I found rather addictive. Feedback mode is a novel addition, with some of the output energy from the spring being fed back to the input to create self-oscillation, and you can tweak the Drive control along with the Spring and Optical levels to change the feedback sound from squeals to drones. If you use the momentary footswitch mode or an expression pedal to control the amount of feedback, it's possible to create some harmonically complex sustain effects and drones — a bit like a waterphone meeting an underwater cathedral organ. I can see a lot of use for this one in ambient music.

Harmonic mode adds a shimmer type of effect, in which the reverb is pitch-shifted and some of it fed back to the input; the Ctrl knob seems to adjusts the degree of shift. The sound is, as expected, much more 'springy'

As well as the mono unbalanced in/out jacks, there's an expression pedal input — which can help you get the most out of Feedback mode in particular.

than from a normal shimmer reverb and it can sound particularly effective fed through further modulation effects, such as a rotary speaker pedal or plug-in. At the lower settings of Ctrl the effect is dark and grumpy, whereas at higher settings it shifts upwards, adding more sparkle to the sound. Again, the sound takes on a waterphone type of character, and it is oddly addictive. Below that there's a Drive control for controlling how hard the spring is driven and a gate, which can either cause the reverb to decay more rapidly or act as a reverb ducker if turned in the opposite direction, causing the reverb level to swell at the end of a note. In fact, it behaves more like a decay time control than a traditional gate.

The Light Fantastic?

As a straight spring reverb, the Light Pedal has an endearingly vintage sound, with plenty of watery twang and a long sustain time. Switch to the optical sensors and the sound becomes less twangy and generally warmer, so there's plenty of scope for setting up 'normal' reverb treatments, especially as the Gate control allows for tweaking the decay time. Blending spring and optical reverbs often produces the most musical results.

Both the Sweep and Trem modulation options are capable of adding useful types of modulation but, if you want something a bit more experimental, there's a wealth of sonic exploration to be enjoyed in the Feedback mode. Really, you'll need to hook up an expression pedal to get the most out of that mode, as just leaving the effect switched on invites chaos. Alternatively, though, you can set the footswich to non-latching and just bring it in at the end of notes. Shimmer is also usefully unusual: less polite-sounding than a typical digital shimmer effect, with occasional overtones somewhat reminiscent of whales and dolphins singing in a scrapyard. I liked it!

Ultimately, then, the Light Pedal works really well as a conventional spring reverb, albeit a mono one, but to use it only for that purpose would be to miss out on a lot of creative potential: whether processing guitar, synths, voice or even drums, there's endless scope to create unique sounds. Having first heard this pedal fighting to be noticed above the noisy chaos of a NAMM show in 2020, I wasn't quite sure what to expect but, having now tried it for myself, I can confirm that it is possible to create some nicely subtle reverb effects alongside the more gnarly things the pedal can do, especially if teamed with a fuzz box or Gamechanger's own Plasma pedal. Ideally, I would have liked the ability to split the spring and optical modes to different outputs — that could be the basis for some really interesting stereo effects, and I'll be interested to see if that can be added in a future model — but as things stand, it's already capable of some different-sounding and really engaging effects; I've heard nothing else quite like it.

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Proximity Effect

In Theory And In Practice

We all know that close miking boosts the low end. But why? New research from DPA explains that some instruments are more affected by proximity effect than others.

SAM INGLIS

Proximity effect is the tendency for low end to become exaggerated as a directional microphone is moved closer to the source. Sometimes we can use it to our advantage in recording; other times it's a nuisance that has to be worked around. Yet, although anyone who's ever used a microphone will undoubtedly have come across it, proximity effect remains poorly understood. How many of us can confidently say that we understand why proximity effect happens, or how that theory translates to practice?

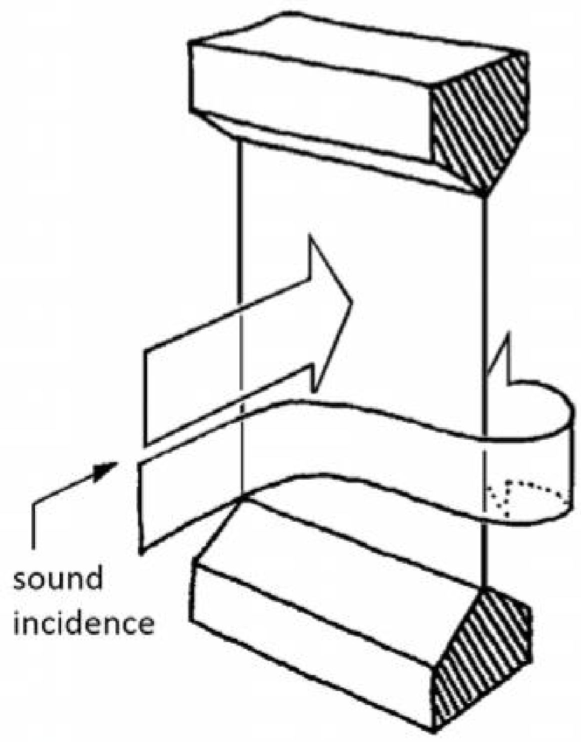
Eddy Brixen and DPA Microphones have carried out some fascinating research into proximity effect, which we'll be sharing later on in this article. As we'll see, it turns out that real-world sources and microphones don't always behave quite as you'd expect, and that proximity effect is highly dependent on the type of source you're miking, as well as on the mic choice and placement.

To understand this research properly, though, we need a proper explanation of the phenomenon in question. How, and why, does proximity effect occur?

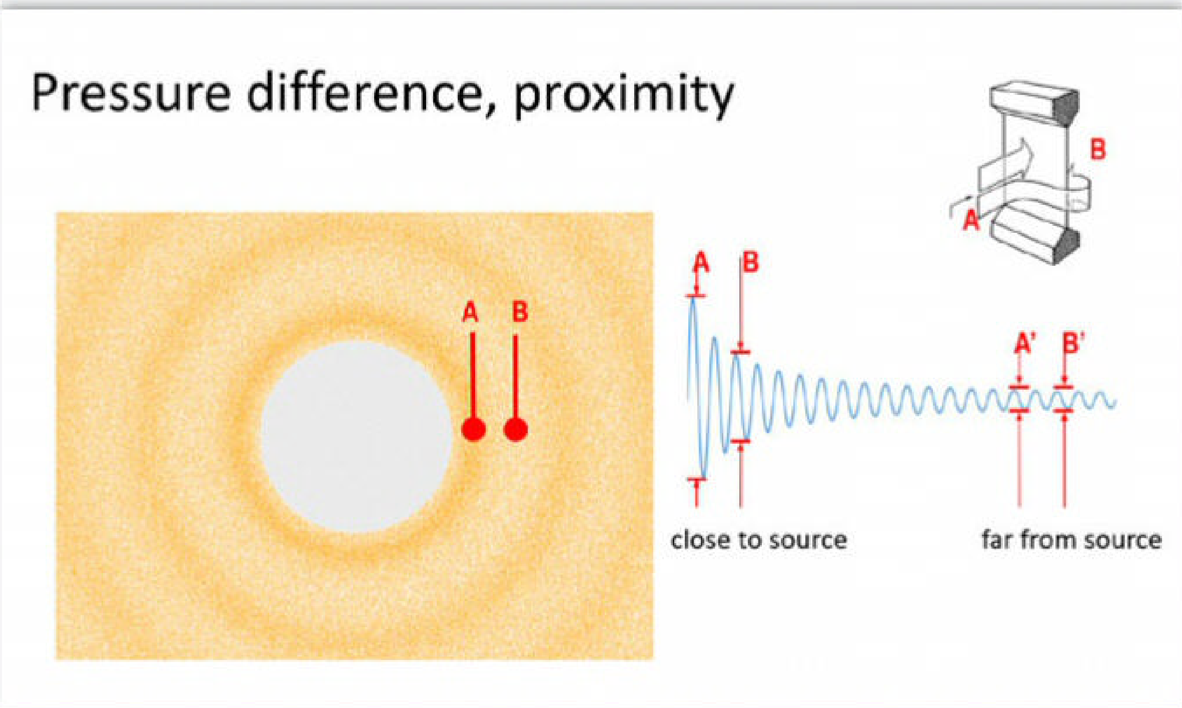
What Is Proximity Effect?

There are two fundamental types of microphone. One measures air pressure, like a very fast-acting barometer; in effect, its output reflects the excursion of the diaphragm from its rest position. The other measures the rate of change of air pressure, and its output represents the velocity with which its diaphragm is moving.

A pressure mic has a diaphragm that is open to the air only on one side, and is intrinsically omnidirectional. By contrast,



A simplified view of a pressure-gradient or velocity microphone. Sound arriving on-axis has to travel further to reach the rear of the diaphragm than the front.



As this schematic diagram illustrates, the difference in acoustic power across the microphone due to the inverse square law is negligible when far from the source, but can be significant when close to the source.

the diaphragm of a velocity or 'pressure gradient' mic is open on both sides, and the mic has a figure-8 polar pattern. A pressure mic does not exhibit proximity effect, but a pressure gradient mic does.

The diaphragm of a pressure-gradient mic is set in motion by pressure differences between the front and back of the diaphragm. These are created because the front and back encounter an acoustic wave at slightly different points in its cycle. The path length from the front to the rear is typically around 1cm. Peaks and troughs arriving at the front of the mic thus reach the rear of the diaphragm fractionally later, so at any given moment, a pressure difference is created across the diaphragm.

Power Down

However, there's another factor that can cause a pressure difference across the diaphragm: the inverse square law. This states that, for a source that radiates sound equally in all directions, sound level falls according to the square of the distance from the source. Put another way, the sound pressure level falls 6dB for each doubling of the distance from the source.

If we place a pressure-gradient mic just 1cm from a such a source, the rear side of the diaphragm is twice as far from it as the front side. This means the rear of the diaphragm isn't only encountering the sound wave slightly later than the front: it's also encountering a version of it that's half as loud! By contrast, if our mic is 10 metres from the source, the difference in acoustic power between front and back due to the additional 1cm path length is negligible.

But why does this affect the low-frequency response of a microphone? Our hypothetical 1cm difference in path length from front to back is fixed, but the wavelength of sound varies with frequency. A 5kHz sine wave has a wavelength of 7cm or so, whereas the wavelength of a 50Hz sine wave is almost 7 metres. If both waves have the same peak-to-peak amplitude, the 5kHz sine wave will generate a far greater pressure gradient across a 1cm gap than the 50Hz sine wave. Any pressure gradient created by the inverse square law will be negligible compared to that created by the 5kHz wave, but not when compared to the very small pressure gradient generated by the 50Hz wave. In other words, the *relative* contribution of the inverse square law doesn't only depend on distance, but also on frequency.

This discrepancy between the pressure gradient created by low and high frequencies gives the simplest form of pressure-gradient microphone a naturally rising frequency response: its sensitivity increases at 6dB per octave. To flatten the response, this must be compensated for, either using an electrical filter or by damping the membrane. And this compensation further increases the contribution of the inverse square law at low frequencies.

The Magic Metre

In practice, with a pure pressure-gradient microphone, proximity effect begins to become noticeable at about 1m from a point source, and steadily increases as the mic is moved closer.

Secondary polar patterns such as cardioid, hypercardioid, wide cardioid and so on are obtained by combining pressure and pressure-gradient operation in different proportions. The pressure component remains immune to proximity effect, so the overall degree of 'bass tip-up' depends on the relative contributions of each. In a cardioid microphone, for example,

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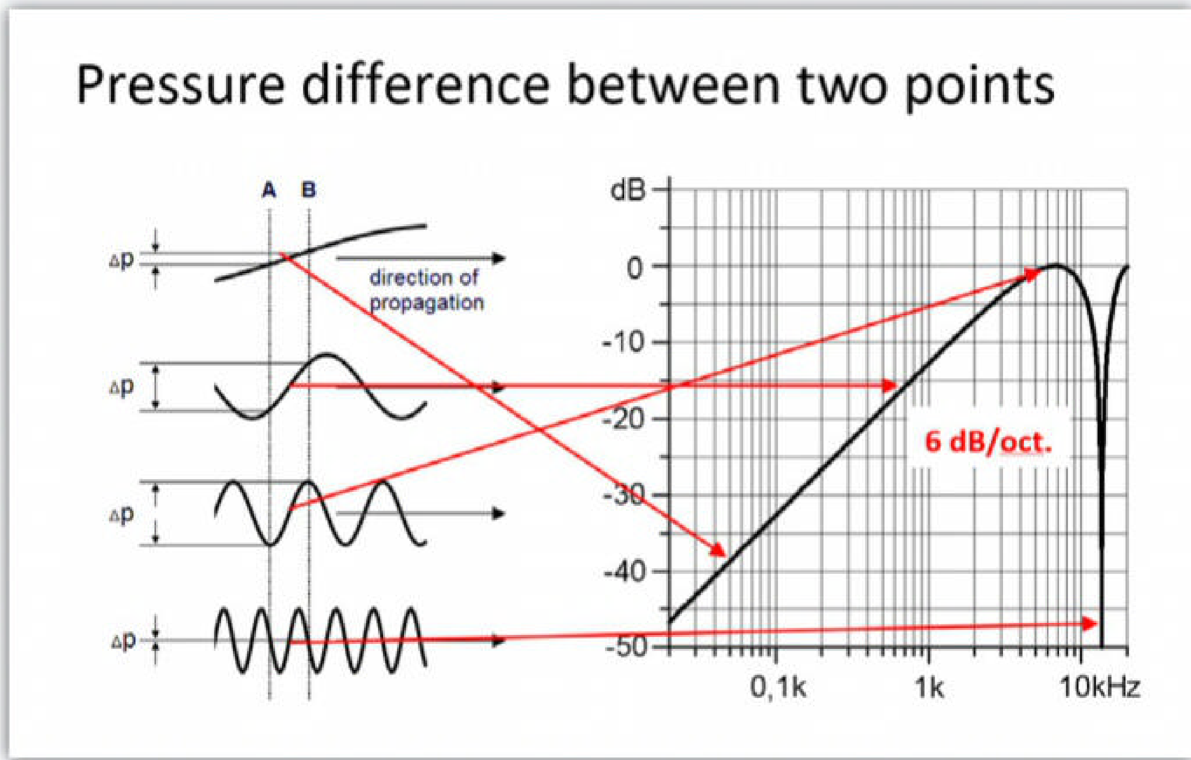
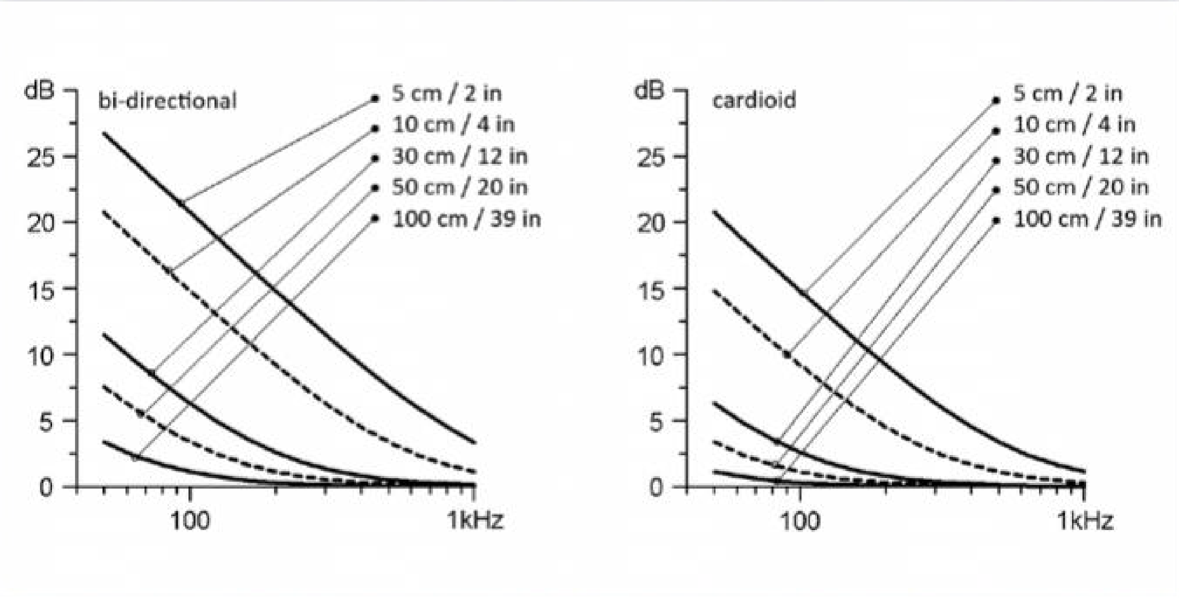


This diagram explains why a pure velocity microphone has a naturally rising frequency response. The pressure difference across the diaphragm from A to B is largest when the wavelength is exactly twice that distance, and falls progressively as the wavelength increases. (At even higher frequencies, a deep null is created where the distance AB is equal to the wavelength, but that's another story.)



This diagram shows the theoretical rise in low-frequency response due to the proximity effect in an ideal bi-directional (figure-8) and cardioid microphone, with a point source radiating sound in free space. At a 5cm distance, the figure-8 mic is boosted by more than 20dB at 100Hz!

pressure and pressure-gradient components are combined equally, so the overall proximity effect is half that of a figure-8 microphone.

Large-diaphragm capacitor mics often have two diaphragms either side of a central electrode, each with a cardioid pattern. This allows the overall polar pattern to be varied; for example, by combining the contributions of these two diaphragms in the same polarity, we get something resembling omnidirectional pickup. By reversing the polarity of one, we get a nominally figure-8 pattern. The contribution of the inverse square law is cancelled or enhanced respectively, so proximity effect behaves as you'd expect.

Managing Proximity Effect

Many directional microphones are designed to be used close up to the sound source either some or all of the time. In this context, proximity effect can result in a very boomy and muddy sound. Again, therefore, designers allow this to be compensated for. In a typical studio mic, we might have an optional high-pass filter that can be switched in for close placement or left out for distant use. Stage mics, however, are often intended only for close-up use, and will have their low-frequency response curtailed at source, either through the acoustical design of the mic or occasionally by the use of a transformer with a restricted low-frequency response.

Something that's not always appreciated is that the proximity effect can actually work to our advantage in a noisy environment such as a busy stage, because it helps to discriminate wanted from unwanted audio. A directional mic equalised to give a flat frequency response at, say, 5cm, will sharply attenuate low-frequency spill from more distant sources. Furthermore, the proximity effect is only manifested in the pressure-gradient component of the mic's pickup and hence within a figure-8 pattern, even if the mic itself is cardioid. So, in effect, when a singer addresses the mic from the front, his or her voice is boosted in the low frequencies, but sources to the sides of the mic are not.

This property can also be exploited by the performer talking 'across' rather than directly into the mic to reduce proximity effect where it's too prominent. Likewise, when miking sources such as guitar cabs, the mic can be angled to reduce proximity effect.

Points, Planes And Lines

The explanation of proximity effect outlined above makes some theoretical assumptions. Most importantly, it assumes we are miking up a source that radiates sound equally in all directions: a so-called point source. Staying in the realm of theory, however, we can also hypothesise two other types of sound source. A line source radiates sound 360 degrees in the horizontal plane, but has no dispersion in the vertical plane. A plane source, meanwhile, shows no dispersion at all. It simply projects a coherent wavefront directly forwards.

The inverse square law doesn't apply to line or plane sources. Because a plane source exhibits no dispersion, there is no loss of power with distance: in theory, its output forms a perfect beam that is equally loud at all distances from the source. A line source is a 'halfway house' between plane and point, with a theoretical attenuation of 3dB for each doubling of distance, rather than the 6dB seen with a point source. Hence, in theory, a directional microphone will encounter no proximity effect at all with a plane source, and with a line source, the proximity effect will be halved compared with a point source.

These ideals can be useful approximations for the behaviour of real-world sound sources. For example, many PA systems are designed to share the property of a line source whereby sound disperses only in the horizontal plane and not in the vertical plane. This minimises unwanted reflections from sound radiating upwards and downwards, and helps to ensure that the sound level does not drop off too much between the front and back of the hall.

However, ideals are just that, and no real sound source behaves exactly like a perfect point, plane or line. Musical instruments and loudspeakers are complex radiators that exhibit elements of all three characteristics, often varying with distance and frequency. Adding to the fun is the fact that we never encounter them in free space: our microphones always capture a mixture of directly radiated and reflected sound.

Proximity In Practice

Eddy Brixen and DPA set out to measure the actual contribution of proximity effect when real-world sound sources are recorded with real microphones. Five common sound sources were captured using six different DPA microphones, ranging from the omnidirectional 4007 to the supercardioid 4099. Of the mics on test, five are designed to produce a flat frequency response at 30-60 cm from the source, whereas the 4018V is a specialised vocal mic intended for close-up use. Measurements were made, where possible, at seven different distances, starting at 1cm and doubling every time until 64cm was reached; however, the physical design of some of the mics precluded use at 1 or 2 cm.

The sound sources tested were a coaxial two-way loudspeaker, a bass drum, a trumpet, an acoustic guitar and a grand piano. The results were intriguing and in some cases counter-intuitive. Least affected by proximity effect was the trumpet, which produces relatively little information below 200Hz in any case. However, the loudspeaker,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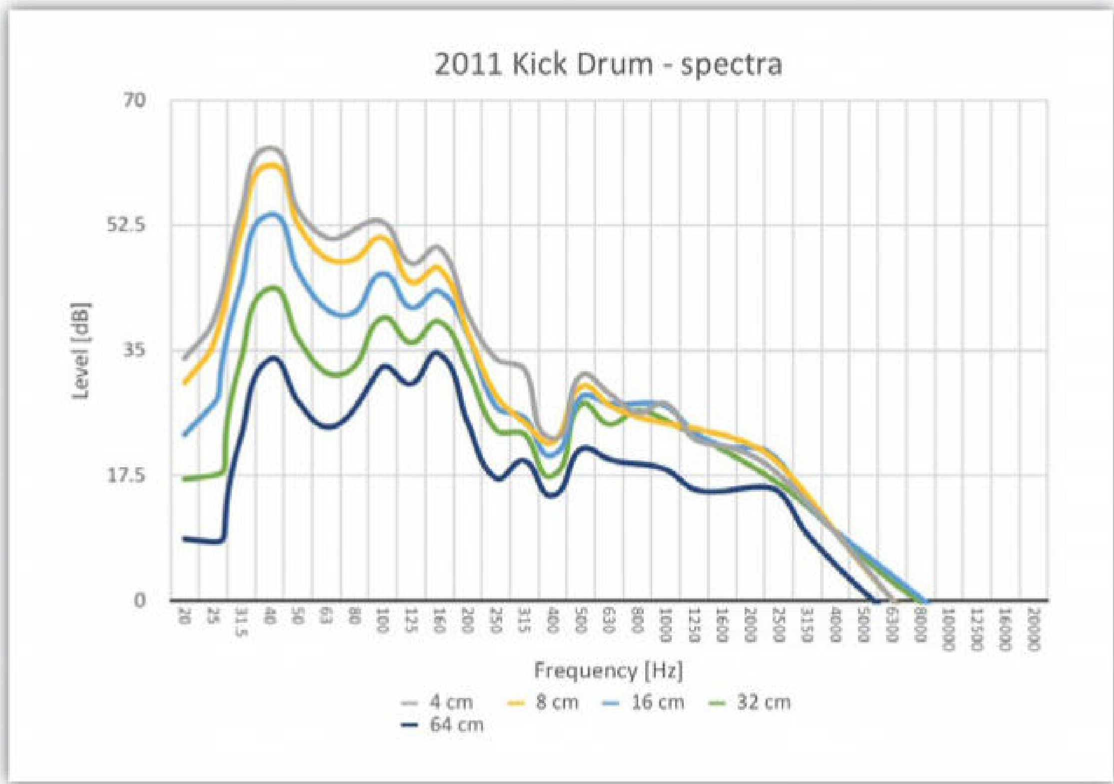
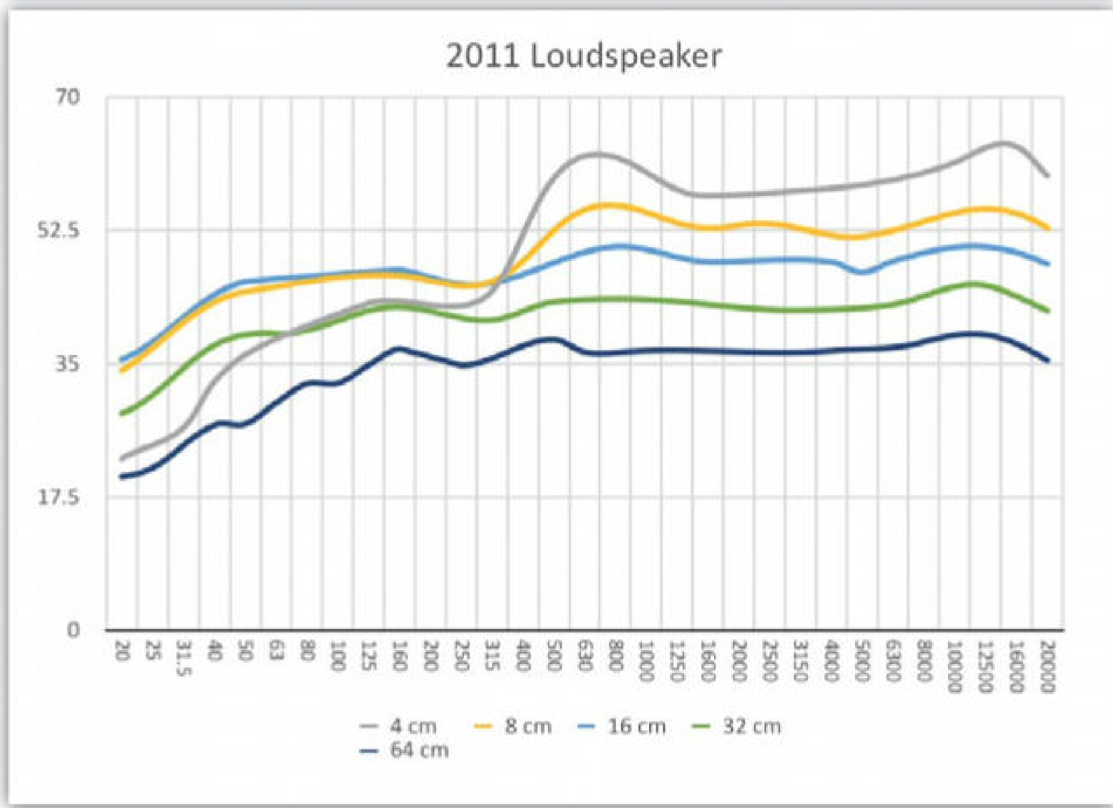
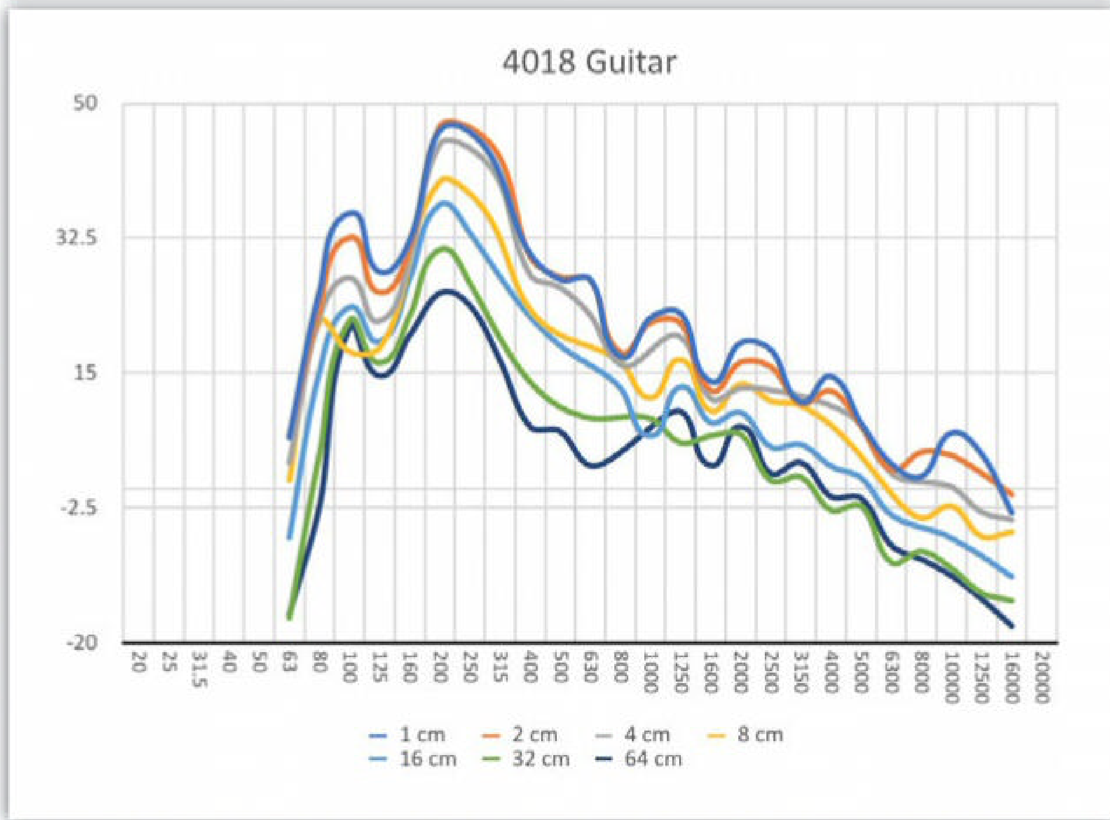
These graphs show how the frequency spectrum of three sources recorded with a DPA 2011 cardioid microphone varies with distance. The kick drum exhibits conventional proximity effect at least until the distance is reduced to 8cm. By contrast, the spectrum of the guitar is similar at all distances, with almost no proximity effect in evidence. The coaxial two-way loudspeaker gives interesting results: some proximity effect is apparent when the distance is halved from 64 to 32 cm, but below that, the bass level plateaus and eventually drops away again (probably for other reasons).







» the grand piano and the acoustic guitar all displayed a similar tendency. As you'd expect, low-frequency pickup is exaggerated as the distance from mic to source is reduced — but only up to a point. With the loudspeaker and grand piano, for example, there's little change in the low end once the mics are 16cm or closer.

A Sense Of Perspective

We can understand what's happening here by means of a visual analogy. The further away we get from something, the more it behaves like a point source. The planet Jupiter is massive, but because it's so far away, it appears as a mere dot in the night sky. Conversely, even small items can start to fill up our field of vision when we get close to them. In exactly the same way, sound sources that only subtend a tiny angle at distance become less and less point-like from a closer perspective. As we get closer to the source, it becomes more plane-like, and eventually this tendency takes over, meaning that the proximity effect plateaus.

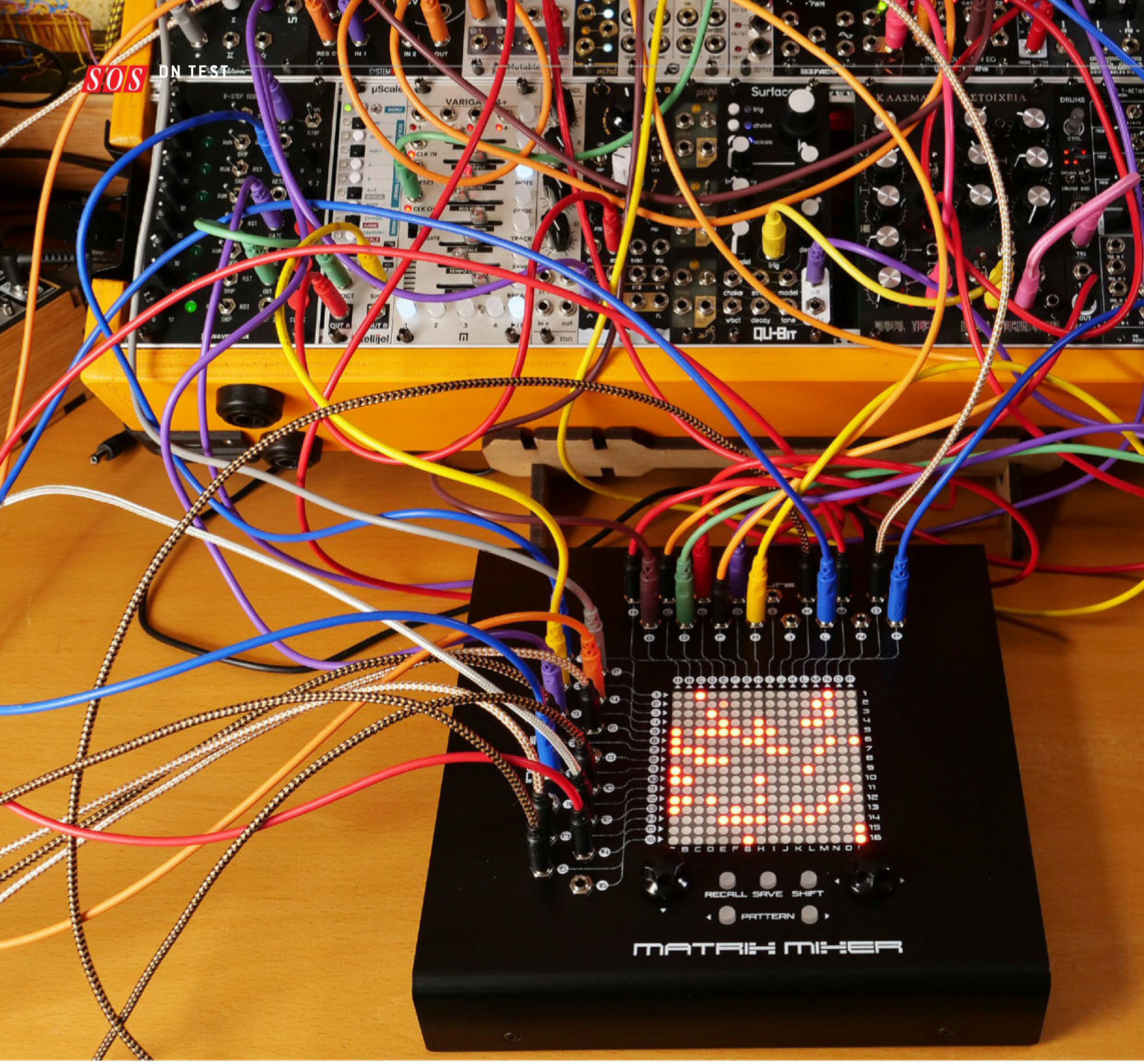
This, incidentally, is why large bass drivers aren't necessarily a good thing in nearfield monitors. The tweeter behaves as a point source at all listening distances, but the output of the woofer becomes increasingly plane-like as the listener moves towards it. The net result is that the level balance between woofer and tweeter changes, and with it the overall frequency response of the monitor. (Eddy's measurments show that the bass level actually falls at 4cm, perhaps suggesting that the shorter mic-speaker distance is outweighed by a reduced contribution from the speaker's bass ports.) It also means that proximity effect is relatively negligible when miking large speakers such as are found in typical guitar amps. The complex acoustical phenomena that are generated around the dome in the centre of the cone have a much greater influence on the tone than any variation in bass level with distance.

Compared with the other sources, the bass drum behaves much more like a true point source, with proximity effect continuing to increase as the mic is moved closer.

What this research shows is that within the normal range of close-miking, there

The soundboard of an acoustic guitar acts like a plane source, so there is very little proximity effect to be encountered even when the mics are moved close to it. Whether that's a useful mic position in practice is another story...

isn't always a straightforward relationship between mic distance and 'bass tip-up'. What's more, this relationship can change depending on the angle of the mic relative to the source. The acoustic guitar is a good example. Miked behind the bridge, as in this research, the soundboard acts like a plane source; but the soundhole may behave more like a point source, which is one reason why miking directly in front of it can yield a very boomy sound. It's just one more example of that universal principle of sound recording: if it sounds right, it is right. And, as Joe Meek didn't go on to say, if it sounds wrong, the proximity effect is only one possible cause!



Erica Synths Matrix Mixer

ROBIN VINCENT

Patch Matrix

Erica Synths' Matrix Mixer could change the whole way you think about modular.

The modulation matrix has long been a staple of synthesizer signal routing. In its traditional form, you have a column of sources on the left and row of destinations along the top and you use pins to make connections between them. It's the sort of pin matrix found famously on the EMS VC3 'Putney' synthesizer. More modern matrixes employ digital connections, using buttons or software. The main advantage of this method of signal routing is that you can make many connections quickly and easily and form complex configurations in a very compact space.

Usually, a matrix system will be built into a particular synthesizer's structure, with set sources and destinations. Erica Synths' Matrix Mixer is pulled directly from their Syntrx synthesizer, but breaks free from all that structured nonsense to offer a 16x16 matrix that can route any source of audio (either external or Eurorack) or CV signal to any destination with a snazzy array of LEDs and a pair of encoders. It's housed in a good-sized standalone box — you get the impression that once they made the decision not to try to squeeze it to fit into Eurorack format they could relax and build it into something that sits very solidly on the desk and has quite

Erica Synths Matrix Mixer
£499

PROS
- Organisation out of chaos.
- Inspires new ideas.
- Large enough to mix many ideas.
- Patch saving and instant recall.
- MIDI pattern select.
- Interfaces with external instruments as well as Eurorack.
- Looks fantastic.

CONS
- No CV control over patterns.
- Can get very busy.
- Dual-encoder interface can get taxing.

SUMMARY
The Matrix Mixer can bring order to the chaos of your Eurorack signal routing while sparking new thought patterns, developing patches and introducing variations at the touch of a button.

a commanding presence. This is the sort of device that wants to be centre stage. You'll find as you get into it that everything wants to go through it, and it moves very quickly from an overblown utility to a creative tool that shapes your entire workflow.

Connections

The connections are comprised of 16 DC-coupled inputs and outputs arranged in a zigzagged collection of patch sockets on the front panel in which you can easily get lost. Out the back, the first two inputs and outputs are mirrored on quarter-inch jacks for external instruments; there's also a MIDI In port and a fun on/off switch that's labelled backwards. The star of the show is the LED-infused Matrix panel, which is set deliciously off-centre and labelled 1-16 for the inputs and A-P for the outputs. The grid is populated by 256 LEDs which you can turn on and off with a pair of clicking encoders; one deals in horizontals, the other in »

verticals. Grab the encoders and you end up chasing connections about the place like you're playing Centipede in an arcade.

Making It Work

To connect an input to an output, you simply use the encoders to locate the intersection and push down on either to enable the connection. You can connect one input to 16 outputs, or many inputs to one or more outputs. You could see it as an overly elaborate mult as well as an overly simple mixer, and that's exactly what it is. All you need then is imagination. So, at a basic level you could take an LFO and route it to up to 16 destinations. But why not combine it with a couple of LFOs, or take a sequence to different oscillators, or the same trigger to different events? You could take an audio signal though different effects in parallel or series, or wire up all your filters and choose which ones to route your audio to. You can mix the outputs of many things into fewer things or different things; you could combine voltage from a keyboard with the output of a sequencer for transposition, or generate interesting waveshapes with colliding envelopes. The more you play, the more possibilities seem to emerge, and 16 channels gives you the space to do several things at once: some modulation here, some audio mixing there, some feedback loops over in that corner while mixing triggers in another. You find yourself designing patches around the Matrix Mixer and pulling in modules you might not have included in a long time. It makes you reassess how you build patches and how signal flows, and enables you to be more courageous and interesting.

Digital Trickery

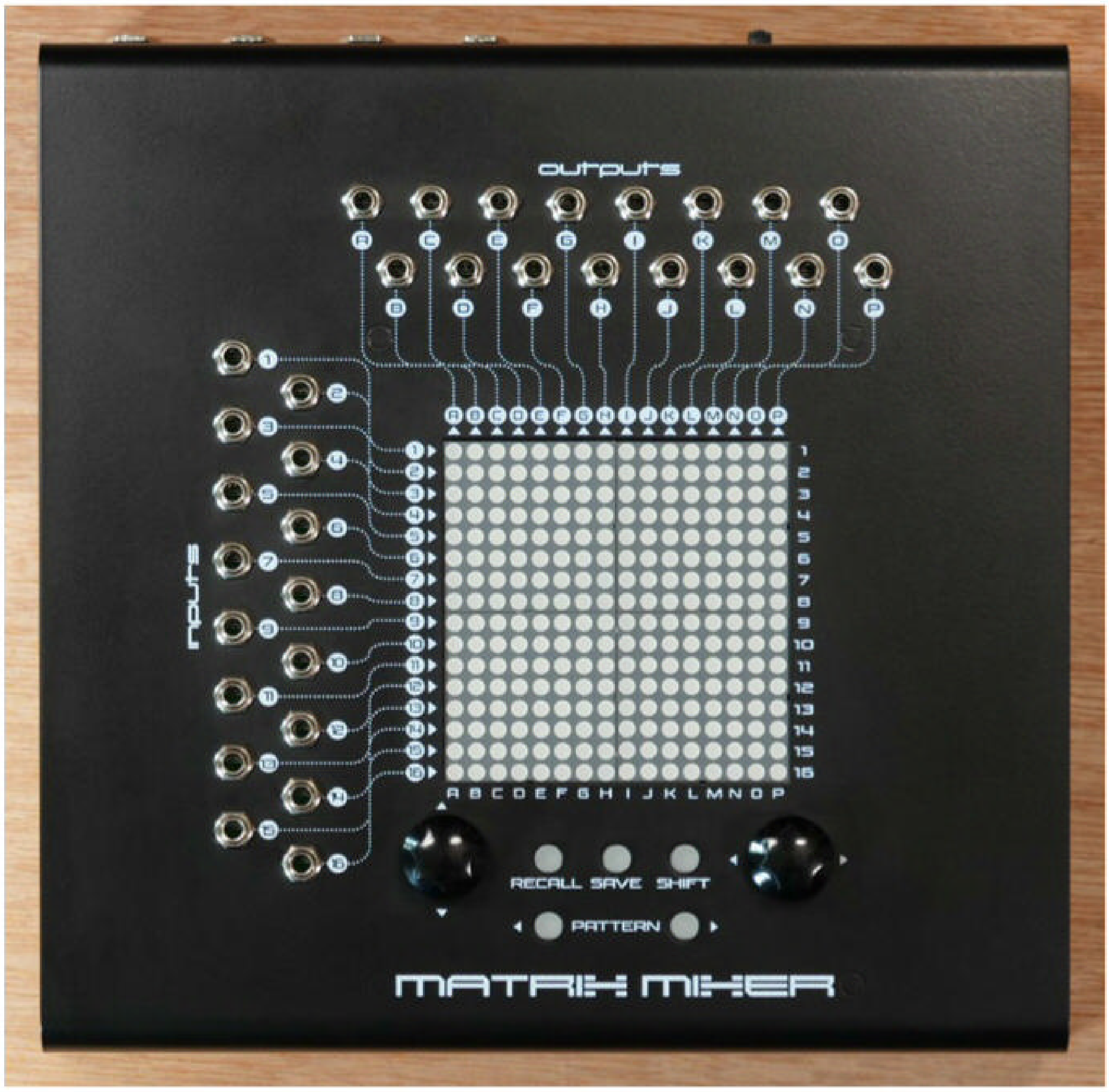
The Matrix Mixer has a few tricks up its digital sleeves which you won't find on analogue matrix mixers. The big one is the ability to save and recall your entire patch. This starts off mildly useful because things can get complex very quickly and it's a good way of trying out different ideas, comparing and organising them. Then it dawns on you that it's a clever way of introducing variation and changes to your patch that you can drop in mid-performance. With the touch of button you can re-route your oscillators to different filters, switch sequences, transpose, turn effects on and off, disable or enable sounds without having to physically re-patch anything. With every patch you're now thinking about layers of change and variation. What might have been an interesting but repetitive patch can run off into lots of different directions that are all instantly recallable and you find yourself in an unusual position in Eurorack where you might feasibly have a verse, chorus and middle eight structure at your fingertips. Alternatively, if you like leaving things connected, you could see it as way of storing Eurorack presets, which is an interesting concept by itself.

Another, more unwieldy feature is the ability to randomise the patch. With a couple of button presses, your carefully crafted patch can be completely destroyed as all the wrong things find themselves patched into each other. However, you can bend it into a more useful tool by defining how much of the grid the randomisation process affects, and how chaotic the randomisation is. Keeping it confined to a certain area or relating to a certain task can generate some interesting results — but you use it at your own risk! There's also no undo, so always save your patch before rolling the dice.

Mixing

I mentioned that the Matrix Mixer is simple when it comes to mixing signals, and that becomes a factor once you start experimenting. When combining signals from multiple inputs the output is just a sum of the input; there's no auto-balancing going on and so it's quite easy to run into clipping and distortion issues with audio, or flattened CV modulations. Erica Synths have provided

Round the back we find a MIDI In socket, and quarter-inch duplicates of the first two inputs and outputs.



Whilst it might have been possible to squeeze the Matrix Mixer into a Eurorack module, Erica have taken the right decision to make it a standalone device. The resulting unit measures 225 x 226 x 44mm.

some help with this by building in a couple of levels of attenuation. Each Matrix connection can be turned on or off by pushing the right encoder, while the left encoder it steps through 30 percent and 70 percent levels, visually represented by a slightly dimmer LED. This enables you to mix both audio and CV signals with a little bit more very welcome control.

In terms of simplicity, though, that's all the mixing nuance you're going to get. There's no panning, no mute without navigating the centipede to the right connection and then turning it off, and no solo without a radical change in patch. To get around these limitations, I ran all my audio inputs through a SSF Muton module first so I could easily turn audio channels on and off. You could do a similar thing using different Matrix patterns, but sometimes it's easier to reach out and mute something. For stereo returns from effect modules, I could run them to two outputs and use the attenuation levels to do a bit of spatial balancing. But the Matrix Mixer is always going to fall short when it comes to being what we normally think of as a mixer, and attempting to build those features in would have made it a very different and much larger device.

Digital Vs Analogue

Perhaps the biggest missing feature is the lack of CV control over the Matrix patterns. It would have been great to be able to send it a pulse to change to the next pattern, or have patterns selected by voltage values so you could run in a sequence to automate the change of patch. Instead, there's a MIDI input which can do exactly that using Program Change numbers, but that's less convenient in Eurorack.

Chasing the LEDs about with the encoders works well and I appreciate the digital abilities that gives us, but it is not as quick or intuitive as making a connection by poking it with a pin or breaking a connection by pulling one out. As it's digital, you feel that this could be resolved by making the LED grid somehow touch-enabled so you could put your finger on an LED to turn it on or off. This would probably make for a much more expensive device but would give us back an analogue feel to the Matrix.

Lastly, with 16 things plugged in and out it's easy to lose track of what module is plugged into which socket. It would have been nice if there were scribble strips alongside each socket so you could annotate it. I guess a bit of masking tape would do the job.

Conclusion

The Matrix Mixer radically changed my approach to Eurorack. It pushed me into all sorts of ideas and experiments that I had never considered before. The ability to swap filters, to move gate patterns to different devices, to suddenly stuff a bunch of audio into the QuBit Data Bender and then swap it to the Mutable Beads, to turn delays on and off, transpose two different sequences from a single keyboard on three different oscillators and so on is amazing. It's one of those devices that blows your mind with the possibilities and has you wiring in modules you've neglected in your thirst to fill every socket and try every angle. Sixteen channels is enough to run sequences, effects and modulations in the same patch and then have them all changed and rearranged through the selection of the next pattern. If you're doing a performance with a couple of cases of modular it's a complete game-changer.

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INSIDE TRACK

SECRETS OF THE MIX ENGINEERS

Travis Harrington

Rod Wave's chart-topping album *Soulfly* was not only recorded in hotel rooms, but mixed and mastered on the road too. Travis Harrington was the engineer who made it work.

Rod Wave (left) and Travis Harrington recording in a hotel room in Mobile, Alabama.

'Street Runner'

Written by Rodarius Green, Lukas Payne, Sterling Reynolds, Thomas Horton & Ruth Berhe
Produced by Karltin Bankz, LondnBlue & TnTXD

»

PAUL TINGEN

"In the music industry the relationship between the producer and the artist used to be central. But what's happened more recently, particularly in rap, is that the relationship between the engineer and the artist has come more to the forefront. Many artists are now in the studio with just the engineer, who has to bring other skills to this partnership as well, like being able to produce and mix."

Travis Harrington is talking about his work with singer and rapper Rod Wave, and his point has been echoed by several other engineers in the Inside Track series who have built long-standing working relationships with particular artists. Among them are Patrizio Pigliapoco, who is engineer and mixer for Chris Brown, Bainz for Young Thug and Gunna, Tillie Mann for Migos and Lil Baby, and Todd Hurtt for Polo G, while examples outside rap include Josh Gudwin for Justin Bieber and Stuart White for Beyoncé.

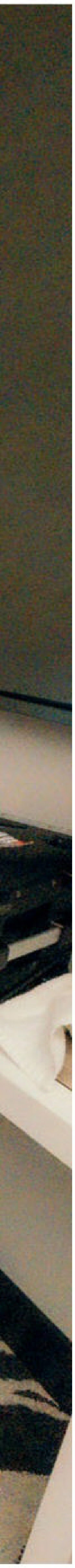
The above-mentioned artists thrive on a personal relationship with a trusted person in the studio to help them get their vocals and musical ideas down. This trend has been going on for a number of years, and in rap and particularly trap it is steered by another trend, which is that producers now make beats completely independently from the artists, and often don't even meet them. Instead, artists develop their vocal parts to these beats with only their trusted engineer in the room.

Back On The Road

Harrington's work with singer and rapper Rod Wave exemplifies this new trend. But there is another aspect of their work that sets them apart: Wave and Harrington record almost exclusively on the road, with the latter setting up his portable Mixing Is Art studio in hotel rooms and other non-studio spaces.

"Rod and I first met in 2018, at 11th Street Studios in Atlanta," recalls Harrington. "We hit it off, and jumped on the road together. Rod is kind of shy and not too keen on working in studios, while I am used to building a studio wherever. In the past you

Travis Harrington's portable setup is based around an Avalon 737 input channel, a Universal Audio Apollo 8 audio interface and a MacBook running Pro Tools. Note also the small JBL loudspeaker.



needed a studio to record in, but my generation is used to working anywhere, whether in a hotel room, a regular house, and so on. Rod and I built a process and a sound around that."

Harrington started working with Wave around the time the artist was recording *PTSD*, his fifth mix tape. He'd signed to Alamo Records in June 2019, and the mixtape was released later that year. One track, 'Heart Once Ice', went viral on YouTube and TikTok, and kickstarted his career. The song ended up going two times platinum in the US. Like the rest of the album, it was mixed by star mixer Fabian Marasciullo. Harrington worked on a few of the songs on the mixtape, and from there his working relationship with Wave developed quickly.

"I recorded and mixed Rod's first album, *Ghetto Gospel*, and then also *Pray 4 Love*, and most recently *Soulfly*. *Ghetto Gospel* is the first project that we did completely together. It is when we figured out exactly what we wanted to do. It is where Rod really started to realise the vision he has in his head, and started to home in on the sound he is after. We recorded the majority of the songs on these albums on tour. That is when most of the inspiration comes to Rod.

"We always find time to make records when on tour. You have to squeeze that in there. Rod is one of these artists who you have to catch in the moment, when he is going through the pain or whatever he's going through when he's writing these songs. He can be up all night after a show, and can call me at any time during the night. After the recordings I mix and master as we go, in my own time, on the road, using my Dre Beats Studio 3 headphones and occasionally my JBL 104-BT speakers."

These improvised circumstances haven't prevented Harrington's work with Wave sounding as good as anything in today's marketplace, or *Soulfly* from reaching number one on the US album charts. Waves' two previous albums also reached number two and number 10, and both went platinum in the US. "Back in the day it was about the equipment and the studio, the SSL desk and so on. But nowadays you can create your own environment, and that creates a better vibe, a better place for the artist to loosen up in and bring their own energy to the music and the recording. Working with Rod in hotel rooms gives me a real

Mixing Is Art: a studio in a suitcase!

opportunity to capture that. He'll call me from the hotel room above me, at three or four or sometimes at six in the morning, and we get to work!"

Dead Is Good

There are not many musicians, or engineers, who like to start work at six in the morning, but Harrington says he's happy to function on just a few hours of sleep each night when on tour, and notes that over time the complains from other hotel guests about these early morning sessions have gone down, "maybe because the hotels have gotten better".

Harrington also notes that he's become so accustomed to setting up recording studios in unconventional places, that he can "walk into a hotel room and pretty much immediately let Rod know whether we will be able to get a good sound, or whether it will be super difficult. Typically I want carpet, soft wallpaper, more furniture, and have a sound that is as dead as possible. Certain hotel rooms have been amazing for this. I can then bring the life back in during the mix. But some hotel rooms are very 'pingy' which is what I call hollow, reflective rooms.

"The song 'Brace Face' on *Ghetto Gospel*, for example, was recorded in an almost empty hotel room, with hardwood floors, and cabinets of granite. So I had to put Rod underneath a blanket. He recorded that entire song sitting underneath a comforter, with the microphone in there with him. I'd rather have as little acoustics at the beginning, and then bring them back in during mixing, than have extra noise and not be able to get the vocal sound that I want. So I often use acoustic screens like the Aston Halo Shadow, Kaotica Eyeball and/or the Sterling Audio VMS Vocal Microphone Shield. But in general, it's about the room sound, and microphone placement."

In addition to the music, the vibe, and the acoustics, there is, of course, the gear. Harrington's Atlanta studio, where he has not been for more than two years, has older gear like a Digidesign Digi 002 and a Mac Mini. As he's been on the road with Wave for these two years, all his focus has been on the portable studio. "Lately I have been taking the most recent MacBook Pro with me, with Pro Tools, and a Universal Audio Apollo 8 soundcard, in some situations an Apollo Solo. My mic pre is the Avalon 737, and I use two microphones, the Telefunken ELA M 251 and a Neumann U87. My monitors are my headphones. I also bring Audio-Technica and Audeze headphones, but my favourite headphones are the Dre Beat Studios 3. I have some Avantone CLA-10 monitors, but I don't take them with me any more. Instead I travel with the JBL 104-BT's.

"The Telefunken is my favourite mic for vocals. It gives me air that is unmatched. The Sony C800 also sounds very crisp, and I love it, but it is not the best microphone to use when you are travelling and have to set it up in a hotel room. Nevertheless, we recorded 'Time »

The Pro Tools Mix Window from the final version of 'Street Runner'. With the beat already mixed to a stereo track (far left), all of the visible tracks are either vocal tracks or auxiliary tracks.

Heals', the last thing we released, on a C800, in Miami. I do not compress or EQ when I record, other than maybe add some high gain with the Avalon, to give the vocal a little more bite. Again, I prefer to add any treatments later on, during the mix."

Big On Vocals

After the 6am phone call, or whenever it is, Wave and Harrington will sit down in the engineer's hotel room and either record an idea that Wave has developed, or pull up a beat. "Rod writes in several different ways. There are songs for which he had written the song and we then had to find a beat to go with it. He writes a lot of things in his head throughout the day. The other way of working is that he finds a beat online, and starts to work with that. It's just him and I, and we'll go through a bunch of beats, and then it's 'Let's try this,' or 'Let's try that.' Usually something will touch him, and then he will be able to move from there.

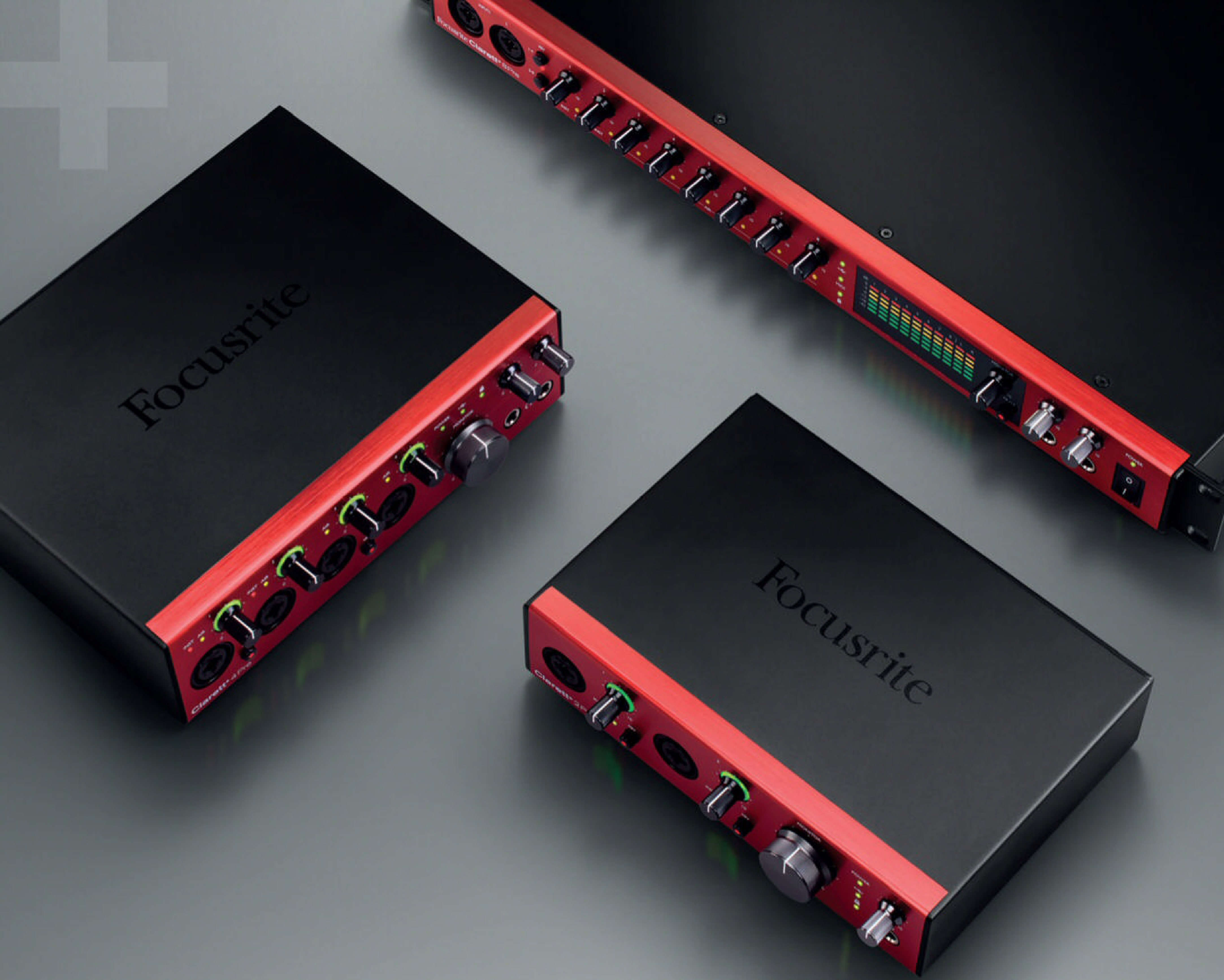
"When we were first working together, he would literally find beats on YouTube. Every now and then he still finds something on YouTube. Sometimes we'd go into the studio with some producers and figure out the sound Rod wants. I got into the game making beats, and I learnt engineering from there. That means that I can also step into the producer's shoes when needed. The majority of the time what I do with Rod is help with the arrangement of the records, and vocal production. At the end of 'Street Runner', where you have the woman talking, that was not in the original version. I added that. I do a lot of production like that.

"But most of my work is about the vocals. Everything in hit music is about the vocals. The people who taught me were big on the vocal sound. So that is where my magic lies. When recording Rod's vocals, it's all about feeling. He's not a trained vocalist, nor am I, so I don't give technical instructions as to how to improve his vocal parts. But it works out because we're after the feeling, rather than the perfect notes. We prefer lines that feel good over lines that are note-perfect. He tends to sing a song all the way through a few times, and then we go over it line by line, section by section, punching in where necessary, always focusing on feel.

"I don't like doing vocal comping afterwards, because I have nightmares that I may not have the perfect take, and once the artist is gone, I can no longer get a better take. So I prefer to have the vocal recordings finished by the time the artist walks out of the room, with the exact takes where I want them in the session. Vocal comping is kind of reconstructing a vocal, and I don't want to do that. So once Rod and I are done punching in, the vocal recording is finished. I don't need to go back and replace certain parts. I'll have everything ready to start the mix."

On The Beats

Talk of the mix leads us straight back to the fact that Harrington mixes and masters all Rod Wave's releases on his Dre Beats Studio 3 headphones. Doing final mixes and mastering purely on headphones is usually regarded as a big no-no, and the Beats headphones in particular don't have a great reputation for accuracy. For Harrington, however, thy are a reliable and indispensable benchmark, and mixing on headphones has become a way of life. "Yes, it is just how the cookie crumbles. I have learnt to mix in studios, and have been working in



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studios for a long time. When we went on the road I had to adapt, because I often don't have a studio available to me to mix, yet the releases still need to be delivered. Because I have spent so much time in studios, I was able to train my ears and understand what was going on.

"So I learned to mix and master on headphones and my Beats are my go-to. I can't work without them at this point. I finalise my work on them and don't go to a studio afterwards, as crazy as it sounds. Many people don't like the Beats, but for me they are perfect. They give me the best sound. I have been using them since the day they came out. I have done all three albums on these headphones!

"I can mix anywhere. I have worked on certain records on the bus, which is possible with noise cancellation. But it's still difficult, so most of the time I am mixing in a hotel room early in the morning before people get up. I just spend time with the record, constantly revisiting, checking, testing, and listening.

"I mix as I go. So four or five months down the road when it is time deliver the album, I have all the mixes done. I am not scrambling to go back and mix things later on. I also try to not take too much time in between the recording and mixing because of the vibe. I want to stay in the same vibe, and I want to still remember certain ideas I may have had while we were recording about what I was going to do in post-production. I also do the mastering, in my Pro Tools mix session. I am not a trained mastering engineer, it came down to how Rod's career blew up as we were going. I just needed to make it louder, and louder!"

Travis Harrington

Travis Harrington was born in 1988 and grew up in Atlanta, where he made beats in his teenager years, starting out in his mum's bedroom. In his late teens he went to the Los Angeles Recording School, after which he interned at Larrabee Studios, home of top mixers Jaycen Joshua and Manny Marroquin. After returning to Atlanta, Harrington became an assistant at Triangle Sound Studios, the facility of legendary producers Tricky Stewart and Terius 'The-Dream' Nash, which is part of their company RedZone Entertainment. Harrington worked there with Josh Gudwin and America's number one vocal producer Kuk Harrell, amassing credits on recordings by Katy Perry, Drake and Justin Bieber.

"I was taught," notes Harrington, "by Tricky Stewart and the mix engineers that he used, Jaycen Joshua and Manny Marroquin. Kuk also had a certain process as a vocal producer that I was able to be a part of. When I came to RedZone, they were on a great run, and getting all these Grammy Awards, like for Beyoncé's 'Single Ladies'. They did so many great records, and I witnessed their process. That helped me to find my own sound and approach."

Harrington went independent in 2011, and opened his own Mixing Is Art facility in Atlanta in 2015. The name, he says, is his brand, and encapsulates his belief that mixing is not purely a technical process, and that achieving perfection, technical or otherwise, is not a crucial objective. In this respect, endless polishing also is not his thing. "When you like music, you don't necessarily go, 'Oh, there's an imperfection.' Instead, there's something that touches you. I always try to stay in that realm, whether working in a hotel room or a studio. There are times when you can work on a song for two weeks straight, and at some point you can take the love out of it, you can take the art out of it. So it has to be a happy medium between the art and the technique. What you do needs to be somewhere in the middle. Much as I like to be perfectionist, we can't neglect that it simply comes down to: 'Does it sound good, or not?'"

Doing A Runner

Harrington illustrates his approach to mixing and mastering with the song 'Street Runner', which is the lead single of Wave's *Soulfly* album, and has already gone platinum in the US. The song relies heavily on a sample of the song 'Mixed Signals' by Canadian singer Ruth B.

"Rod and I found the Ruth B sample," recalls Harrington, "and we sent it to the producers LondnBlue and TnTXD, and they put a beat around it. As I mentioned earlier, I then added the final section with the girl talking. I first mixed the beat. Sometimes I get the track-out of the two beat, sometimes it is difficult to get that. When I do get the track-out, I tend to mix the beat in a separate session, as I did with this track. Mixing beats is not difficult. The vocals are more important."

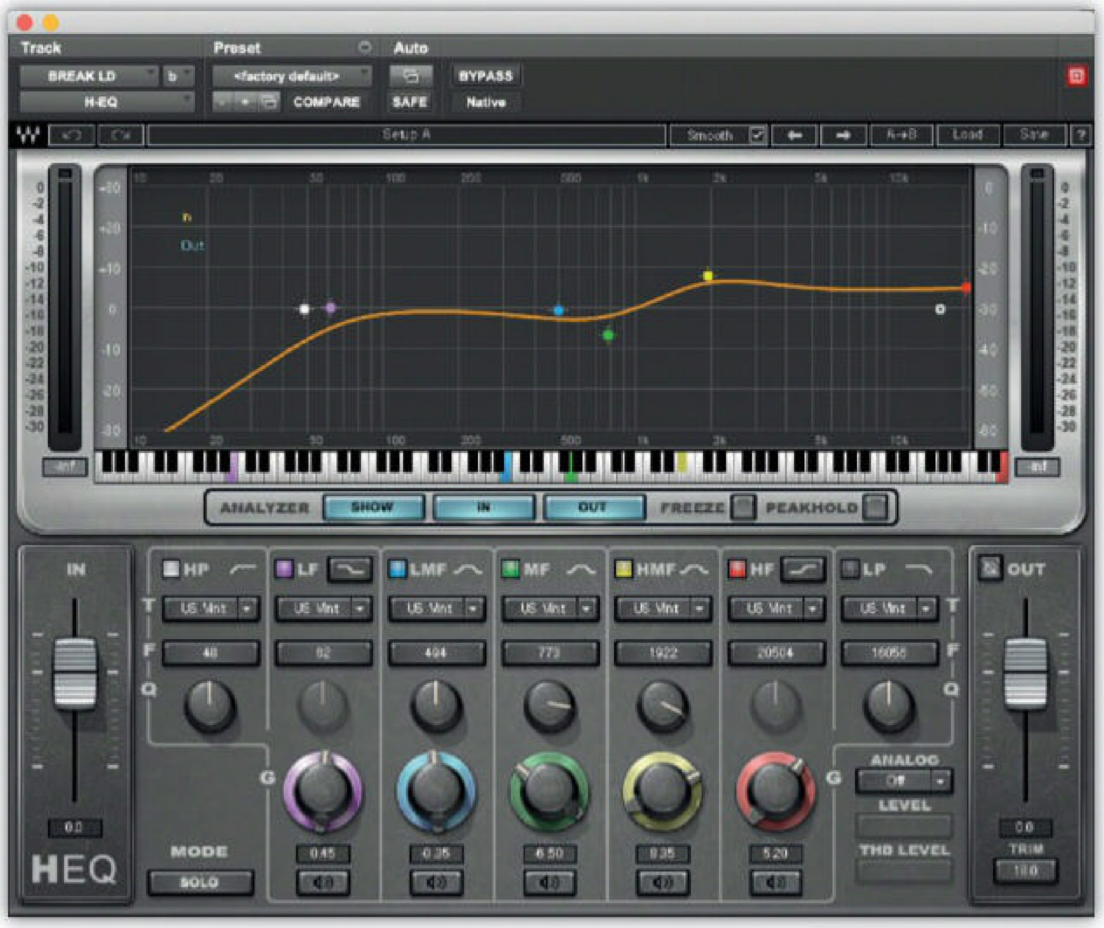
For this reason, Harrington supplied Pro Tools screenshots of his vocal mix session for 'Street Runner'. At the top is his mixdown of the sample and the beat. From there the session is structured with Wave's verse vocals ('VS'), the track with the talking girl ('Phone'), Wave's break vocals, hook lead vocals, ad libs and backgrounds. Each of Wave's vocal sections has its own aux track, and all vocals go to a 'Vox Aux'. At the bottom of the session are Harrington's aux effect tracks, with various synchronised delays, 'Big Verb', another delay track called 'Aux 1', and finally his Master Fader.

The 'Street Runner' mix session is notable for the large amount of plug-ins and sends on most vocal tracks, with up to nine inserts and four sends. The vocal signal chains are very similar, consisting for the most part of Antares Auto-Tune EFX followed by Waves H-EQ, Sibilance, RCompressor and RDeEsser, Avid D-Verb, and Waves CLA-2A and RVox. The sends go to three of the delay aux tracks and the 'Big Verb' aux. There's some variation with a Waves Doubler and an additional Waves De-Esser.

"Rod's style does not rely on Auto-Tune that much," explains Harrington. "Some of his sound comes from a natural rawness of not quite hitting some notes, and I don't necessarily look to correct that with Auto-Tune. I often keep it loose. I also use Melodyne sometimes, but it's only for one or two notes, and never set on the entire performance. The second plug-in, the H-EQ, is one of my favourite EQs, and I often boost Rod's vocal around 2kHz and above 10kHz for some clarity and bite, and I take out around 500Hz, because that's where it gets muddy.

"The Sibilance takes out some esses, and I then squash the vocal pretty hard with the RCompressor, and then there's a De-Essser. Some engineers take issue with placing plug-ins in this order, but it works for me. I'm not into rules when it comes to mixing, as you can tell from the name of my company. I don't think there are rules in art. You need to learn them, and then you can break them.

"Another difference in my mixes is my use of the D-Verb on the audio track, instead of using it on a send. I like to coat the vocal in reverb. After this I compress again, with the CLA-2A, which again is against the rules. What this does in combination with the reverb is brings the vocal out in front of the reverb, yet it also brings the reverb with it. It gives me this airy, reverb-coated vocal, that is still up front, and does not get lost in the reverb. It is kind of a weird trick.



Travis Harringon's extensive use of Waves plug-ins seems apt for an artist called Rod Wave!

"The sends have my standard delays, which are part of my template, using plug-ins like the Avid ModDelay and Waves H-Delay, and then there's the Big Verb aux, which has the H-Delay, Studio Reverb and D-Verb. I don't know who made Studio Reverb. I found it when I was digging online, it may have been free. I love it. I've set it to 'Large Theater', which is a really damp reverb. I'm an excessive reverb user. I do things with reverb that many people don't even try. For me reverb is the be-all and end-all when it comes to vocals and adding depth. Reverbs can also be used to add volume.

"Of course I tailor the settings on this effect chain for each vocal. There's also a Waves Doubler on the ad libs, because I wanted to spread them out wider. I also took out the 'Big Verb' on some tracks, so these vocals are a bit dryer and more in your face. Another interesting thing I did in this mix was to add a filter to the girl phone voice at the end, via the aux track 'End FX Aux'. I added the Waves OneKnob Filter, and the RCompressor and D-Verb, and then automated the OneKnob to get some movement going in the sound.

"The master fader is where I mastered this track. I added the Waves Kramer Master Tape plug-in for some saturation, and then some control from the Waves L2, and then the L3 and the McDSP ML4000 mastering limiter. I usually use the ML4000 to get the loudness from my master. It's the loudest mastering plug-in that I could find!

"The focus of this mix, in addition to getting the vocal right, was to make sure that all the elements, the sample, the vocals, the beat, and the girl talking, are all working together, and yet also have some contrast between them. Using samples is just the name of the game today. You get everything from many different sources all the time, it's part of the process. The main thing is something Dave Pensado once told me, which is about making everything you do sound musical."

Vienna Symphonic Library Synchron Brass

Sample Library

VSL's new hall-recorded brass collection aims for maximum grandeur.

DAVE STEWART

After you've spent 21 years deep-sampling every orchestral instrument under the sun and building an archive of over six million samples, what do you do next? If you're VSL, the answer is simple: find a new recording location and start the whole process over again. As EastWest and Spitfire Audio have done before them, the Viennese busy bees are creating a second complete symphonic collection to place alongside their previous monumental body of work. The project is well under way: following the release of Synchron Percussion and Synchron Strings, the latest addition is VSL Synchron Brass, featuring 136.8GB of instrument recordings performed by a team of first-call players.

Comprising eight solo instruments and nine ensembles, this large library covers all the orchestral brass essentials, ranging from solo trumpet to a layered 'giant tutti brass' section of 28 players. Other outsize groups include 12 French horns, six trumpets and a section of nine tenor and bass trombones, with smaller ensembles and a choice of solo instruments available for less grandiose settings. While specialist items such as piccolo and bass trumpet, flugelhorn, cornet, alto trombone, euphonium and Wagner tuba aren't included, the popular cimbasso gets a welcome airing — and should you need any of those abovementioned non-standard instruments, all are available as individual downloads in Vienna Instruments format.

In stark contrast to the studio acoustic of VSL's stereo-only Vienna Instruments collections, the Synchron series is recorded from multiple microphone positions in the historic Synchron sound stage, and therefore enjoys the twin advantages of a sumptuous concert hall ambience and multi-channel mixing. The miking setup accommodates mix formats of up to 9.1 surround and immersive audio formats such as Auro 3D and Dolby Atmos — but don't worry if you can't cram 10 speakers into your workspace, these samples still sound great in stereo! Like all Synchron titles, VSL Synchron Brass runs exclusively on the company's free Synchron Player and requires a ViennaKey USB protection device, yours for the princely sum of €13.

Although most of the library is newly recorded, some ensembles are taken from the VSL Big Bang Orchestra sound packs Hercules, Izar, Jupiter, Kopernikus and Zodiac, with no additions, omissions or tweaks. Crossgrade discounts towards the Standard and Full libraries of Synchron Brass are available for registered owners of those BBO packs — stick with me, and I'll explain the musical details as we work through the instruments.

Trumpets

Arguably the most traditionalist of the major orchestral sample developers, VSL have loosened their bow ties and edged

The Synchron Player contains all of an instrument's available articulations, with colour-coded keyswitches enabling you to switch artics in real time.

away from classical formality in recent years. Coincidentally or otherwise, Synchron Brass' lead trumpet player Marc Osterer has worked with major pop/rock artists (including a distinctly non-classical electro-swing act) and Grammy nominated jazz big bands, as well as occupying the principal trumpet seat in the Mexico City Philharmonic and Synchron Stage Orchestra.

Mr Osterer's musical experience comes to bear on the accomplished set of Solo Trumpet 1 articulations. The combination of a beautiful tone and VSL's trailblazing legato mode produces magnificent melodic results — expression is enhanced by the gentle, lilting vibrato used in the long notes, a pop-friendly alternative to the standard straight orchestral delivery (which he also plays supremely well). A cool 'auto-speed' feature automatically switches between regular and fast legato transitions according to your playing speed, resulting in stunningly realistic lead lines. I also enjoyed the laser-beam precision of the player's fast note repetitions, played in a choice of four tempos.

Solo Trumpet 2's samples were performed by Peter First, another first-class musician who contributes a tasteful, quiet and reflective expressive legato style along with excellent short staccatos which are ideal for fast rhythmic ostinatos. In a similar vein are 'upbeats' consisting of a single short note preceded by one, two or three fast repeated notes of the same pitch, great material for creating fanfare-like figures. Both players also perform scorching renditions of the blasting, in-your-face sforzatissimo style, traditionally the most forceful and aggressive articulation in VSL's locker.

Moving on to the ensembles, six unison trumpets display immaculate tuning on their long notes. If you layer their 'con fortissimo' loud sustains with the strong attack of the abovementioned sffz delivery, the result is both regal and explosive: the perfect soaring high brass timbre for a superhero movie theme. The tighter sound of four Bb trumpets (taken from BBO Kopernikus) lends itself well to staccato rhythm passages, and their marcato long notes (which utilise an overlaid staccato attack) and imperious sforzandos sound good and strong.

French Horns

Flown in from BBO Jupiter after a brief refreshment stop at the International Space Station, a six-piece horn section is joined by 12 horn players from BBO Zodiac's 'Supermassive Ensembles'. This cosmic line-up makes a glorious collective noise: the Zodiac horns' intonation is uncannily precise across all articulations, which include lovely warm, plush quiet sustains, stentorian fortissimos, wonderfully dramatic crescendos and maniacally loud sffz fanfare-friendly deliveries fit to greet the entrance of Thor into Valhalla. If you want grandeur, these are the guys. The six horns' sustains are a great asset for chord pads, and their smooth note attacks work well for melody lines, even without the added smoothing of the included legato transitions. My *SOS* review of these BBO Jupiter horns mentions their creamy legatos for triumphal heroic themes, brusque short notes for rhythmic ostinatos, in-your-face blaring sforzatissimos, luxurious soft swells and jeering flutter tongues. I also praised their terrific set of octave rips, and can now add that the 12 horns' rips are even more uproarious, bringing to mind the bellow of an enraged bull elephant.

Synchron Brass' two solo horns are played respectively by Péter Keserű and Viliam Vojčík. Hailing from a Hungarian city whose name I daren't type for fear of overheating my spellchecker, the first gentleman was principal horn player with several European orchestras before taking up the post with the Synchron Stage team. His colleague also occupies »

Vienna Symphonic Library Synchron Brass

From €435

PROS

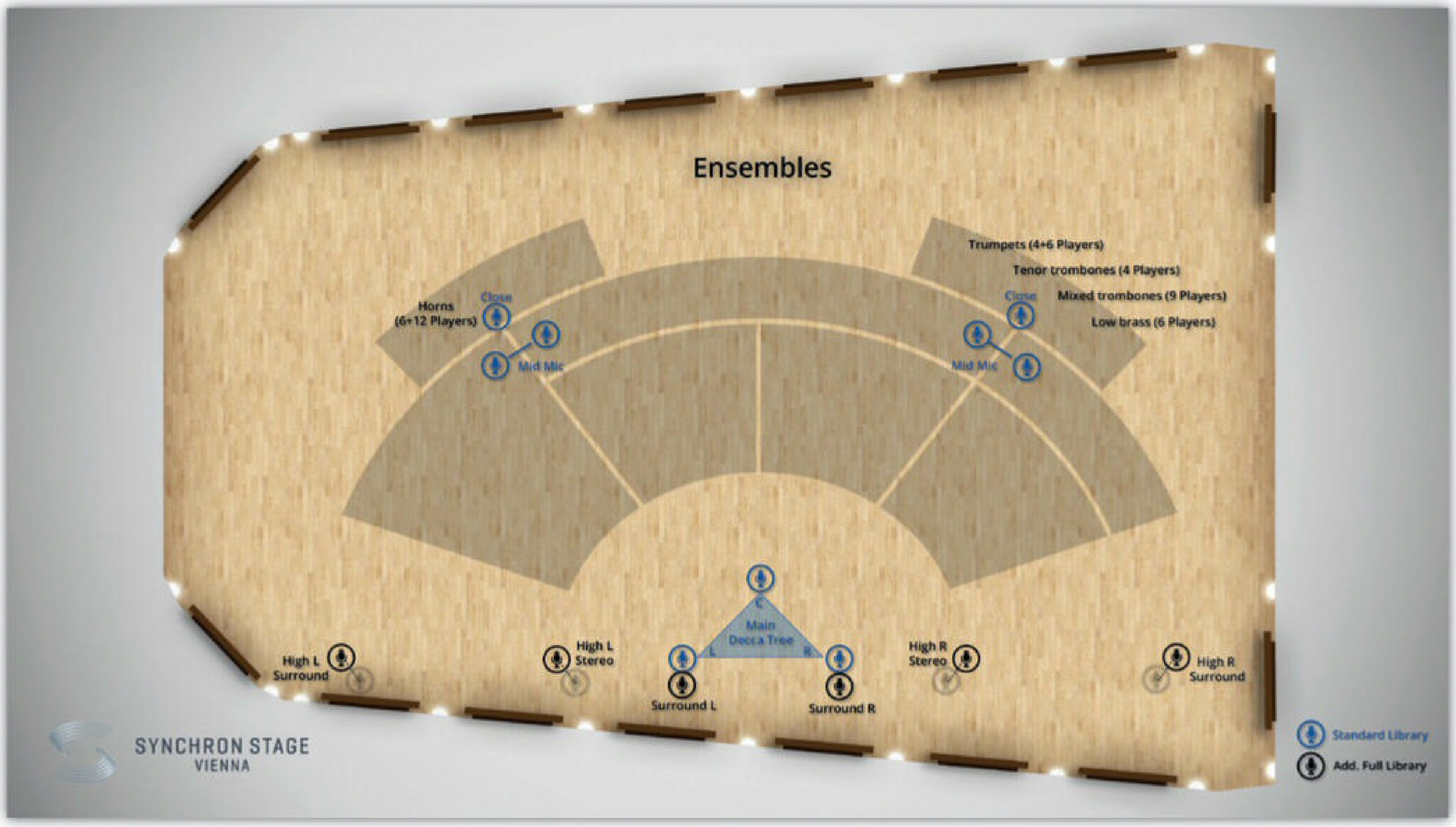
- A select team of top European players perform a comprehensive set of articulations.
- Recorded from nine mic positions in a top-class historic sound stage.
- A generous instrumentation includes solo instruments and ensembles of between four and 12 players.
- The 28-player 'Giant Tutti Brass' preset is to die for.

CONS

- It's a bit expensive, but quality doesn't come cheap — and you don't have to buy the extra mic positions!

SUMMARY

Recorded from multiple mic positions in a sumptuous hall acoustic, this large library offers everything from solo instruments to extra-large ensembles. The playing is super-precise, the articulation menu is comprehensive and the dynamic range is stupendous, covering the spectrum from soft, quiet long notes to blasting, aggressive sforzatissimos. If you're thinking of investing in an orchestral brass collection, I recommend you give VSL Synchron Brass serious consideration.



The nine-channel miking scheme used to record the Synchron Brass ensembles.

a horn hot seat, this time with the Slovak Philharmonic Orchestra.

French horn is a difficult instrument to master, but both players achieve admirable pitch accuracy over their instrument's C2-F5 range and maintain it across an identical menu of articulations. Player number one turns in an exemplary set of legato long notes, while his co-worker whips out tight, short staccato 'agile' performances. Both musicians perform lovely, mournful espressivo legato notes with a subtle built-in swell and, by way of contrast, their own personal set of stirring fierce rips.

Trombones

The library's solo tenor and bass trombones are played respectively by Matthias Reindl and Bernhard Vierbach. The tenor trombone's precise 'agile' staccatos and impressively accurate note repetitions work well for motoring rhythms, while the bass instrument's loud staccatos are the classic oom-pah sonority heard in marching band bass lines. Dynamics range from quiet sustains (a great asset for supportive chord pads in orchestral or brass band arrangements) to super-raspy sffz notes, which can sound sinister or funny depending on the context. The icing on the cake is an excellent set of crescendos and diminuendos in a choice of two, three and four seconds durations.

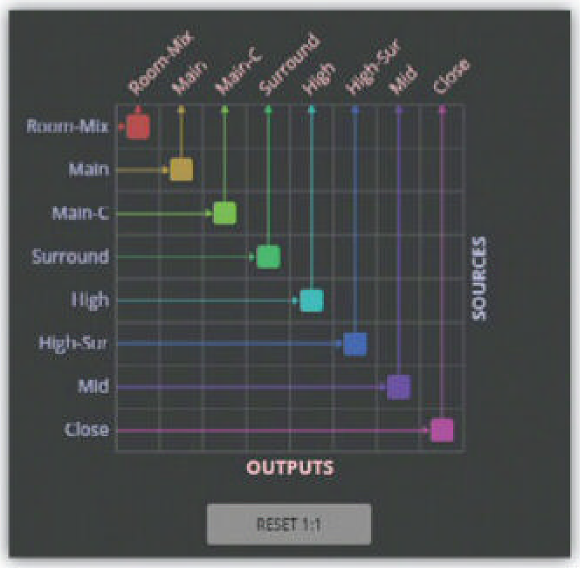
Two excellent trombone ensembles add to the symphonic splendour: four tenor trombones make a pleasantly broad, noble sound, with loud, bright, expansive sffz performances, powerful crescendos (which include a subtle softer option) and some tremendous glissandos. Played at two dynamics over a choice of minor second, major second and fourth intervals, these slow, measured pitch slides end conveniently on a looped sustain. User tip: playing chords with the descending fourths version sounds pretty amazing.

If you thought trombones weren't the most exciting of brass instruments, think again — a contingent of nine unison tenor and bass trombones (one of BBO Zodiac's large ensembles) are superbly sonorous, generating great gusts of raw brass power over a D1-G4 range. Their sffz, ff and four-second crescendos are astonishingly forceful and will kick any arrangement in the proverbial backside, or you can go the other extreme and program quiet chordal passages with their excellent soft sustains, to which the enveloping hall acoustic adds warmth and size.

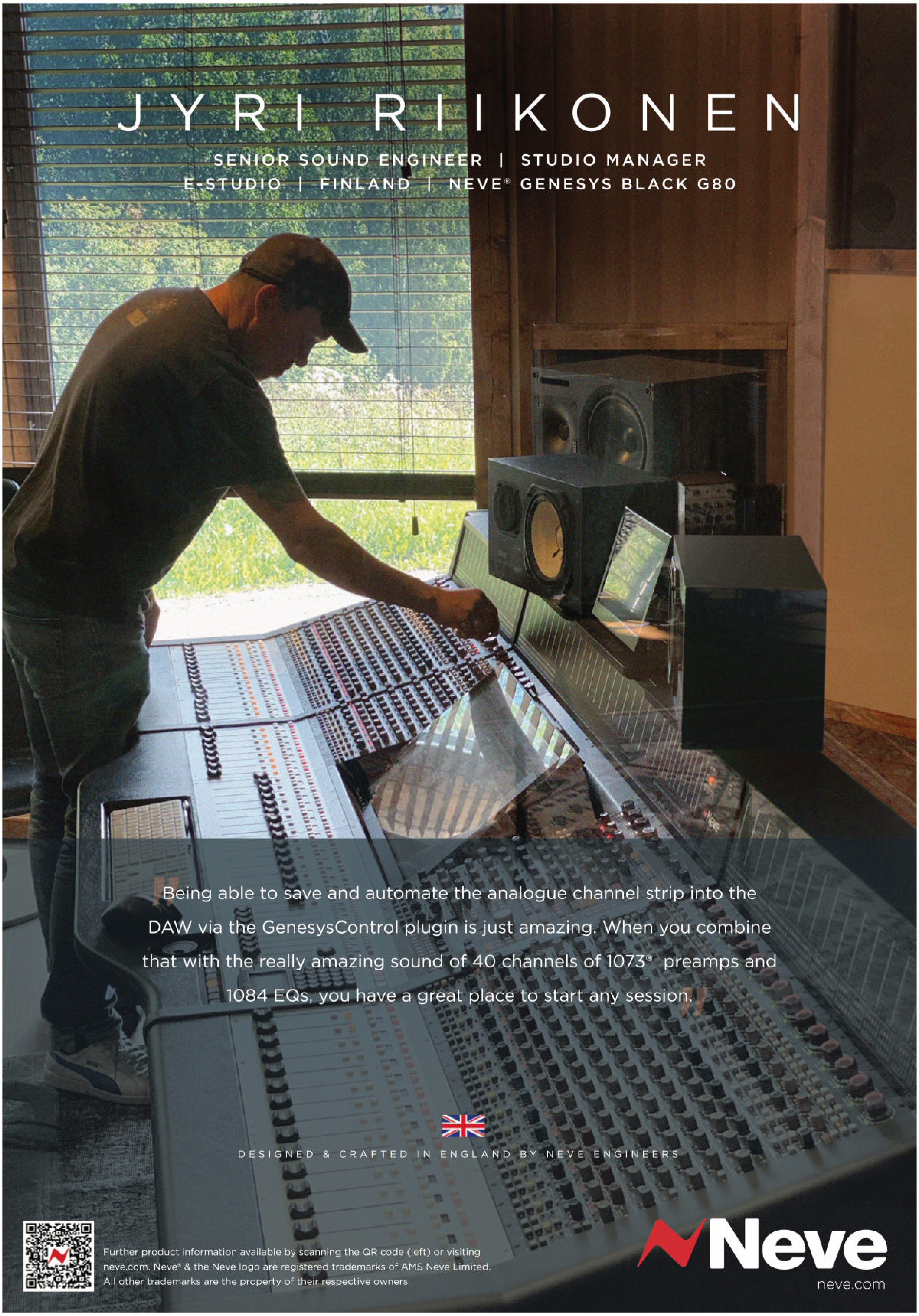
Cimbasso & Bass Tuba

This pair of solo instruments operates at the low end of the brass instrument pitch spectrum. Though the cimbasso's mad cylindrical tubing and 'bent' shape won't win any design prizes, the instrument is surprisingly expressive, with player Stefan Hirt's quiet sustains providing a good option for low, brooding horn-like pads. Alternatively, you can follow Hollywood film composers' example and go for broke with loud, percussive and metallic sffz bass stabs. My advice: put a banging drum & bass loop over the cimbasso's shuddering low-register 160bpm note repetitions and you could have a surprise dancefloor hit on your hands.

With a bottom note of C1 (traditionally the lowest pitch of the orchestra), the obvious use for a bass tuba is low pedal



You can customise the Synchron Brass output configuration to suit your multichannel mixes.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The library's six-piece trumpet section.

notes propping up brass harmonies, but the instrument (played by Lukas Hanspeter) also works well for plaintive and touching legato melody lines, bringing to mind the comic waddling pathos of American actor Oliver Hardy. If you're inclined towards more ribald comedy, a few choice blasts on the bass tuba's sffz low notes are bound to raise a titter.

Low Brass & Tutti Ensembles

The library's low brass ensembles pack enormous power. On loan from BBO Hercules, a combo of four bass trombones and bass tuba are bolstered by a contrabass tuba in the lower octave. The sound is truly monstrous, evoking the thunderous solemnity of pipe organ pedals and providing a massive bass foundation for grandiose cues. As mentioned in my *SOS* October 2020 review, an optional fortissimo layer blasts out the popular explosive, end-of-days cinematic brass sound while ominous low-register glissando slides generate instant drama. As well as a colossal bass presence, the section sounds good on soft midrange chords, with horn-like swells creating a warm pad timbre.

BBO Hercules' evil twin Izar also rears its head in Synchron Brass, duplicating its sibling's articulations in the form of dissonant three-semitone clusters calculated to shred the nerves. The contrabass tuba takes a rain check and there are (mercifully) no legatos, but as I remarked back in 2020, the clusters' mad helicopter-like fast repetitions sound great in ostinato action passages, and descending cluster glissandi combine a mournful quality with a hint of psychosis. The 'Reg. XF Clusters' patch lets you crossfade between straight notes and clusters — if you set the pitch-bend to 12 semitones and perform crossfades while twirling the wheel, it sounds like nothing on Earth. No idea who Reg is, but congrats to him for this utterly mad sound.

Synchron Brass' instrumentation concludes with a formidable 'giant tutti brass' super-ensemble which layers the aforementioned six trumpets, 12 horns, nine trombones and bass tuba into a gargantuan 28-player confection (you can also play the sections individually). Two-handed keyboard players will love it — mapped over a five-octave D1-D6 range, this preset includes staccatos, long notes, legatos, sfz, sffz and strong crescendo two and four-second articulations. The latter style lives up to the 'super-massive' tag, a hair-raising dynamic surge which erupts in great gold-plated sheets of sound. Absolutely tremendous, one of the best full brass patches I've played.

Instrumentation

Trumpets
- Solo trumpet 1
- Solo trumpet 2
- Four trumpets
- Six trumpets

French Horns
- French horn 1
- French horn 2
- Six French horns
- Twelve French horns

Trombones
- Solo tenor trombone
- Solo bass trombone
- Four tenor trombones
- Nine tenor & bass trombones

Low Brass
- Cimbasso
- Bass tuba

Low Brass Ensembles
- Four bass trombones, bass tuba
- Four bass trombones, bass tuba, contrabass tuba

Tutti Brass Ensemble
- Six trumpets, 12 horns, nine tenor & bass trombones, bass tuba (layered)

General Points

As you may have deduced, Synchron Brass contains no phrases or licks other than


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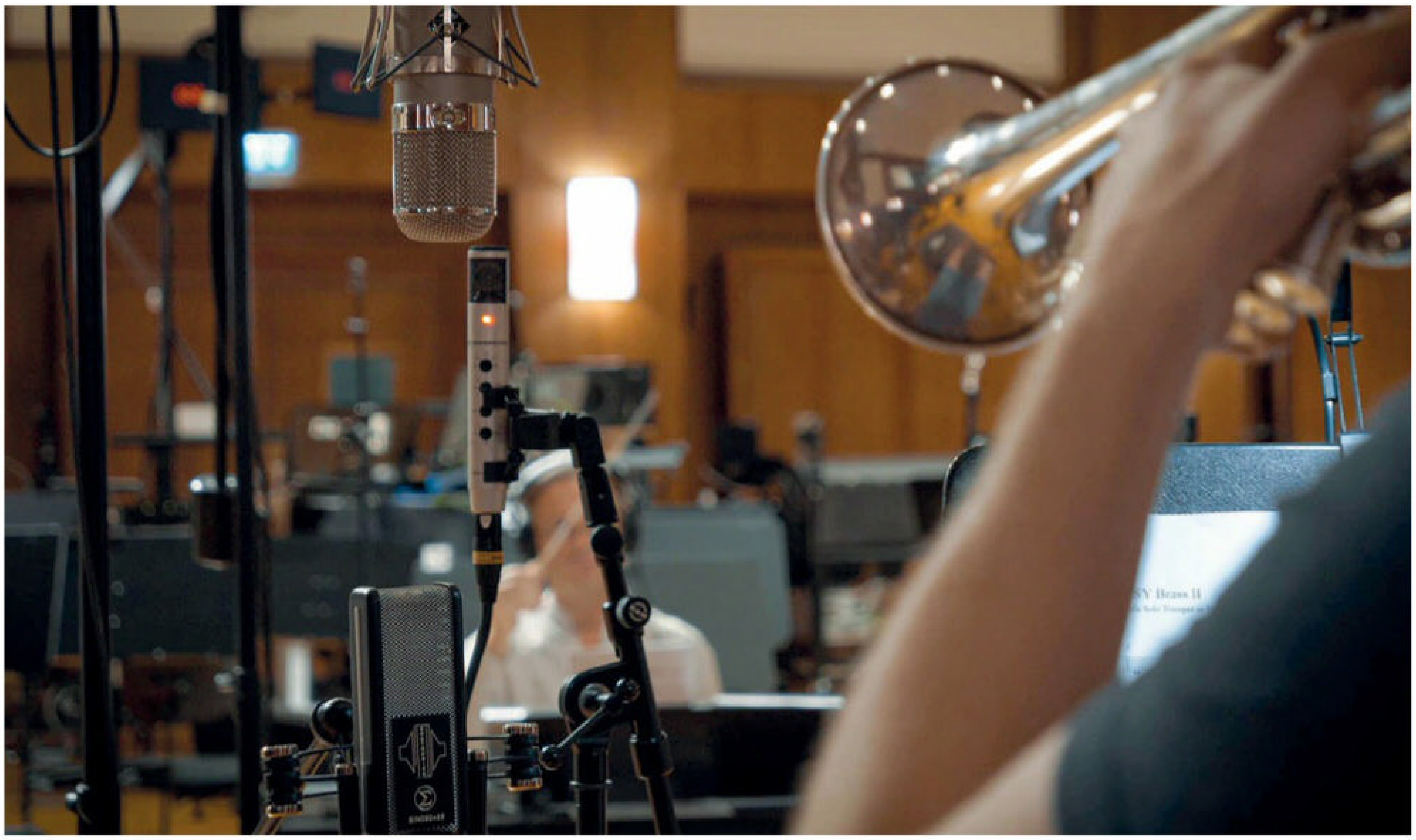
Microphone Positions

VSL's Synchron-range Standard and Full Libraries include the same articulations, the only difference is the number of mic positions: the lower-cost Standard Library contains stereo Decca Tree Main/Room mics, a stereo Mid position, a mono close mic and the Decca Tree mono centre mic. The Full Library adds five extra positions: Main Surround, High Stereo (3D) and High Surround (3D) (all stereo), plus a stereo Solo Horn Room ribbon mic and Solo Horn mono Back mic.

The makers advise that as well as serving as a rear speaker source in a 5.1 mix, the Main Surround position may be substituted for the stereo Main/Room room mics in order to achieve a wider and more ambient mix. To facilitate mixing, the library includes a large number of pre-configured mixer presets created by VSL technical director Bernd Mazagg which provide a wide variety of tonal characteristics, including heavily processed mixes which utilise the Synchron Player's built-in channel effects such as saturation, compression and EQ.



» note repetitions. There are no mutes, and no jazz styles such as shakes or falls — the vibe is decidedly orchestral, but that's not to say you couldn't use these excellent trumpets and trombones in a pop brass arrangement. It's worth noting that the note repetitions' four fixed tempos can be non-destructively stretched in real time to a tempo which fits your arrangement, a very handy facility.

The browser for each instrument lists three preset types, marked 'VelXF sus', 'VelXF' and 'Velocity'. In the first, velocity crossfading is activated for long notes and can be controlled with MIDI controller CC1 via the mod wheel, while the dynamics of short notes are controlled by keystroke velocity. In the second, all articulations have velocity crossfading activated for MIDI CC1, while the third preset type reacts only to keystroke velocity.

For this library VSL have introduced a new 'Timbre Adjust' feature as an alternative or supplement to velocity crossfade. This intelligent filter lets you apply dynamic timbral changes while staying in one velocity layer, so you can (for example) soften the decay of a long note without switching to a new layer. It's also useful for adding expression to single-velocity layer articulations such as sforzatissimo, or to bypass the velocity crossfade artefacts that can sometimes be heard in exposed solo instruments.

Conclusion

As VSL painstakingly chip away at their second major symphonic project, one wonders where all this might end. It's logical to assume Synchron Woodwinds will emerge from its Austrian birthplace at some point, but in the wider world of orchestral sampling it's anyone's guess where the ongoing proliferation of libraries will fetch up. If the crystal ball looks a little cloudy, it's clear that for the time being users are continuing to buy, enjoy and make great creative use of these collections, and long may that continue.

Regardless of what the future may hold, if orchestral arranging is your thing you'll want to get your hands on the best sample collections on offer, and VSL Synchron Brass is definitely one of them. Created by an industry heavy hitter which has been raising the bar for orchestral sampling for over 20 years, this superior product features some of Europe's finest players performing an extensive range of articulations with pinpoint accuracy in a hall described by Hans Zimmer as "a killer stage to record brass instruments". The net result is a superbly realistic, versatile and wonderfully dynamic orchestral brass library which will add grace, grandeur, expression and sheer power to your scores.

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Way Huge Atreides

Analogue Effects Processor

Described by its creator as a "weirding module", the Atreides doesn't do things by halves!

WILLIAM STOKES

With his name printed on their products, Jeorge Tripps clearly wants you to know that Way Huge pedals are his designs. And why wouldn't he? He's been responsible for pedals as iconic as the Line 6 DL4, and his influence at Dunlop, through the Way Huge brand, has been felt for almost 15 years.

His latest creation, the Atreides "analogue weirding module", is somewhere between a synthesizer pedal and a multi-effects unit. Taking not a little inspiration from Electro-Harmonix's 1980 Mini-Synthesizer, its name is a reference to the cult '80s sci-fi classic *Dune*, a film befitting the invitation in this pedal's manual to "ascend the high dune and gaze upon an expanding world of musical delights full of pure imagination". In more down-to-earth terms, it blends an envelope follower, a fuzz circuit, a phaser and a sub-octave generator, and it's capable of fashioning sounds that range from deep, synth-fuzz beef, to phasey, wah-infused leads.

Glitchy, Mangled Charm

A handful of characteristics ensure that the Atreides lives up to its "weird" strapline, for better and for worse, and it's definitely intended more for those moments of wildness and flourishes of crazy than being left on for your entire set.

It comes into its own when all the sliders are positioned just right to balance their respective contributions to its wild character: when you get this spot on, it has attitude, it's off-kilter and it's fun. Beyond this sweet spot, though, there isn't a huge amount of room to customise your sounds, and the overall message is clear: it's the Atreides way or the highway!

For one thing, it doesn't have a wet/dry mix control, which would have been so useful. And beyond a certain range, it doesn't offer masses of control over the sound. The leftmost Volume slider has zero headroom, so goes straight from 'off' to heavily distorted; and while its slider is incredibly sensitive up to about halfway, it's not really sensitive enough after that point. Neither does the Fuzz slider offer any real gain control.

To my mind, it was an interesting decision to include fuzz instead of a more conventional gain control, considering how much fuzz and synth pedals

already have in common. They both add thickness, distortion and subharmonics to your sound, and favour simple harmonic content. This might help to explain why, at high levels, the Fuzz slider seems simply to add brittleness more than anything else. Thankfully, this can be compensated for in part with the addition of a sub-octave.

The closest thing to an actual tone control on the Atreides is its Brightness slider, which offers a nice EQ range, particularly given the way it compounds the effect of the filter envelope. Dial it up for a quasi-digital distortion sound (think Queens Of The Stone Age with Mark Ronson). Or dial it down a little, crank up the fuzz and envelope sensitivity, and you'll achieve retro-futuristic psychedelic fuzz.

The envelope follower is excellent and, frankly, can cover a multitude of sins. Snappy, wide and articulate, its dynamic response feels very natural, whatever the setting, and its filter resonance is tuned nicely: not too harsh, not too murky. Gold star. At its highest sensitivity setting, which, oddly, is at the bottom end of the slider, and with a wide range, digging in even slightly will throw the filter wide open with a nice quack. It's great for lead lines and show-off solos, and with both parameters set lower it adds a subtle tone shift that reacts pleasantly to your playing.

To be honest, I've always been a bit suspicious of envelope followers, and have even found them a little cheesy. But in this most synthy environment this one works very well indeed, adding real character and some crucial room for expression. It also means the Atreides' impressive sustain is usually followed by a gritty, stilted shutting of the envelope,

The input and output are mono, on the usual TS quarter-inch jacks.

which at best sounds like a cool, crackly, almost ripped-speaker effect. (At worst it resembles a loose connection of some sort, or a broken tube.)

This segues into the potential frustration that, even with their respective sliders on zero, none of the Atreides' individual effects can be bypassed, except for the sub-octave (and I infer that this is likely not actually bypassed but rather too quiet to hear). Don't get me wrong: it's fun to be hit with a wave of thick sub, swooshing phase and envelope filtering when mixing the Atreides' different circuits and playing them off against one another. But I dare say it would be more fun — and more versatile — if you were able to bring in different functions incrementally and in different combinations. It also means that you can never escape the residual sounds of the envelope being triggered by minor bumps against the body of your guitar, for example, or by muted string movements registering on the pickups.

"It's definitely intended for moments of wildness and flourishes of crazy."

If there's one function I could have done without on this pedal it's the phaser. It definitely has a sweet spot, at around three-quarters up, and I really enjoyed the sound that resulted from pairing it at this rate with a relatively sensitive, wide-ranging envelope. But any higher than this and it feeds back, and to my ears unattractively so. Using an internal trim pot, you can attenuate the phaser feedback, which is helpful, though I'd have much preferred this to be on the exterior and, better still, for it to be replaced by the option to adjust the phaser's depth.

At a low rate, the stages don't feel particularly smooth or deep, an issue that is possibly accentuated by its interaction with the envelope follower, and another reason why bypass options would have been useful. I almost wonder if this a case of attempting too much — although I must stress that, as with so many other aspects of the Atreides, the whole equation carries an unmistakable glitchy, mangled charm. Emblematic of this is the sub-octave generator. This interacts with the Atreides' other circuits to produce various unpredictable characteristics that aren't shared by conventional octave shifters; it hops between harmonics in curious ways. A second under-the-hood control toggles between one- and two-octave intervals, which is useful given that this pedal behaves very differently depending on which register in which you play.

Weirdly Wonderful?

If you've read this far, you'll have that the Atreides has its limitations. Many potential users would probably expect there to be much more distinction between the different functions, and more versatility overall, in a stompbox costing this much. But perhaps I'm overthinking it... All of the Atreides' functions are available individually in a litany of separate pedals, which you could set up in different configurations if that's what you want. But could you set them up to sound quite like the Atreides? I suspect not, and further, that Jeorge Tripps is very much aware of that. For this pedal it's a case of switch on, stomp and strap in for a wild journey across the galactic desert.

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Way Huge Atreides

£199

PROS
- Great fun.
- Unique sound.

CONS
- Could offer more control.
- Fuzz circuit leaves a little to be desired.

SUMMARY

A deliberately oddball stompbox that will divide opinion, the Atreides is a unique sound-mangling tool.

Bluetooth LE Audio

A New Standard In Wireless Sound

The next evolution of Bluetooth contains a number of innovations that should appeal to consumers and professionals alike. And it's coming to a device near you.

RICHARD EINHORN

When it comes to successful digital technologies, few can top Bluetooth, the wireless data and audio streaming standard that was named for some reason after an ancient Danish king with really bad dental hygiene. In the past five years, over 23 billion Bluetooth devices have been sold: a mind-boggling amount of gear, equating to roughly three devices for every human on the planet. Bluetooth audio streaming devices, such as headphones, earbuds, headsets, portable speakers and TVs, comprise some 28 percent of that total (1.3 billion units last year), much of the rest being smartphones and non-audio data transmitters like computer keyboards, mice, and FitBits.

In the wake of this spectacular success, the Bluetooth Special Interest Group (SIG) has begun publishing the draft specifications for Bluetooth LE Audio. Enough information is already available for manufacturers to get started on designs, and the full spec should be approved by early next year. This is a brand-new standard intended eventually to supersede the Bluetooth we know so well, and which is now dubbed Bluetooth Classic.

As Nick Hunn, CTO of WiFore Technology and one of the major developers of Bluetooth LE Audio puts it, the goal of Bluetooth LE Audio is to provide a "set of tools that will cope with any wireless audio application — any combination of music and voice — that anyone is likely to come up with for the next 10 to 20 years". For those of us in professional music and audio production, this is exciting news. Not only will Bluetooth LE Audio allow audiences to hear and share our work in much higher quality, but the new standard is so flexible that it may very well spark new ways to conceive of sound-centric art and entertainment.

Into The Jaws Of Bluetooth

Simply put, Bluetooth is a short-range (10 metres) wireless communication standard that operates over the 2.4MHz frequency band. There are two basic kinds of Bluetooth. What's now called Bluetooth Classic was developed for telephone and audio transmission, which were conceived as separate use cases. Bluetooth Low Energy (LE) was originally intended for transmitting small amounts of data, such as keyboard and mouse information to a computer or sensor data from health and fitness devices to your smartphone. "Bluetooth LE was never designed to transmit audio," explains Nick.

'Profiles' are Bluetooth-speak for code that defines possible applications and use cases: they specify the general ways that Bluetooth devices communicate with each other. The first audio profile that Bluetooth developed was Headset, which was soon replaced by Handsfree

to accommodate car connectivity. It was intended for wireless, single-sided headsets in call centres or cordless phones at home.

Then in 2006, Bluetooth released another audio profile, A2DP, which allowed for higher-quality audio transmission. It took 10 years, but once audio and video streaming caught on, people wanted to listen to music and watch movies everywhere. Sales of wireless earphones exploded and Bluetooth established itself as the all-but-universal wireless audio streaming standard for mobile devices. "I did a graph looking at [*the growth of*] Spotify subscribers against sales of Bluetooth headsets," Nick says, "and they track almost identically."

Today, Bluetooth Classic has become, well, a little long in the tooth. The Handsfree and A2DP profiles "don't really work together," Nick says, because "the Handsfree and A2DP profiles were designed without anybody ever really thinking that you'd want to swap between them." For example, if you're listening on your AirPods to some desert blues and a phone call comes in, classic Bluetooth has to go through considerable hoops just to stop playing the song and take the call. "It's not a clean solution," he said.

Also, Nick continues, "We have nothing [*in Classic Bluetooth*] that supports a separate left and right earbud. All the stereo [*earphone and earbud*] solutions you see on the market are all proprietary ways of trying to cope with A2DP." The Bluetooth Classic audio spec has other limitations that became apparent as Bluetooth usage spread far and wide: it uses a fairly significant amount of energy, doesn't fully support simultaneous bidirectional streaming, is rather slow and, except for proprietary solutions, permits only one device to be connected to another device at a time.

The Big Hack

Around the same time that sales of headphones using Bluetooth Classic went through the roof, Bluetooth Low Energy acquired audio streaming capability. And therein lies a tale.

Many hearing aid companies provide expensive, proprietary remote controls to wirelessly adjust the volume and program presets of hearing aids. They also sell expensive, proprietary devices that stream landline phone calls directly to hearing aids. But once everyone started using smartphones, no one wanted to lug around these extra gadgets. They wanted to use their phones to change volume and take phone calls, just like people with wireless earbuds.

There was one major problem: hearing aid batteries are extremely small, and Bluetooth Classic's energy requirements would quickly drain them. So, Apple (and eventually others) essentially hacked Bluetooth LE for, as Nick explains, "a very simple bi-directional audio stream" that provided wireless audio connectivity for hearing aids. The sound quality was limited in terms of bandwidth but it worked and met an important need. 'Made for smartphone' hearing aids were an instant hit. And that's when something remarkable happened...

Universal Design

It's rare that the needs of people with disabilities are central to the development of a technology intended for mass use. Yet meeting their needs often leads to products that benefit everyone, a concept called Universal Design. That is exactly what occurred when, according to Nick, "The hearing aid industry sat down with the Bluetooth SIG and said, 'Look, things are changing. We want to connect hearing aids [*via Bluetooth LE*], but we think there is a bigger market for this.' Sound today is being used in a whole variety of different ways that need something a lot more flexible than the monolithic approach [*of Bluetooth Classic*]."

Originally, hearing aid companies wanted "to gain better performance and battery life", recalls Chuck Sabin, Senior Director of Market Development for the Bluetooth SIG. But as discussions ensued, it quickly became clear to many audio companies in the Bluetooth SIG that a new protocol which not only ran on very low energy but also met the needs of hearing aid users would be of great interest to everyone. And so, work began on the new standard: Bluetooth LE Audio. "This was our opportunity to think about this holistically, to build a new architecture for the next generation of audio with new device types, high-quality audio at low bit rates using less power, and new location-specific audio services."

Shiny New Features

Bluetooth LE Audio extends the wireless audio capability of Bluetooth LE in several important ways. For a deep dive into the fascinating details, take a look at the 'Cracking The Codec' box. But briefly...

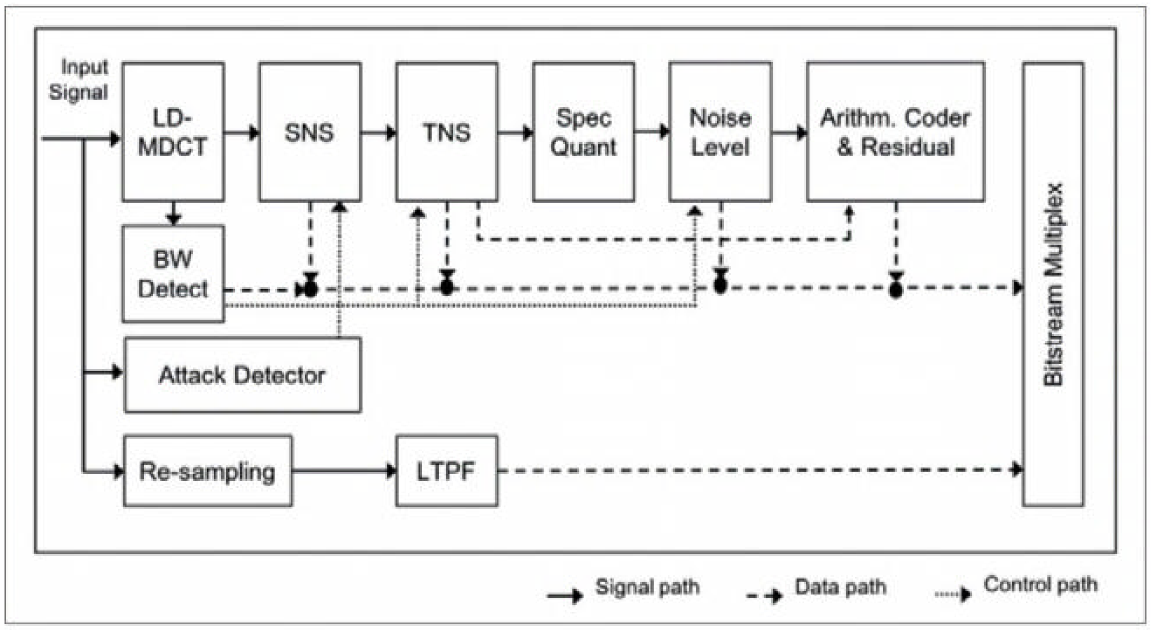
A new set of low-latency codecs was developed by Fraunhofer specifically for Bluetooth LE Audio, called LC3 (mandatory for all LE Audio applications) and LC3plus. These are not your Dad's codecs. Compared with SBC and aptX — two codecs used in Bluetooth Classic that date back as far as the '70s and '80s — LC3 requires much lower energy but can deliver audio up to 24-bit/48kHz; the high-resolution LC3plus can even go to 32/96. Extensive listening tests conducted by Fraunhofer and Bluetooth confirm the transparency of high bit-rate LC3 coding. According to Nick Hunn, LC3 "really is a major leap forward in terms of a state-of-the-art codec".

Second, Bluetooth LE Audio provides 'multi-stream' signal-synchronisation capability with latency potentially down to 25ms, both of which are considerable advances over Bluetooth Classic. LE Audio supports not just a simultaneous left and right earbud — which solves the single-stream problems inherent to A2DP — but permits an arbitrary number of synchronised audio channels (dependent upon bandwidth). In other words, with Bluetooth LE Audio, not only are multiple language translations possible in a single wireless stream, but even multichannel audio, 5.1 surround, and Dolby Atmos speaker systems can be catered for.

Third, Bluetooth LE Audio provides a suite of bi-directional audio sharing technologies. One simple example: instead of handing your friend an earbud when you want to share some music with her, you'll simply tap a button on your phone and the music will now stream to her own earbuds as well as yours. But that's just for starters. Unlike Bluetooth Classic, which allows only one device to connect at a time to another device, Bluetooth LE Audio's 'one-to-everyone' broadcast capability can be used, say, for private listening to separate TVs at sports bars or elsewhere.

In fact, the broadcast potential of LE Audio is far greater than simply short-range audio transmission. The effective broadcast capacity of LE Audio extends much further than Bluetooth Classic (as much as 30 to 50 metres, even further if multiple transmitters are deployed). This capability, combined with exceptional sound quality, will make it possible to use LE Audio not only for next-gen assistive listening at concerts »

Cracking The Codec



To learn how the LC3 and LC3plus codecs work, I got a tour from Alex Tschekalinskij, a software engineer in the Low Delay Audio Communications Group at Fraunhofer, who helped develop them.

What exactly is a codec? A lot of data gets slung through the air during wireless audio transmission, and that takes a lot of energy. To conserve battery power, developers have worked out clever algorithms that compress the audio data to a manageable length for transmission (ie. encode it) and decompress (decode) it when it is received. This 'encode/decode' process is called a codec.

A good audio codec has to accomplish two neat tricks: minimise complicated computing (which takes energy and time), and do so in such a fashion as to maintain good sound quality. The Low Complexity Communication Codec (LC3) does just that, supporting sample rates up to 48kHz. LC3plus extends the capabilities of LC3 with somewhat more elaborate handling of the data stream that allows for sampling rates up to 96kHz, lower total harmonic distortion, additional low-delay modes and features that come in handy in difficult transmission environments.

LC3/LC3plus are frame-based codecs. They can look at 7.5 or 10 milliseconds worth of samples, analyse those samples, and calculate a way to compress the information in those samples so they can be broadcast more efficiently. LC3plus offers an additional 2.5ms frame duration.

Encoding begins in the Low Delay Modified Discrete Cosine Transform module (LD-MDCT) which is "basically a very widespread time-to-frequency transformation used in perceptual audio coding. This is where the codec's time delay occurs," notes Alex. "The total algorithmic delay of the codec is the sum of the frame duration, say 10 milliseconds, and the lookahead from the encoder side contributed by the MDCT (2.5 milliseconds for most cases). That adds up to a total delay of 12.5ms for the 10ms frame duration. With LC3plus, the total algorithmic delay can be as low as 5ms for the 2.5ms frame duration."

The LD-MDCT feeds a bandwidth detector (BW Detect) which does exactly what it says on the tin. "If you configure the codec to work at 32kHz but someone is using a legacy 8kHz [*wireless phone*] handset, this could potentially lead to artefacts," said Alex. "The bandwidth detector can detect such band-limited signals, and basically control other tools to avoid problems."

The frequency components generated by the LD-MDCT are passed to the Spectral Noise Shaper (SNS), where they are quantised and processed. Alex described the SNS as an algorithmically complex tool that is implemented efficiently with DSP (digital signal processing) and which "maximises the perceptual audio quality by shaping the quantisation noise so that it's minimally perceived by the human ear."

Next, the Temporal Noise Shaping module (TNS) "reduces pre-echo artefacts for signals with a sharp attack," Alex explains. Codec designers test the quality of the encoding with recordings of castanets. These have sufficiently sharp transients to make pre-echo artefacts apparent, which the TNS (in conjunction with another module, the Attack Detector) can eliminate.

After spectral and temporal noise shaping, the Spectral Quantiser "calculates a global gain for a single frame and then quantises the spectrum with this global gain. The Spectral Quantiser works in an iterative way and estimates the number of bits required to encode the quantised spectrum later in the arithmetic encoding."

At lower bit rates, spectral quantisation inevitably creates 'spectral holes' which sound like robotic bird chirps. These 'birdies' are fixed with the Noise Level/Filling module that uses a pseudo-random noise generator to fill the holes.

To further reduce coding noise at lower bit rates in frames with pitched or tonal information, a Long Term Post Filter is applied. At higher bit rates this tool is turned off during decoding.

» and cinemas, but also personalised audio for anyone with LE Audio-enabled earbuds. Think silent disco or your own high-res music mix at a pop concert: all doable with the new standard.

Advanced Uses

The SIG is imagining Bluetooth LE Audio will be used for personal music sharing, announcements in airports, assistive listening, and similar consumer uses. But LE Audio is such a flexible and powerful standard that, like MIDI, it has the potential to create brand-new use cases and maybe even product categories far beyond the designers' original intent.

Can Bluetooth LE Audio and LC3 be used for professional audio production? It's quite possible. Wireless headphones with sound quality rivalling wired models are quite conceivable, as are wireless monitor systems for personal studios. LE Audio may also find a place in location sound for film, especially for scenarios where very small mics and transmitters are needed. Because LE Audio is multichannel, it may even be possible to use it with multiple wireless microphones. As for the new codecs, Alex Tschekalinskij, the engineer who worked on the LC3 and LC3plus codecs, reveals: "We had one exciting request from a service that allows you to create and play music together via an app or website. It's basically online music jamming. The service can be used by musicians who want to rehearse or produce music online. LC3plus offers low-delay modes and a dedicated high-resolution mode, so it would be a perfect fit!"

The ready availability to the entire public of high-quality, multichannel, wireless audio also suggests possibilities for new kinds of interactive drama and film. Nick Hunn notes: "You've got the potential to have 10 or more tracks," that can be streamed simultaneously. It would be entirely possible to create a new type of immersive cinema-in-the-round in which, say, part of the audience faces front, listening on their earbuds to some sci-fi baddie berate his army before the great battle for galactic control while, at the same time, a different part of the audience faces to the rear, tuned into the good guys' plans to defeat them.

Whether any of these or currently unimaginable new equipment and media get developed is anyone's guess. But what's certain is that in the years to come, Bluetooth LE Audio will open up numerous new ways to use wireless sound.

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Polyend Tracker

Sequencer & Sampler

Polyend breathe new life into an old technology with an instrument that's as forward-looking as it is retro.

RORY DOW

Trackers have a proud place in the history of computer music. Back in the late 1980s, the home computer revolution was in full swing. Computers like the Commodore Amiga and Atari ST were popular with gamers but also offered sophisticated programs for anything from desktop publishing to music-making. Trackers first made their appearance on these machines. By the time MS-DOS PCs became popular in the early '90s, the tracker was a well-established platform and formed the basis for communities of gamers and demo-scene makers to share and publish music.

Ultimate Soundtracker for the Commodore Amiga, by EAS Computer Technik, was the first tracker (and gave this type of sequencer its name). It used a vertical grid-based sequencing system that scrolled from top to bottom. It was designed for game developers, which might explain why the slightly spreadsheet-like programming interface caught on. The source code for Ultimate Soundtracker was disassembled by mischievous hackers and spread around. This spawned new versions such as NoiseTracker and ProTracker which were distributed as freeware, helping broke music-makers to get started and ultimately spawning a huge free-music movement.

Part of the tracker phenomenon's success was the ease with which you could share songs. A finished song is saved as a 'mod' file, which contains everything needed to play it: instruments, samples, sequencer and song data. Mod files are compact enough to be shared online and used in games without them becoming bloated. You could fit a lot of mod files on a floppy disk.

Skip forward to today. Trackers are still around. Renoise is probably the most well-known software tracker still under development, and has matured into a full DAW environment. But something that's rarely been seen in

35 years of tracker history is a hardware tracker. Enter the Polyend Tracker.

Hit The Track

Housed in a sleek 282 x 207 x 33mm enclosure, the Polyend Tracker revolves around a 7-inch colour screen, a large aluminium data encoder, and plenty of buttons. The screen is bright and big enough for programming, editing audio, file management and the like. The mechanical buttons to the right allow you to navigate to different areas such as pattern and song editing, sample recording and editing, instrument editing and so on. Just below are buttons for transport, arrows for on-screen navigation, and copy/paste/delete. The eight buttons below the screen serve as soft-function keys. Their purpose will change depending on what's on screen. Finally, there's a group of 48 soft silicone keys known as 'the grid'. This is mostly used to play instruments chromatically, but can also function to select instruments, play slices, choose values, and other things. They are not velocity or pressure-sensitive, but serve well enough for programming.

A Tracker project contains a song with up to 255 patterns. A single pattern is eight monophonic tracks. Those eight tracks can be used to play samples from a pool of up to 48 instruments loaded »

Polyend Tracker

£459

PROS

- It's a hardware tracker.
- Intuitive to use once you grasp the tracker concept.
- Solid metal build.
- Some great included content, including many full songs.

CONS

- No support for stereo samples.
- One one line output.

SUMMARY

Polyend have done a fantastic job of turning the humble tracker into a hardware device. It feels great to use and should appeal to both hardcore tracker aficionados and newcomers alike.

As well as the micro-SD card slot, there is a USB Type C port for power and bidirectional, class-compliant MIDI, a power button and (all on 3.5mm jack sockets) stereo line output, stereo line input, mic input and MIDI In/Out.

into RAM. Samples are mono, 44.1kHz (stereo samples will be converted automatically). The total sample memory is just 8MB, which is around 130 seconds, although you can double that by using a low-quality import function that halves the sample rate. If 8MB seems rather ungenerous by modern standards you wouldn't be wrong, but a large part of the tracker ethos is keeping things streamlined. In practice, I didn't find the 130-second sample limit to be a constraining factor.

Sampling

A tracker song starts with loading or recording some samples. The micro-SD card can be used to store and load WAV files, as well as Tracker projects. Samples can be loaded into one of the 48 empty slots available. Alternatively, you can sample from several sources. The line and mic inputs on the rear are the obvious choices, but Polyend have added another option: an FM radio. The cable in the line output doubles up as an aerial and it works remarkably well. Sampling from FM radio is a neat touch and can be a great source of unusual material.

After recording, you can trim and save it to the micro-SD card. The silicone keys double up as a keyboard to name your file. The layout is shown on screen. If you're feeling lazy, Polyend have included an entertaining auto-naming function: simply press the Auto Name button until you find a name you like. Whether an SD card full of samples named things like 'soggy frogs', 'nauseating wind' or 'acid ladybug' is a good thing or not is open for debate, but it certainly takes the hard work out of it.

Once you've got a pool of samples loaded, it's time to decide how the samples will play back. The default is a simple one-shot mode, in which case you're ready to start sequencing. But samples can also be crafted into instruments in a variety of ways. Samples can be looped, sliced or used as wavetable or granular sources.

Wavetable sources are sliced into 2048 sample waveforms. An LFO or envelope can be used to modulate the wavetable position. This works better when loading WAV files specifically designed as wavetables, of which a good number are included in the factory content. But thanks to the standard wavetable specs, you can also load in wavetables created with software like Serum and WaveEdit.

Granular playback is similar to wavetable playback. It will chop the sample into small windows which loop endlessly when a note is held. Again, an LFO or envelope can move the position of the window within the file, turning almost any sample into a pad-like drone.

There is also a suite of destructive effects which can be applied to samples. These include normalising, reverse, overdrive, fades, time-stretch, and more standard effects like chorus, flange, EQ, bit crush, compression, and limiting. You can preview any effect before applying it.

The final steps to crafting your sample into an instrument involve adjusting volume, panning, tuning, overdrive, bit-depth and filtering. Simple low-, band- and high-pass filters can be applied and modulated with a dedicated envelope or LFO. There are also dedicated envelopes or LFOs (not both) available for volume, panning, cutoff, wavetable position, granular position and fine-tune.

Lastly, each instrument can send to two global effects, reverb and delay. Both are reasonably simple but sound excellent and are much needed to help add stereo glue to what might otherwise be a very monophonic soundstage. Whilst on the subject of global effects, the master section also contains an EQ, limiter (with side-chain), and two single-parameter effects named Bass Boost and Space. All of these can be used to shape the overall output of the Tracker and give you a more record-ready sound.

> "Every time I switched it on, I was immediately making music. I think that more than anything sums up the Tracker — it's the efficiency and speed with which you can be creative, and that's priceless."

Pattern Mode

The uniqueness of tracking, of course, doesn't come from samples or filtering or effects — it comes from the sequencing environment. Each pattern can be up to 128 steps, and contains eight tracks. Each step of each track can play back one of the 48 instruments loaded into the pool.

A step can hold four bits of information: the note pitch, the instrument number, and two 'FX'. An FX can be chosen from a large list that includes simple items like volume, panning, tuning, gate length and swing. But there are rather more exciting options too: trigger probability (aka Chance), rolls, slides, randomisers, sample reverse, slice selection, sample »

It's Also A Gaming Device?

As unlikely as this seems, the Tracker contains a complete Nintendo Entertainment System (NES) emulator. In the File screen is a Games button that shows the contents of the Games folder on the micro-SD card. In this folder are a few NES ROM files that can be loaded and played using the arrow keys and various other buttons. It should make that long plane journey a little less dull.


MAKING PASSION HEARD
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OD505
ACTIVE DYNAMIC
VOCAL MICROPHONE
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The OD505 is a tough stage mic with the soul of a studio microphone. Our proprietary Open-Acoustics-Technology as well as the state-of-the-art dual capsule design result in a superior noise suppression, making the OD505 the perfect choice for capturing brilliant vocals in a range of uses, whether on stage, in live studio recordings or in rehearsal situations.
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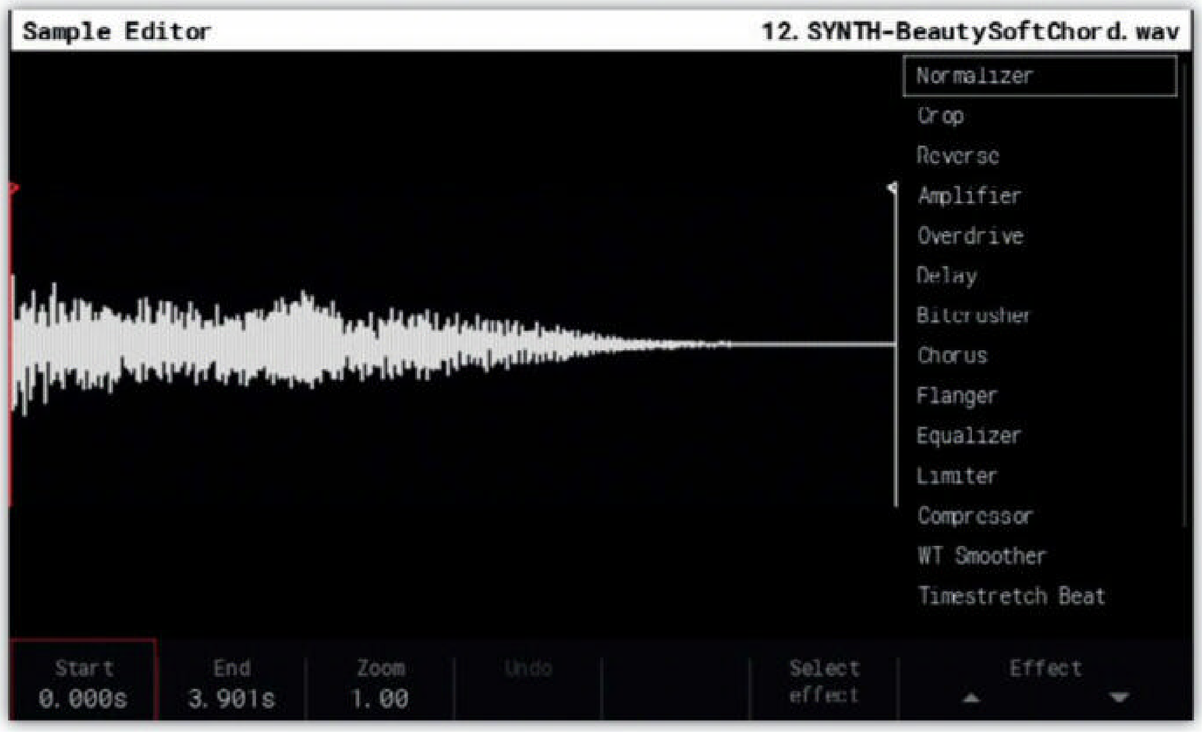


The sample editor offers a suite of effects with which to mangle your samples.

Artist Editions

In April 2021, Polyend announced three special Artist Edition Trackers. Limited to 300 units each, the Trackers were adorned by artwork created especially by each artist. Legowelt, Bogdan Raczynski and Pete Cannon also supplied tracks written on the Tracker which were pressed to vinyl. At the time of writing this review, some online retailers still had stock of the special editions but by the time you read this, they could be sold out. The good news is that you can head to the Polyend website and download the song projects from the three artists to open, study and remix.

start position, filtering, delay, or reverb send. All of them can be used to spice up your sequencing.

I won't go through each of the FX one by one as many of them are self-explanatory, but I'll highlight a couple of favourites. Rolls are an excellent way of creating snare rolls (with any sample of course) without the laborious effort of inserting each step. On an existing note, you add the Roll FX and choose a value. The values are given shortcodes. For example, R2 will play standard roll (a simple repeated note) twice per step. This will continue until another non-empty step is encountered. A value of Rv4 will play four notes per step with velocity (amplitude) decreasing for each step. RV12 will play 12 notes per step (heading into glitch territory) with the amplitude increasing. Programming these rolls using a traditional piano roll sequencer can be time-consuming, but trackers make it a breeze, especially if you want to easily experiment with the timing or slope of the roll.

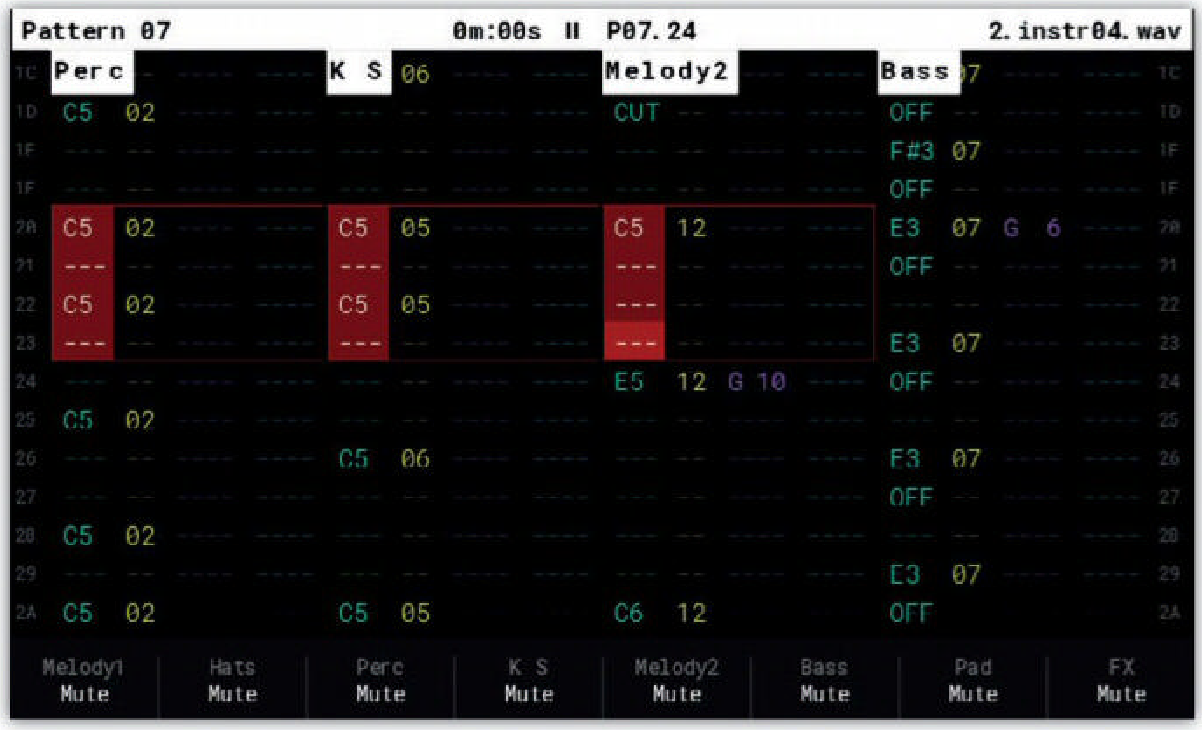
Another favourite is Glide. With a normal MIDI synthesizer, getting a note to glide smoothly from one pitch to another can be a multi-step process. You have to make sure that you have portamento enabled, you have to set a suitable glide value. You have to make sure the sound has suitable retriggering options so that when a legato note is played, the envelopes retrigger. Then you have to program your MIDI sequence with legato notes for those you wish to glide. With trackers, glide is a simple FX you choose on a per-note basis. You don't need to worry about any of the synthesis elements: it just works and as a bonus, you can have a different slide speed for each note if you wish — something that is pretty much impossible with many MIDI instruments.

The basic process of editing and recording a pattern works as follows. The eight tracks are arranged in vertical columns. Each step is a cell, so if a pattern is 64 steps long, there will be 64 rows. You can move the edit field around horizontally and vertically by using the arrow keys or the data encoder. By pressing Shift and using the arrow keys, you can select an area larger than a single step. This allows you to perform actions such as preview, delete or copy/paste on multiple tracks and steps.

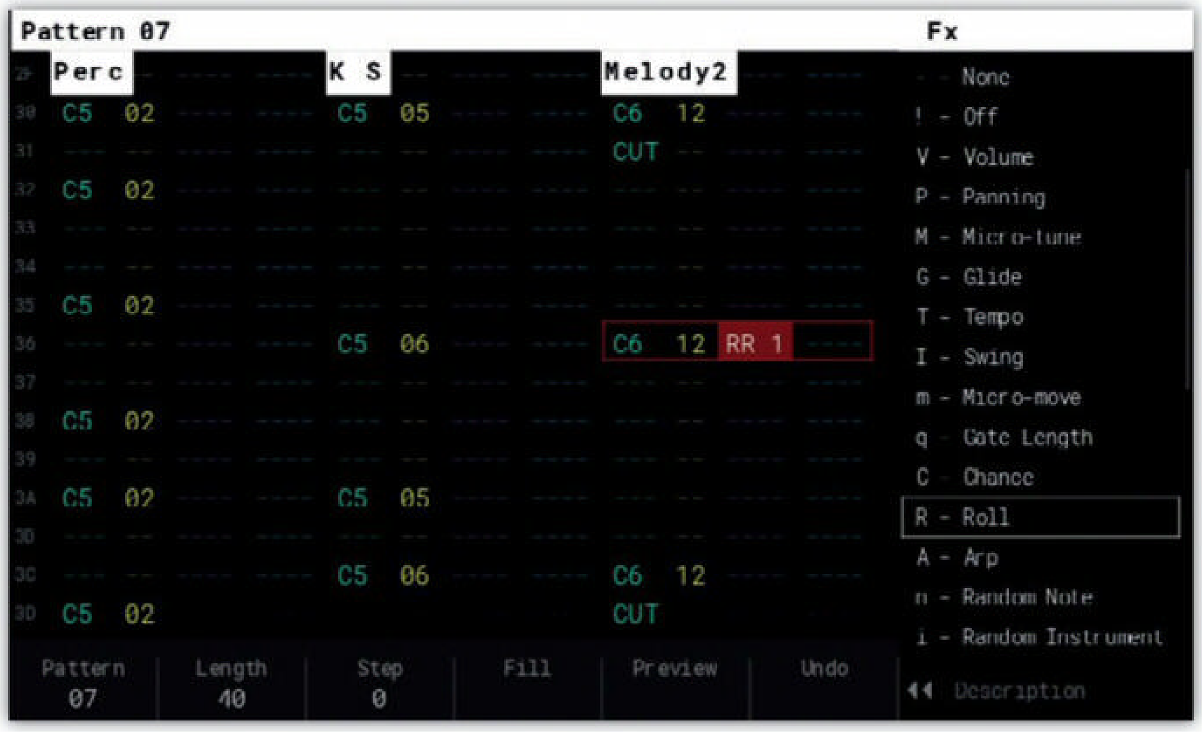
Editing a step involves choosing one of the four step parameters to focus on: note, instrument, FX1 or FX2. You do this by tapping one of the four pastel-coloured buttons on the right of the unit. A long press will pop up a menu allowing you to choose a value to insert. If you're editing notes, this will pop up a keyboard. If you're editing the instrument field, it will pop up a list of the instrument pool. For the FX, it'll be a list of all available FX. Once you've chosen, you can edit the value using the data encoder.

There are advantages with this kind of editing, versus a more traditional piano roll. Firstly, you can always see what's happening on other tracks. Having an overview of what's happening in your entire pattern whilst you work on one part of it is incredibly useful. Secondly, the ability to just move from one track to another whilst remaining in the same window speeds up editing tremendously.

Of course, moving around the spreadsheet and editing everything manually isn't the only way to build up a sequence. You can record notes from



Tracker sequencing may look like accounting software to some, but there's method in the madness.



The real power in tracker sequencing is the FX. Every note of your pattern can have two different FX from a long selection.

» a standard MIDI keyboard attached to the MIDI in or USB port whilst the sequencer runs. As tracks are not polyphonic, any polyphonic playing will add additional notes to adjacent tracks. Optionally, you can record off-grid by enabling micro-timing, where notes will be recorded with a micro-timing FX enabled which records how far off-grid your playing is. Velocity can also be recorded in a similar optional manner.

The last way to input data involves algorithmically filling steps, which can be a great way to create pseudo-randomised sequences. If you select a cell, or a range of cells, and press the Fill soft-key, the Fill Notes menu will pop up. If you have Instrument or one of the FX fields selected instead of Note, the Fill function will adapt itself to work with that field. When filling Notes, you can choose several parameters including where to fill (optoins include random, existing notes, every X notes and so on), a scale, a fill type (constant, range or random) and then a note or range of notes to choose from. Using Fill for Instrument or FX lanes is similar: you choose the frequency and range to fill and hit the button. If you don't like the results, there is Undo, which remembers the last 20 edits.

There are yet more functions to help you edit. A suite of pattern-based functions are available on the soft-keys at any time and include pattern copy and paste (much used when you start to develop your song), pattern shrink and expand (which half or double the time of a pattern), duplicate (doubles the length and copies the contents), and invert (reverses the order of selected steps). There is also a Render Selection key, which will render the current selection to a new sample. This can be an excellent way to remix patterns on the fly.

If you want to involve external instruments in your sequences, the Tracker can output MIDI. If you scroll to the end of the Instrument pool, you'll find 16 MIDI channels that can be selected instead of an instrument. Any step using a MIDI command cannot also be playing an internal instrument, so it will reduce the available voice count. The Volume FX can be used to alter the velocity of a MIDI note, and there are even FX lane options to send MIDI Control Changes, Aftertouch and Program Changes. A MIDI Chord FX can play a triad in a single step. The codes are somewhat cryptic, so having the manual nearby so you can figure out whether chord 057 is a sus4 or a dim7 is advised. Overall, I'd say the MIDI capabilities are basic but welcome. You won't be scoring an orchestra, but if you need to change patterns on an external drum machine, or trigger a basic bass line on an analogue synth, you should be fine. One thing to note is that you cannot address the USB and 3.5mm MIDI outputs separately, so if you have devices connected on both they'll need to be set to different MIDI channels, otherwise they'll play together.

Song Mode

Once you have a few patterns prepared, you'll want to arrange them into a song. Song mode allows you to chain patterns together. You choose a song tempo and add slots to the Song. Each slot represents a single pattern. If you wish to repeat a pattern more than once, you simply add it again as many times as you'd like it to repeat.

Although tempo is song-based, this can be overwritten using the Tempo FX on a note in the pattern. In this way, tempo is easy to hijack on a pattern-by-pattern, step-by-step basis.

The Song mode can play in one of two ways. At the push of a soft button, it toggles between playing through the pattern playlist in order and repeating the current pattern indefinitely. Using this can form the basis of a live performance where you follow a predefined list of patterns, but can pause the playlist at any time to allow a particular pattern to loop for longer.

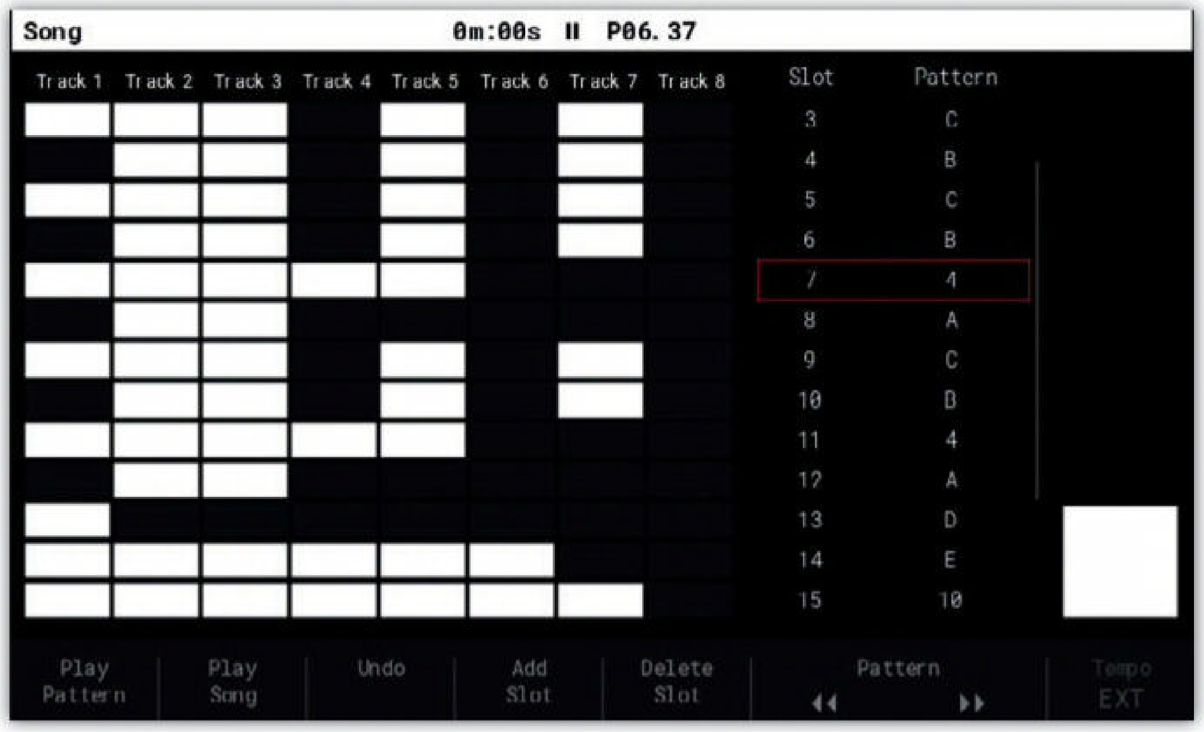
Performance Mode

Speaking of performing, there's a whole page dedicated to the live manipulation and remixing of the currently playing pattern, and rather clever it is too. The Perform page is centred around 12 parameters that can be manipulated: volume, panning, tune, low-, high- and band-pass cutoff, delay send, reverb send, sample position, sample end, sample playback (direction) and volume LFO speed.

These 12 parameters can all be assigned three offset values. Those values can be triggered as the pattern plays by using the 48 silicone keys. The top row of 12 activates the normal pattern playback. The three rows underneath trigger offset values which you can enter. So for example, whilst your pattern is playing you might set up the Tune parameter offsets to -12, -5 and +3. Then by triggering these, you can transpose the entire pattern down an octave, down to a fifth or up to a minor third.

You might not want to affect the whole pattern at once of course, and this is taken care of with the eight soft-keys, which enable or disable their respective tracks from being affected by the performance offsets. From the same screen, you can also mute tracks entirely to create breakdowns.

The last trick performance mode has is potentially my favourite. By holding down one of the eight soft-keys and scrolling the data encoder, you can remix patterns by choosing a track and having it play back from a completely different pattern. For example, you might have kick and snare on Track 1, and bass on



Song mode chains patterns together and offers a useful overview of which tracks are used.

Track 2. If you have five patterns, you can combine the kick and snare and bass from any pattern. When you apply this across all eight tracks, you have a huge number of possible combinations.

I can imagine entire live sets being played from just the Performace page. If you organised your tracks correctly, you could use the track-swapping feature to gradually build up songs and introduce variations all whilst essentially playing one pattern. Performance mode works equally well with entire songs, though, and you can switch between performance mode and instrument editing without losing the performance tweaks.

One last thing to note for anyone who plays live is that the Line input can be used to plug in another synth, sequencer, or drum machine, with the level being accessed in the mixer pages. This is a useful addition, as mixer inputs can be in short supply when you're gigging with the least amount of gear possible. Overall, I feel the live potential of the Tracker is impressive and a worthwhile addition to the tracker workflow.

Taking It Beyond

The final optional step of your song-making experience may end with a need to transfer your song to a DAW. As much as any hardware device is fun to play, nothing can compare to mixing and mastering on a computer. To help, Polyend have added a bunch of export options. You can export an entire song to WAV, or song stems which include eight mono tracks, the delay and reverb sends and a single stereo master. Renders are 16-bit, 44.1kHz. Additionally, there are options to export single patterns as either a mix or as stems. After that, it's simply a case of copying the WAV files from the micro-SD card and importing them into your favourite DAW.

Conclusion

It would be a mistake to write off trackers as a relic of the past. When I was 18, I had an Amiga and many of my first electronic compositions were written in Octatrack. It served to introduce me to the wonderful world of creative sampling. Of course over the years, I progressed to an Atari with Cubase, then PCs with DAWs.

But the Polyend Tracker is more than just a nostalgia trip. I'd forgotten just how quick tracker sequencing can be. The permanent visibility of all tracks, the ease with which you can move between them, and the efficiency with which you can create complex effects like rolls, slides, and reverses all adds up to a unique way of crafting electronic music.

Polyend have done a great job of transitioning the tracker from software to hardware. The package as a whole feels fun and intuitive to use. I daresay anyone who already knows their way around a tracker will take to it like a duck to water. There is even an option in the preferences to enable hexadecimal numbering for those that want the full authentic '90s experience. For others who might find hardcore tracking a little intimidating, there are some nice options to ease you in. For example, the version 1.4 update that landed just as this review was wrapping up adds a horizontal pattern arrangement setting for anyone freaked out by the vertical option.

Import

Traditionally, tracker songs were distributed around the internet in a few different formats. IT and MOD files were two popular formats that can be imported by the Polyend Tracker. Basic properties like samples, instruments, patterns, song structure, and volume info should all import correctly. The Tracker can also export IT files if you ever need to load your Tracker song on your old Amiga.

Tracker music, far from being simplistic as one might imagine, can be deep and complex. And it goes way beyond the archetypal chiptune game music. Jungle, rave, IDM, glitch, breakcore, electronica, and techno all owe a debt to the humble tracker. If you want some evidence of the depth and beauty of tracker music, seek out the album *Claro* by Brothamstates or *Rossz Csillag Alatt Született* by Venetian Snares. Artists such as Aphex Twin, Legowelt, Richard Devine, Cristian Vogel and John Tejada use trackers as part of a wider palette of tools. Tracking as an art form seems to have a bright future.

The Tracker does have a few limitations. It doesn't always play brilliantly with other equipment, for example. You can use it to sequence external MIDI instruments with the MIDI output, but this will use up precious tracks, of which there are few to begin with. Also, the inclusion of only one stereo audio output means that you can't use external effects very easily. In short, the Tracker feels self-contained. That's not necessarily a bad thing — it has all it needs to write full songs, after all — but if you're hoping to introduce it to all your other toys you may find stumbling blocks.

Despite this, I can't get over how much fun the Tracker is. I forgive its loner nature because it's such a pleasure to use. The Tracker somehow reminds me that music-making is supposed to be enjoyable. As a lap-top device or a holiday beat-maker, it would be at the top of my recommendation list. When the larger studio feels like a grind, picking up the tracker and banging out some jungle beats is a refreshing palette cleanser. Every time I switched it on, I was immediately making music. I think that more than anything sums up the Tracker — it's the efficiency and speed with which you can be creative, and that's priceless.

£ £459 including VAT.
W www.polyend.com

Focusrite Clarett+ 8Pre

USB Audio Interface

Necessity may be the mother of invention, but it can also significantly improve audio interfaces...

HUGH ROBJOHNS

The well-known UK pro-audio manufacturers Focusrite introduced their original Clarett suite of Thunderbolt-connected interfaces in 2015, aimed at high-end project-studio owners seeking really top-spec performance. Three years later, updated models were shown at NAMM 2018, re-engineered with the ubiquitous USB interface format, and the resulting 2Pre, 4Pre and 8Pre models quickly became enormously popular. One notable attraction of the whole range was that even the smallest model could be expanded via ADAT inputs to accept eight additional analogue channels, ideally from the matching Clarett Octopre mic preamp/converter. My colleague Sam Inglis reviewed both the original Clarett 8Pre-Thunderbolt (in the October 2015 edition), and the Clarett 8Pre-USB (in the March 2018 edition of *Sound On Sound*).

However, last year Focusrite faced a big problem with its Clarett USB range! As explained in the 'Chip Crisis' article published in the *Sound On Sound* September edition, a huge fire last year all but obliterated the Japanese factory of Asahi Kasei Microsystems — AKM being one of the major suppliers of high-quality A-D and D-A converter and digital interface chips to much of the pro-audio industry.

Needless to say, Focusrite's Clarett range employed AKM converter chips extensively, and without a consistent supply of converter chips Focusrite can't build audio interfaces. So the company was effectively forced into redesigning the Clarett product range to use alternative brands of converters — something which is far from a trivial task!

However, having been handed life's sour lemons Focusrite decided to make tasty lemonade by taking the opportunity to build upon and enhance the Clarrett range's already impressive performance. New high-end converters were selected from a well-respected manufacturer called Cirrus Logic; the A-Ds are CS-5381 chips with an advanced multi-bit delta-sigma architecture and 24-bit output, while the D-As are CS-43198 converters, which use 32-bit oversampled multi-bit modulators. Focusrite's engineers have also meticulously designed the associated output filtering parameters for optimal performance. The technical improvements brought about by this substantial redesign are such that Focusrite thought the product name warranted enhancement too, hence Clarett+.

The most obvious benefits of the Clarett+ redesign are greater dynamic range and lower distortion figures across the board. Of course, the original Clarett was hardly a slouch in this regard, but the improvements are not only measurable but audible too, especially when working in challenging conditions. Given how good modern digital equipment already is, Focusrite's update actually represents quite a significant step forward in engineering terms.

Overview

I was provided with the flagship model for this review, the Clarett+ 8Pre, which is described as "a powerful studio-grade 18-in/20-out audio interface". Aside from the incremental performance lift afforded by its technical redesign, the Clarett+ 8Pre is pretty much identical to the preceding model and I'd urge you to read Sam's reviews mentioned above for the full 'SP'. However, key practical points to note are that the unit is mains-powered via an internal switch-mode PSU (accepting 100-240V AC through a standard IEC inlet), and it connects to the host computer via a USB-C connector on the rear panel (USB C-C and USB C-A cables are included in the box).

Its eight premium preamps are all connected via 'combi' XLRs for mic/line inputs, with six on the rear panel and two more on the front panel (which also have high-impedance JFET instrument input modes). Ten analogue line outputs are provided on TRS sockets at the rear, two being dedicated as monitor outputs, while a pair of independent stereo headphone outputs is available on the front. Ten channels of digital connectivity

Focusrite Clarett+ 8Pre

£869

PROS

- Raises the audio quality and technical performance to even higher levels.
- Even the smallest model in the range can be expanded via ADAT.
- Focusrite Control utility and driver performance remains very good.
- Air EQ feature is a useful effect.

CONS

- None given the price/performance ratio.

SUMMARY

An enforced but very welcome and unexpected performance upgrade to Focusrite's ever-popular Clarett range of seriously capable interfaces.

in and out are provided through stereo S/PDIF (RCA-phono or optical) and ADAT (assuming the ports aren't being used for S/PDIF). The ADAT interface provides eight channels at base sample rates, of course, and proportionally fewer at higher sample rates using the S/MUX protocol. A pair of DIN sockets is included for old-school MIDI in and out, while a word-clock BNC output allows synchronisation of other connected digital equipment (such as the Clarett Octopre as an input expander).

The transformerless preamp design claims ISA110 heritage, but while the gain stage is based around an NE5532 op-amp just like the ISA110, the original Lundahl input transformer has been replaced with an electronically-balanced discrete transistor input stage. Nevertheless, the design achieves remarkably low noise (specified as an EIN of -129dBu A-weighted) and has excellent headroom margins — the maximum mic input level is a very generous +18dBu, while the line input can take +26dB before complaining, and even the instrument inputs can cope with +15dBu. So there shouldn't be any problems with either high or low source levels, although the maximum channel is gain is 57dB, which precludes working directly with really quiet sources.

Focusrite have also retained the 'all-analogue Air EQ' feature, which can be applied to individual preamp channels. When activated, the high frequencies are boosted in two stages by +4dB overall to bestow crisp transients, enhance presence-range clarity and, yes, to 'add air' to whatever you're recording. It is quite an alluring effect on vocals and acoustic guitars, in particular, but one which should probably be used with cautious discretion, I think — you really can have too much of a good thing! Interestingly, activating the Air option also reduces the microphone preamps' input impedance from 6.2kΩ to 2.2kΩ — but while that might alter the tonality of some dynamic mics slightly, any changes would be thoroughly swamped by the Air effect anyway!

The unit's front panel carries physical controls for phantom power (switchable in two groups of channels 1-4 and 5-8), preamp channel gains, headphone volumes, and the dedicated monitor section (with its own volume control and dim/mute buttons). Further configuration is via the elegant Focusrite Control app (for PC, Mac and iOS devices), which provides remote access to some relay-switched preamp settings (line/instrument mode on channels 1-2, and the Air EQ on all channels), as well as configuring the internal signal routing between the USB channels and the physical inputs and outputs to create monitor mixes, selecting the internal sample rate, and so forth.

Bench Tests

On the A-D side of things the new Clarett+'s dynamic range figure remains much the same as the previous model. My own test bench measurements using an Audio Precision system gave an AES17 dynamic range measurement of 117.7dB (A-wtd), which conforms to the published specification of 118dB. I'm not sure why, but the Clarett+ 2Pre is claimed to achieve a slightly better result than its predecessor at 119dB (A-wtd).

However, while the A-D's dynamic range figure is essentially unchanged, the distortion and noise performance has been improved, with the THD+N parameter being reduced to -110.1dB (or 0.0003 percent) from -107dB (0.0004 percent) in the previous generation product. That translates in the real world to a marginally clearer, cleaner, and more transparent source recording, especially if tracking with generous headroom margins.

It is the Clarett+'s line outputs which show the greatest improvement, though, with a claimed 6dB improvement in dynamic range combined with 3dB lower distortion. The 8Pre's new converters are aligned such that 0dBFS from the DAW or digital inputs corresponds to +18dBu at the analogue outputs, and my bench tests gave an AES17 dynamic range figure of 123.9dB (A-weighted) for the D-A stage. This corresponds to Focusrite's published specification of 124dB and is nearly 6dB better than the previous model's 118dB (A-weighted).

The THD+N figure for the D-A measured -106.1dB (the published spec is -106dB), which is also a very worthwhile improvement over the previous model's -103dB. The headphone outputs have also gained 3dB more dynamic range and 3dB lower THD, too — all of which means lower noise and more pristine low-level details in reverb tails and room atmospheres.

To provide some real-world context here, an AES17 dynamic range figure of 124dB (A-weighted) — which the Clarett+ scores for its balanced line outputs — means it rates in the top five of all the interfaces and converters I've measured to date, alongside high-end products like the Benchmark DAC2 HGC, and out-performing RME's ADI-2 Pro. The 118/119dB (A-weighted) figure for the A-D conversion places the Clarett+ on the borderline of the top 10, rubbing shoulders with classy products like the Focusrite's own RedNet and ISA digital card, Crookwood's M1 mastering console, Prism's Titan/Lyra interfaces, and Cranborne's R500 rack. So these really are very impressive specifications by anyone's standards, let alone for a moderately priced project studio interface.

Conclusion

Although the need to update the Clarett range was forced by circumstances beyond Focusrite's control, the results are impressive and well worthwhile. The new Clarett+ is the same but better, both measurably and audibly, and it stands up extremely well against seriously high-end equipment.

£ £869 including VAT.
W www.focusrite.com

Positive Grid Bias FX 2 Elite

Amplifier & Effects Modelling Software

With both desktop and mobile versions, and a tiered pricing model, there's a Bias FX 2 version to suit any budget.

BOB THOMAS

As its name suggests, Positive Grid Bias FX 2's main focus is on modelling pedal and rackmount hardware effects units. It does include amps and cabs too, but the idea is that it can be used in conjunction with the company's separate Bias Amp 2 if you feel the need for more control over that side of things.

Bias FX 2 features 122 effects pedals, 18 rackmount effects units, dedicated Fuzz, Time and Harmonizer modellers, and 100 amplifier and loudspeaker cabinet models, all of which have been arranged in 210 carefully curated factory presets. As with Bias Amp 2, which I reviewed in *SOS* October 2018, both desktop (standalone and plug-in) and iOS versions are available in three tiers: Standard, Pro, and Elite. All three tiers share a common core, but the latter two add more features and functions. It's the Elite version, the one with the most extensive feature set, that's under review here.

Presets

One of the attractions of Bias FX 2 is its intuitive drag-and-drop GUI. This makes editing, building or saving presets (signal paths made up of effects, amplifiers and loudspeakers) a simple process. Helpfully, the factory presets are grouped by genre: Pop, Blues, Rock, Metal, Insane, Alternative, Bass and Acoustic. The

The signal can be split and recombined, to create parallel effects/processing/amp paths.

individual preset names contain fairly obvious descriptions of their sonic character, such as 'Whole Lotta Page' in the Rock bank.

Editing a preset is simply a matter of adding, moving, removing, replacing, activating or bypassing amplifier, cabinet and effects models along the signal path, and setting their individual controls to get the sound you want. If a single signal path isn't sufficient, you can switch to a dual path. This creates two parallel paths by inserting a two-channel Splitter before the current amplifier and a stereo two-channel Mixer after its loudspeaker. Although neither the splitter nor the mixer can be moved, you can 'reposition' them by shifting FX models to and from either side of both, making it possible to create a dual path that starts with a splitter and ends with a mixer. One operational quirk is that the mixer at the end of the dual path defaults to centre-panning both paths, so it's necessary to manually pan left and right if you wish to create a stereo output.

In the original Bias FX program, amps and cabs were permanently paired, but in Bias FX 2 they can be selected individually. This makes possible a new feature, the Middle Effect, which allows you to place multiple pedals or rackmount effects between an amplifier and its loudspeaker, on one or across both sides of a dual path. To build a preset from scratch, you have to clear the routing of a current preset, which leaves you with a default single signal path containing an MXR-ish gate pedal, a '66 AC Boost V2 amplifier and a British 30 speaker cabinet as your starting point. Edited presets can be stored in one of two User Banks.

While amps and cabs are included, note that any amplifier editing (other than adjusting its main front-panel controls) must be done in the separate Bias Amp 2 software. To edit a Bias FX 2 amplifier in the desktop standalone and plug-in versions, you first have to find that amp either in Bias Amp 2 or in Positive Grid's web-based Tonecloud. The latter is a vast resource that's accessible directly from any Bias program, in which you'll find additional free and updated amp models and Bias FX 2 presets, both from Positive Grid and the user community. Having

MIDI Control

Bias FX 2's MIDI control implementation is comprehensive and flexible, as a MIDI CC (Continuous Controller) can be assigned to every switch, knob or operational function using the program's Learn function. In the desktop standalone version, global MIDI controls (Amp/FX switching, Preset Previous/Next, Bank Previous/Next, Utilities including Tuner, Tap Tempo and Scene, and Looper commands) are assigned in the Settings menu. All Amp and FX control commands in both standalone and plug-in versions are assigned only from the signal path view by right-clicking on the pedal or amp. In the plug-in version, those commands can be controlled both from an external MIDI source and by MIDI automation in the host DAW. In Bias FX 2's iOS incarnation, the Settings menu gives access to its Global MIDI Control assignments (Preset Previous/Next, Metronome On/Off and Tap Tempo), and to the Live View (foot controller) assignments. MIDI commands for individual Amp and FX controls are assigned as in the desktop version, but with a long tap replacing the right-click. The iOS tiers also support IAA (Inter App Audio) and Audiobus.

Positive Grid
Bias FX 2 Elite
£216

PROS
- Attractive and intuitive user interface.
- Wide choice of factory amplifier and effects emulations.
- Integral looper/backing-track player.
- The free Tonecloud repository is an amazing resource.
- Available for Windows (8 and above), Mac (10.12 or later) and for iOS (11 or later).

CONS
- If you're into redesigning amplifiers, you'll need Bias Amp 2.
- No option for Android users.

SUMMARY
A comprehensively featured, highly accurate amplifier and effects modelling package with a great sound and an attractive and intuitive user interface.

The Looper can be placed pre or post the processing.

edited the amp in Bias Amp 2, you save the result and load the edited model into Bias FX 2 from that program's Amp Finder menu, where you'll find direct links to folders in Bias Amp 2.

The amplifier editing process is much simpler in Bias FX 2 Mobile, as both it and Bias Amp 2 Mobile have dedicated on-screen buttons that copy and paste amp models from one to the other. Editing is simply a matter of selecting the amp to be edited in the Bias FX 2 signal path, pasting it into Bias Amp 2, editing it, and pasting the result back into Bias FX 2. The edited amplifier can also be saved in Bias Amp 2 and/or uploaded to the Tonecloud.

Should you feel that Bias Amp 2 and Bias FX 2 don't offer quite enough choices as standard, you can purchase optional Acoustic, Bass and Metal Signature expansion packs for Bias FX 2. Also available (and activated for free when you purchase the desktop Bias FX 2 Elite) is the Bias Pedal effects pedal modelling software. That program's extensive stompbox collection is available in Bias FX 2 and features a Pedal Match function, should you wish to create models of your own pedals. Your favourite loudspeakers are also catered for in Bias FX 2, as not only can you load third-party loudspeaker impulse responses (IRs) but also there are three official Celestion loudspeaker IR expansions available for purchase. A forthcoming Celestion Inside feature will offer access to official Celestion IRs within Bias FX.

Scenic View

Bias FX 2's new Scene facility (desktop standalone and iOS versions only) is analogous to storing pedal on/off combinations in a loop switcher. You can store a maximum of four Scenes in each preset and, should you require, recall these individually using MIDI. Incidentally, Scenes are not the same as Quick Snaps, which are snapshots for instantly comparing various presets and edits, and can only be recalled using a mouse or touchscreen. In the standalone version, the Scene function is on screen, whereas in the iOS version it comes under the (MIDI-controlled) Toggle switch menus in the Live (foot controller) View.

> "In the original Bias FX program, amps and cabs were permanently paired, but in Bias FX 2 they can be selected individually."

Although not new to Bias FX 2, the Pedalboard view is a nice touch that simplifies the preset view, displaying only enlarged versions of the amp head(s) and pedals to make on-screen editing slightly easier. I like the concept, though I'd prefer a more logical layout than the current vertical-ish arrangement, which I find slightly confusing.

Looper

Bias FX 2 (desktop standalone and iOS only) contains a convenient chromatic tuner, and, perhaps more significantly, a Looper, which allows you to record and layer loops and, in the standalone version only, to play or record along to backing tracks. Loop lengths can be set to 1, 2, 4, 8, 12, 16 or Free, the last of which doesn't confine you to a set number of bars. The tempo can be sync'ed to your DAW, tapped in, or entered manually in values from 40 to 400 bpm. Countings (ie. time signatures) range from 1/4 through to 9/4 in 1/4 increments, and you can select, separately, a one-bar count-in and a click track.

Before recording a loop you can set the Action after Record (play or overdub) and, if you have set the loop to play after record, you can also set when overdub recording starts (next loop start or immediately). Operation of the Looper is controlled by a large REC/DUB 'button'

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Guitar Match

A significant new expansion of Bias FX 2, for both the standalone and iOS versions, is the Guitar Match feature. This is designed to allow you to have your guitar sound like any one of 20 emulations of well-known (and some not so well-known) guitar types, across several models of the more popular shapes and pickup combinations. It's an extremely useful feature if you don't have a good selection of guitars. Setting up Guitar Match requires you to first create a 'source guitar' by profiling your guitar and its pickups. This is a simple process in which you specify your guitar's type (body shape and pickup combination) and then simply play 'arpeggios' across all six individual strings, at seven positions on the neck (open, followed by barres at two-fret intervals up to the 12th fret) for each pickup. Having done that, you save the resulting profile, select a pickup, pick the emulation you want your guitar to sound like, EQ to taste, set the blend between the original and emulated sounds, and start playing. The whole process is quick, easy and produces good results, especially if you match source and target pickup types. If you need them, multiple source guitar profiles can be saved.

Positive Grid's Guitar Match facility enables you to impose another guitar's character on your own guitar.

that shows the current bar position in the loop, and which changes function once a loop has been recorded, in conjunction with a vertical menu offering Play/Stop, Replay (stops and restarts the current loop), Undo (clears the last overdub) and Clear (erases the current loop). You can loop hands-free too, since all these operations can be controlled using MIDI.

You can position the Looper pre or post Bias FX 2's processing, so that you can record either clean for reamping (pre) or with a different sound on each overdub (post). Importing a backing track (standalone only) to play along with, or to loop or record to, is a simple browse-and-select process. When monitoring the Looper, the relative volume levels of the metronome, the recorded loop and (standalone only) the backing track can be adjusted. You can export the result, without the backing track, as a 24-bit/48kHz WAV file, and import that into a DAW.

Personal Bias

Whether you're considering investing in dedicated amp and effects modelling software for the first time, or are thinking about adding to your existing arsenal, I highly recommend Bias FX 2 Elite.

Like some of the other current high-performance amplifier and effects modelling software, the re-engineered DSP core of Bias FX 2 and Bias Amp 2 can deliver impressively accurate facsimiles of the original hardware. And, assuming your preferred platform is supported (there are plenty of DAW plug-in options, fewer standalone offerings, and very few for smartphones), the choice between products comes down to a matter of budget, workflow and personal preference.

From a workflow perspective, there's plenty to commend it, whether using it alone or with other products in the Bias ecosystem. Indeed, since Bias FX 2 Elite's standalone, plug-in and iOS versions share a common layout and menu, and integrate with Bias Amp 2 Elite and Bias Pedal, all those programs now sit at the core of my studio workflow for electric guitar. I do own and still use other amp and effects modelling software, some of which sounds great, but Bias FX 2 Elite is where I now do the vast majority of my work, partly because of the quality of its emulations and sonic performance, but also because there's such ease and convenience of operation. Even at its full price it is good value for money but the other versions are more affordable still, and Positive Grid offer generous bundle prices too.

Finally, just as we were going to press we caught wind of something new and interesting that will be coming to Bias FX 2; we'll cover that soon.

ALTERNATIVES

IK Multimedia's Amplitube is the closest competition that's available for Mac, Windows, Android and iOS. If you're happy with plug-ins, and don't need smartphone or standalone operation, your options expand considerably.

£ Desktop: Elite £216; Pro £144; Standard £71; Expansion packs £57 each. iOS version free to download, with in-app purchases to unlock Standard (£13.99), Pro (£34.99) and Elite (£59.99) features, and expansion packs (from £8.99). Prices include VAT.

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PROFESSIONAL AUDIO HARDWARE AND SOFTWARE

JM Acoustic ORTF Microphone Clip

Microphone Mounting Bracket

When amateur recordists ask about simple stereo mic techniques, the most widely recommended mic array for novices is the ORTF arrangement. The spaced capsules gather inter-channel time differences in addition to the inter-channel level differences afforded by the angled polar patterns — the combination delivers a reliable, spacious, and professional-sounding stereo sound stage. However, rigging two pencil microphones with the required 170mm capsule spacing and 110-degree mutual angle is often fiddly and frustrating, especially as one of the microphones usually has to be raised above the other to allow the cables to be plugged in.

Thankfully, the age of 3D-printing has made it practical for designers to manufacture all manner of specialised microphone mounting brackets at relatively low cost. Designs can be created in a CAD system by enthusiasts and shared over the Web for printing at home, or by a commercial company. One such 3D-printing supplier is an American company called Shapeways, who have a subsidiary in the Netherlands. Look up 'custom microphone mounts' on their website and well over 300 different microphone mount designs will appear! Some are simple and basic, some quite ingenious, and some others will probably never work as intended... but the variety is astonishing.

I have recently been looking for a simple mount to make rigging an ORTF array quicker and easier and, after trawling the Interweb, I came across an elegant design by JM Acoustic, who also offer mounting plates for X-Y, M-S, DIN and NOS stereo formats, as well as acoustic spheres for omni mics, and various other handy mic-related contraptions.

JM Acoustic's ORTF mount comprises a flat plate with integrated mic clips on the top and bottom surfaces. Strategically placed holes allow mounting directly over a 5/8-inch mic stand thread, or suspension lines can be attached at three points. It is 3D-printed-to-order exclusively by Shapeways from a black nylon plastic in a matte finish. The mic clips are available with nominal sizes from 19mm to 22mm, or there's a 'universal' model for pencil mics of any size, secured with elastic 'hair ties', O-rings, or rubber bands. There are also options intended for specific mic models. I bought the 20mm clip version, and have found that there is enough 'give' in the plastic to cope with 21 and 22mm diameter mics, if fitted with care.

JM Acoustic recommend using the plate with a ball-joint adapter, so that the mics may be angled up or down easily from a vertical stand. That makes a great deal of practical sense already, but as all of my mic stands have the European 3/8-inch thread, selecting a ball-joint adapter with a 5/8-inch top thread also made fitting the plate easier. (I bought the Onstage MM01, but alternatives include the Triad M1-R and WindTech MA-1).

The JM Acoustic website lists the ORTF plate at £25.90 for customers here in the UK, which is very reasonable for a custom-made specialist device like this, although Shapeways' processing and shipping charges (plus VAT, our sales tax) brought the final price for a single unit up to £45.34. Tax and shipping differences make it slightly less in the USA ($44.77 in total). It arrived within eight days and the product is perfectly suited for my purposes and the cost is, for me, justified by the minimised rigging time and hassle. I have installed a pair of Rode TF-5s semi-permanently on the plate and can put up an ORTF stereo array in, literally, 10 seconds! *Hugh Robjohns*

£ £45.34 including VAT and shipping.
W www.jmacoustic.com
W www.shapeways.com

Bram Bos Woott

Multiband Compressor For iOS

I've reviewed a number of apps by developer Bram Bos over the last couple of years. These have included the absolutely brilliant Troublemaker and Ripplemaker synths and, more recently, a line of iOS effects processors including Kosmonaut and Perforator. All of these use the AU plug-in format, making it easy to integrate them into your workflow when using a DAW/sequencer host such as Cubasis, AUM or Auria Pro.

Bram's latest release is another iOS (AUv3) effects app called Woott. Suitable for iPhone or iPad running iOS 11 or later, Woott is described as a 'dynamic hype enhancer' and, as you might guess, this suggests we are in for a bit of not-so-subtle compression. Indeed, Woott's name is derived from OTT, which itself started life as a preset in Ableton Live's Multiband Dynamics effect before being released in a dedicated free plug-in by Xfer. The Ableton preset became a bit of a thing for electronic music producers, particularly in terms of the way it was used on the low-end elements of lots of dubstep mixes.

So, what is it that OTT actually does, and Woott attempts to emulate? Across three frequency bands, it combines both conventional downwards compression of your signal peaks (loud signals are reduced in volume) with upwards compression of the quieter sections of the signal (low level signals are made louder). If both processes are pushed to extremes, you can significantly reduce the overall dynamic range of your signal for a full-on, in-your-face result.

Bram's take on the OTT concept is beautifully presented in Woott. I tried the plug-in in both Cubasis and AUM, and it worked very smoothly in both. On the left of the UI is a set of dials for adjusting the up/down compression ratios, compression time response and the output gain for each of the three bands. The central graphical portion allows you to adjust the up/down »

» compression thresholds for each band. These are linked to the Squash control on the right, which dictates just how much breathing space (the Woott manual refers to this as the 'black hole') your dynamics will have before they're compressed in one direction or the other. You also get an overall limiter, make-up gain control and an Amount (wet/dry) knob. On the far right, you can access the bundled presets, and a Save button lets you create your own.

It takes a little time to get your head around exactly what Woott is doing, just as it does with OTT. That's not because you can't hear what it's doing — it's designed to be the polar opposite of 'transparent'! — but more in terms of getting a feel for how the various controls interact, and how the compression is changing both the dynamics and tonal character of your audio source. If you want your EDM to slam though, Woott is a powerful tool. It can bring out subtle details in sounds, level unruly sounds, and make things monstrously loud. And anything can be a target, whether bass, drums, synths or a full mix. Woott is designed to be abused.

If your iOS music-making focuses on electronic music styles then, at this pocket-money price, Woott is another Bram Bos no-brainer. It won't be to everyone's taste, and it's certainly not intended as a conventional compressor, but if you like to make elements of your mix or, indeed, the whole mix sound loud and proud, Woott does 'over the top' with some considerable style. *John Walden*

£ £3.99.
W www.ruismaker.com

Soundevice Digital Voxessor

Vocal Processing Plug-in

Sold through and marketed by United Plugins, and supporting all the usual Mac/Windows formats, including AU, VST3 and AAX, Voxessor provides a user-friendly toolkit for improving the sound of spoken voice. It's intended to appeal to anyone doing podcasts or voiceovers but, as with most outwardly simple plug-ins designed by this company, what goes on under the hood is based at least in part on a spectral algorithm. We're informed that although it's been optimised for the spoken word, Voxessor can also be used on conventional sung vocal tracks, and that it may well appeal to some rap artists too. The process also includes dynamic EQ and a proprietary de-essing algorithm, as well as a more conventional compressor, a gate and an automatic EQ matching system that works by analysing a few words of your own voice.

The large Mode pointer control at the top of the window covers the range for both male and female voices, and this determines what part of the spectrum will be emphasised — this has to be set by ear. It is as easy to dial in as EQ though, even if exactly what goes on behind the scenes is not divulged. Designer Boris Carloff explained that the processing references archetypes of good-sounding spoken voices at different timbres, presumably to help massage the input in the right direction. An Intensity control, which essentially sets the wet/dry mix, adjusts the amount of processing, so at this stage only the Mode and Intensity controls need to be adjusted. There are level controls for the input and output, and a large metering section is located to the right of the window. An Autolevel button provides a more constant volume.

For a more personalised solution, you can turn to the analysis section, which is directly below. Once you have pressed the button and recorded your voice sample, the orange Match knob determines how much processing will be applied. Again, this isn't a straightforward match EQ but instead works at a spectral level, referencing those 'ideal' voices. What it adds is also related to the setting of the Mode dial. We're on more familiar ground with the lower sections, as the de-esser has just two controls: one for frequency, one for the amount of attenuation. The compressor and gate are even simpler, with just one knob apiece.

It's difficult to quantify exactly what is going on inside Voxessor, but the end result appears to be a balancing of the chest frequencies, adding more weight and richness where needed, plus a massaging of the mids and highs to provide a sense of presence and intelligibility. While similar results may be obtainable using combinations of other plug-ins, you'd probably spend a lot more time finding the ideal settings, and for anybody making podcasts using budget mics, the speed and ease of use could be a big improvement. Match mode provided a good fix for my own voice, and the only thing you really need to be wary of is adding too much processing — you do still need to listen, and let your ears decide what is enough. The de-esser works well, especially for something with two-knob simplicity, while the no-frills one-knob compressor adds a welcome bit of weight and consistency. Best of all is that if you aren't sure whether or not the plug-in will work magic on your own voice, you can download a demo and try it for yourself. *Paul White*

£ £111 including VAT.
W www.unitedplugins.com

Win!

Lauten Audio Eden

Worth £2669

The Lauten Audio Eden LT-386 is essentially three distinct tube microphones packed into a single, hand-finished enclosure. Developed in close collaboration with Grammy Award-winning producer/engineer Fab Dupont, the Eden makes picking the right microphone for any sound source as easy as flipping a switch. From the modern, tight and bright sound of Cardi B's 'Bodak Yellow', to the smoky vintage velvet of David Crosby's 'Lighthouse', the Eden offers an analogue abundance of mix-ready sounds in a single microphone.

Three unique signal paths are readily available for effortless sonic selection. The Gentle option is warm and intimate with a vintage character. It's strongly reminiscent of a German classic, but with a bit more clarity (and at a third of the price). The Neutral selection is classy and balanced with a smooth top end, which is perfect for acoustic instruments like acoustic guitar and piano. The Forward selection is bold, present and modern, but never thin or harsh — this is the quintessential sound of today's pop, hip-hop and R&B, but with a balance and density that many recordings lack.

Extending the versatility of the Eden is a three-position low-cut switch, which actually sits first in the signal path. Set to flat, the Eden Allows the full-frequency spectrum to be passed directly to the multi-voicing circuitry, the tube and the output transformer, for maximum density and harmonic saturation. The mid position gently reduces murkiness and overbearing low frequencies, while also resulting in a cleaner, more contemporary sound. The max selection shifts the timbre of the microphone entirely, bringing a clean, modern focus and articulation to the mid and high frequencies, and is ideal for use in combination with the Forward voicing selection for pop and R&B vocals.

The capsule of a microphone captures music's most meaningful moments and converts them into electricity with unforgiving results. Lauten's precision-crafted 38mm capsule is an original design, and one of the largest microphone capsules in the world — tuned by hand, Lauten say, to capture "the spirit, soul, and wonder of a performance with unprecedented authenticity and realism". An added benefit of this unique design is that it excels on harsh and traditionally challenging sound sources.

Each Eden is designed and crafted in Silicon Valley, California, by the Lauten Audio family — and this month, we're giving you the chance to win one! To enter the competition, simply visit the URL shown, and answer the questions there, by Friday 3rd December 2021. Good luck!

To enter, please visit:
https://sosm.ag/lauten-comp-1121

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W www.lautenaudio.com


LAUTEN AUDIO
LT-386

SAMPLE LIBRARIES

In Session Audio

World Percussion Creator

Kontakt Instrument

★★★★★

Nick Magnus reviewed In Session Audio's Taiko Creator and Drumatic Creator in the May 2019 and June 2020 issues of *SOS* respectively. The company are now back with a third instalment of the 'Creator' series — World Percussion Creator — and, given just how impressed Nick was by the first two titles, the bar is set very high.

As the title suggests, the emphasis here leans more towards percussion but there are still plenty of drum sounds included to provide a complete sound palette for creating ethnic rhythms. Sampled instruments therefore include bass drums, cajon, congas, darbuka, djembe, dunun, frame drums, snares, surdo, udu, agogo, clave, pulli, rattle, tambourine, woodblocks and a whole list of others. As with the other titles in the series, these core sounds are supplemented by a collection of sound-designed and auxiliary elements spanning booms, slams, cymbals, chimes and various others. Individually, the sounds are absolutely top-notch and, with seven velocity layers and seven round-robins at each velocity layer, there is plenty of scope for a realistic and dynamic performance.

Nick described the underlying Kontakt-based engine in some detail when reviewing Taiko Creator, so I'll not dig into the details here other than to say that, spread across the three main screens — Global, Groups and MIDI — it integrates a very powerful and flexible feature set into a user-friendly and intuitive user-interface. The ability to design your own ensembles (and sub-groups within them), as well as options to humanise the performances at a group level, are brilliantly executed.

As with previous releases, you can choose to purchase the 'core' version (with 10 kits and 20 suites of MIDI grooves) or a version with expansion packs that triple the number of kits and grooves. For the purposes of the review, I had access to the expanded version, and it's an impressive array of sounds and MIDI patterns to explore. With the ability to reassemble the sounds into your own custom kits, and easy MIDI drag and drop to create new patterns within your MIDI sequencer, there is plenty of mileage to be had.

Each of the preset kits loads up with a custom background graphic, and the references to ancient civilisations and wildlife give a strong suggestion of the sonic direction In Session Audio were aiming for. That said, there is a huge palette of sounds here, and while trailer composers might prefer Taiko Creator, World Percussion Creator is immensely flexible; it can easily go from delicate to bombastic.

There are some sample library/virtual instrument developers who just seem to always be on the top of their game. Having used a number of their products over the last few years, I'd certainly put In Session Audio within in this group. World Percussion Creator is a fabulous option for media composers, delivering top-quality sounds in a UI that is superbly designed. Given what's on offer, it is also very competitively priced. Highly recommended. *John Walden*

$139.99

www.insessionaudio.com

Soniccouture

Orchestral Chimes Collection

Kontakt Instrument

★★★★★

Under the careful direction of sound designers James Thompson and Dan Powell, Soniccouture have assembled a distinctive range of sample libraries since the company started in 2005. A fascination with unusual 'niche' instruments such as the Skiddaw Stones (a 19th Century Cumbrian lithophone) and the enchanting Cristal Baschet (a 'crystal organ' constructed of metal and glass rods, part of the Glassworks collection) has given the company a reputation for innovative esotericism, but their latest release focuses on more traditional tuned percussion: tubular bells, crotales and glockenspiel, collectively known as the Orchestral Chimes Collection.

Confusingly, our Transatlantic cousins refer to the first instrument as 'chimes', but I must say I prefer the more descriptive British name. Though these long suspended metal tubes look nothing like church bells, they do a remarkably good imitation, to the point where a single strike can induce feelings of reverence. The model sampled here is the Musser Symphonic Chimes M661C, played with felt, rawhide and plastic mallets over a C4-G5 pitch range. The rawhide hits sound the most natural to my ears, and the 'Space' effects preset (based on an IR taken from London's All Saints Hall) adds a fabulous large concert hall reverb to this classic orchestral sonority.

Crotales are sets of small, tuned thick bronze or brass discs resembling miniature cymbals. Sampled over a C6-C8 range, the library's Paiste Symphony crotales produce exquisitely pure, ethereal high-pitched chimes which sound like signals from the edge of the universe. In a similar vein, the library's lovely glockenspiel (a Yamaha Orchestra Bells YG-250D model, F5-C7) is ideal for adding twinkling, penetrating stellar chimes to arrangements.

Both instruments were played with plastic, brass and rubber beaters: the plastic option creates a classic full tone and the brass is more ethereal and 'fairy bell'-like, while the rubber beaters produce a softer attack. Delving into the built-in effects, I was able to create a fabulous shimmering keyboard patch by adding a Leslie cabinet-style rotator effect and All Saints Hall reverb to the glockenspiel's rubber hits.

Surprisingly, no sound-design patches were included: apparently these went missing in the initial release version for some reason. Soniccouture's generative sequencer/arpeggiator tools were also omitted from the review copy on the grounds that they tended to sound messy when applied to such resonant instruments, which is understandable. Happily, both

omissions will be rectified in a v1.1 update, which should have been made available by the time you read this.

I can unhesitatingly recommend this pristine, deep-sampled collection. Studio-recorded from two mic positions at up to 17 velocities, the instruments have an ultra-realistic dynamic response and are presented in a choice of 'free ring' and sustain-pedal modes, the latter giving control over their natural long decay time. Orchestral Chimes Collection (13GB) requires Kontakt 6 or the free Kontakt 6 Player version 6.2 or later. *Dave Stewart*

£89

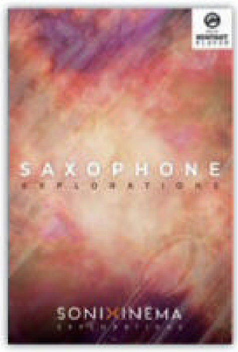
www.soniccouture.com

Sonixinema

Saxophone Explorations

Kontakt Instrument

★★★★★



Do you ever go on a musical exploration? And did you ever consider taking a saxophone with you? If the answer to both of those points is 'No', then you're in luck, because Sonixinema have been pretty productive over lockdown, working remotely with saxophonist Jay Reynolds on a project with saxes at its core.

Saxophone Explorations is not your traditional instrumental sample library. If you're looking for 'Careless Whisper' or 'Baker Street', you're firmly in the wrong ball park, but what we do have is a really eclectic and diverse set of samples, which take the instrument to the outer edges of its technical and tonal capabilities.

The first couple of patches begin with arpeggios, played both fast and slow. It's clear that they were performed to a metronomic pulse, but it is regrettable that there is no DAW tempo matching; then again, it could be argued that the whole point of these samples extends further than the norm. In this case, the samples are presented over a two-and-a-half-octave range and played, not triggered. This means that while the playing is very strong, there are moments of welcome imperfection. Once the sample reaches its conclusion, the remaining harmony drifts off into the ether, thanks to some rather beguiling back-end effects.

Part of Jay's sonic-sax arsenal is a Eurorack rig, complete with modules providing delay, resonance and even granulation. This doesn't appear to come into play straight away, but there are a number of controls, both immediately available and from the FX panel, that allow you to give the exploration some colour.

Apart from the MIDI assignable expression control for instrument volume, a see-saw fader drifts between Subtle and Vivid, controlled via MIDI CC 1 (mod wheel). This moves from a centred tone to a more ambitious and treated sound, with interplay from other nearby controls labelled Blur and Gloss. These effect the tone of the instrument quite radically, in an almost filtered sense, although there is a traditional subtractive filter available from the FX page as well.

Many of the included patches take the sax to the edge of its comfortable capabilities; there are howls, buzzes, shakes and overtones, all of which make fantastic starting points for creative sound design, and there is plenty lurking under the hood to provide sonic interest.

By way of contrast to the Philip Glass-like arpeggios which form the first patches, the presets labelled Harmony build up chords over much longer time periods. The Dominant Long patch is an excellent example, as a dominant chord manifests slowly. Add in further note triggers across its metronomic time zone, and you immediately create a sax-coloured backdrop of enormous proportions. There is also a Gating page which allows for rhythmic control alongside the DAW.

Saxophone Explorations is a surprising find! It's full of terrific sax content, with true creative potential away from the more usual instrumental constructs. *Dave Gale*

£69

www.sonixinema.com

Sonic Atoms

Zaria

Halion Instrument

★★★★

Hosted by Steinberg's free Halion Sonic SE 3.4 (or later) sample player as well as Halion 6 and Halion Sonic 6, the Zaria sound library is based on evocative female vocal sounds performed by soprano Roksana Korban. Some follow the traditional sustained-note route, some presented as gentle vocal stabs and others treated to become ethereal pad sounds. Sonic Atoms describe the genre as cinematic and ambient, which seems as good a description as any. This is not a traditional vocal library in that it is more about texture and ambience than creating realistic lead melodies, though the samples are musically pitched and playable as such. Halion's free Sonic SE 3.4 player is easy enough use and supports AU, AXX and VST 2/3 hosts as well as operating in standalone mode. Effects can also be added within Halion Sonic SE, though in this instance, the sounds are so rich that piling on further effects will often be too much.

Zaria's patches are categorised as Pads, Atoms and Performance Programs, the latter comprising layers, spread across the keyboard: low organic pads, vocal performances split according to key range offering multiple articulations, and a high pad. Articulations are independently switchable directly from the GUI or by key-switching. Reversed versions are included to add some textural variety. The Pads are all vocal-based and seem to have quite a lot of reverb baked into them, so you don't need much in the way of additional processing to make them sound good, although in some situations you might want to apply some low-cut EQ to thin them out a bit. They all sound organic and mostly calming, though some include just a hint of chaos or disquiet. Being vocal-based, there is a certain family sameness to them but the designers have done what they can to coax as much variety as possible out of the pad section.

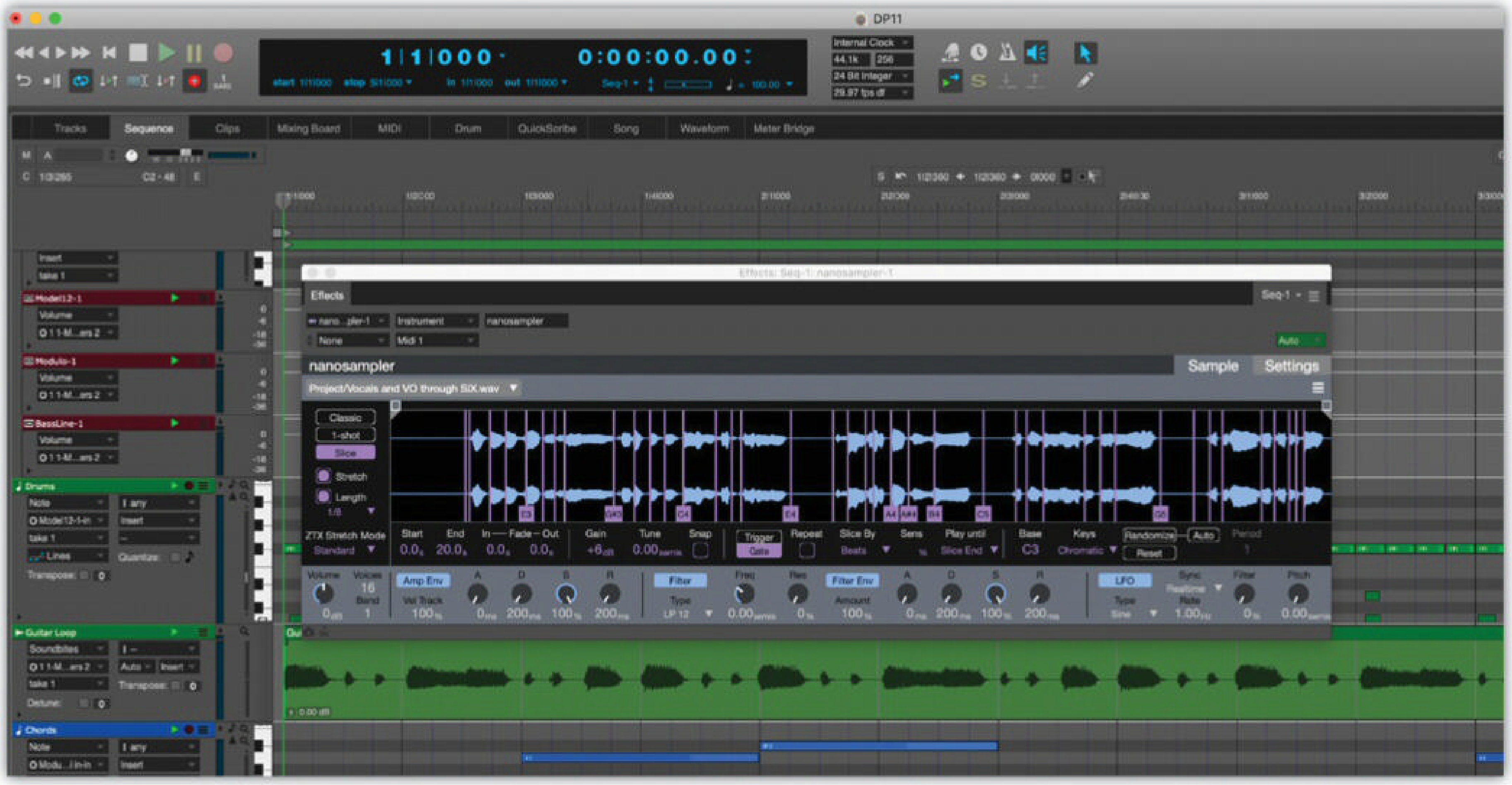
The Atoms patches cover more discrete melodic sounds based on different vowels, breaths and other utterances, plus some sustained vowel sounds that have a useful degree of movement built into them. Single Atom programs also offer articulation control.

With so many vocal sample libraries around, you might wonder if Zaria can bring anything new to the table, but it actually provides a palette of lovely sounds that would slot well into soundtrack, ambient or chillout compositions with the shorter Atom sounds offering something a little different from the usual choral 'oohs' and 'ahhs'. And it does so at a very modest price. In some cases I would have appreciated some drier sounds so that I could use my own processing, but in the main the sounds are very polished and instantly usable. *Paul White*

£58.99

www.timespace.com

[Audio examples of this month's libraries are available at www.soundonsound.com.]



Nanosampler 2 adds features like sample slicing and integrated ZTX time- and pitch-shifting.

Nanosampler 2 offers a wealth of creative sample playback options.

MIKE LEVINE

Among the many new features MOTU unveiled in DP11 is Nanosampler 2.0, a significant overhaul of its bundled sample-playback instrument. While the original Nanosampler had one playback mode, now there are three, including one that slices up your sample. The new version also integrates DP's powerful ZTX time-stretching and pitch-shifting technology, giving you more control over how the samples play back.

Shape Shifter

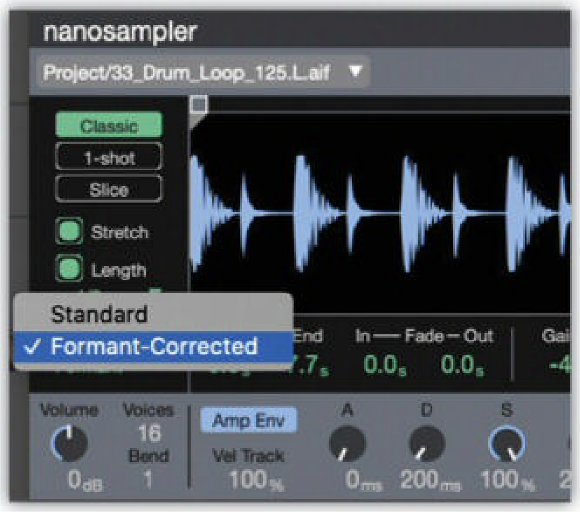
When you first open Nanosampler 2, you can tell from its GUI that it's significantly different from its predecessor. It's long and rectangular rather than square, which allows for a much larger waveform display. Its design and colour scheme are darker and more contemporary.

MOTU also replaced the knobs for Sample Start and End, Loop Start and End and Fade with sliders directly above the waveform display, making those controls more visually connected. The bottom row contains the synth settings: the amplitude envelope, filter, filter envelope and LFO. The available parameters for those remain essentially the same as before, albeit with a different layout. But what is new is graphic editing. If you click on the Settings Tab, Nanosampler switches to an alternate screen that features graphic editors for all the synth parameters.

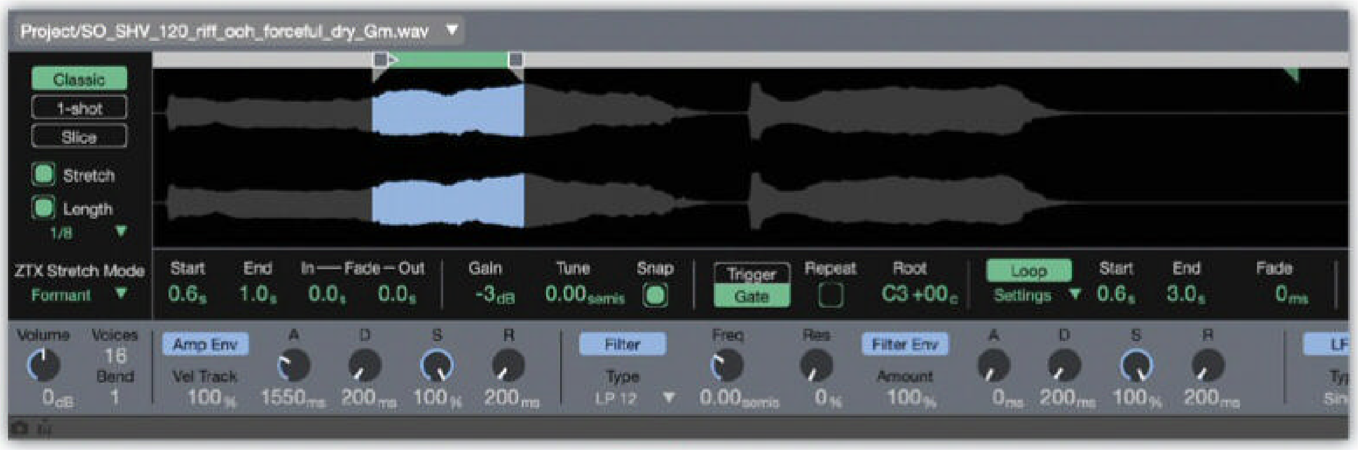
Time To Stretch

To fully understand the capabilities of the new playback modes, it's helpful to first look at a couple of powerful new features in Nanosampler 2. One, called Stretch, uses ZTX time-stretching algorithms and is available in

Clicking on the Settings tab brings up this graphic editing screen for the synth parameters.



Once you turn on Stretch, you can choose which ZTX algorithm to use.



The green slider above the waveform lets you set loop in and out points, and fade-ins and -outs.

all three playback modes. It allows you to control the length of samples as they play back, keeping them sounding natural when transposed from their original pitch.

Unlike in the previous Nanosampler, where every time you transposed a note up, it played faster, and down it played slower, Stretch makes them all play at the same length. When you click on the Stretch button, you get a couple of additional options. One is to choose one of two ZTX Stretch Modes: Standard or Formant-Corrected.

Either will even out the length of the sample at any MIDI note. Choosing ZTX Formant-Corrected makes the pitch transpositions more seamless, especially for vocal tracks, avoiding the dreaded 'Mickey Mouse' and 'Monster' effects.

When Stretch is on, you can also access a setting called Length. It lets you constrain the sample duration by a selectable rhythmic value. According to MOTU, it's especially beneficial on drums and percussion. For example, if you've loaded a two-bar loop, you can set the duration to two bars to quickly conform the loop to your project timeline and tempo.

Triggering The Gate

The other significant new feature available in Classic and Slice modes is the choice of Trigger or Gate, which govern playback behaviour. In Trigger mode, when you play a note, it lasts for the entire length of the sample (subject to your start and end point settings). You can shorten playback by choosing a small duration with the Length parameter of Stretch mode.

In Gate mode, the sample is cut off as soon as you release the note on your keyboard. But you also get a couple of features, Repeat and Loop, which are not available in Trigger mode.

With the Loop button pressed, a green slider appears above the Waveform display and lets you set the boundaries of the loop, overriding the settings of the Sample Start and End sliders. You also get a movable arrow to set a crossfade for the loop.

If you turn on the Repeat function, the sample will retrigger automatically when it gets to the end for as long as you hold the key down. If Loop is on, Repeat is also subject to the Loop setting.

Modus Operandi

The three new playback modes are Classic, 1-Shot and Slice, and they take Nanosampler into new territory, in particular with the Slice mode. Classic mode operates similarly to the original Nanosampler or any basic sampler. When you load a sample, it's mapped to the MIDI keyboard so that each note triggers a different pitch. It's useful for creating instruments from single notes, or in any situation where you want to transpose the pitch of a sample.

With the Stretch feature turned on and ZTX set to Formant-Corrected, Classic lets you turn a single sample into an instrument while maintaining consistency in the sound on different pitches.

The second playback mode is 1-Shot. It's similar to Classic, except it only works in Trigger mode, and the polyphony is constrained to one voice. When you play a note on your keyboard, the sample will play until its end unless you trigger another note. While not as flexible as Classic mode, its monophonic operation and unique playback behaviour give you additional choices.

In 1-Shot mode, playback is monophonic and samples play in their entirety.

Slice, the third playback mode, is what really differentiates this version of Nanosampler from its predecessor. It slices your sample into pieces and maps each one to its own MIDI note. You have several choices for how to slice the sample. The default is Beats, which cuts based on the transients it detects. You can adjust the Sensitivity of that detection to control how many slices get created.

MOTU included several other slicing options. The Mensural setting lets you specify equal-length increments based on rhythmic values like eighth notes, 16th notes and so forth. If you choose Division, you can select how many equal-length slices you want Nanosampler to create. And the final mode, Manual, lets you alter the slices created in any of the other modes by dragging their dividers.

Whenever you're in Slice mode you can audition any slice by clicking on the square at the bottom where you see the note name. If you're feeling adventurous you can turn on the Auto Randomize feature, which will randomly change all the MIDI note assignments for the slices.

By The Numbers

Whichever mode you're in, the area directly below the waveform display offers

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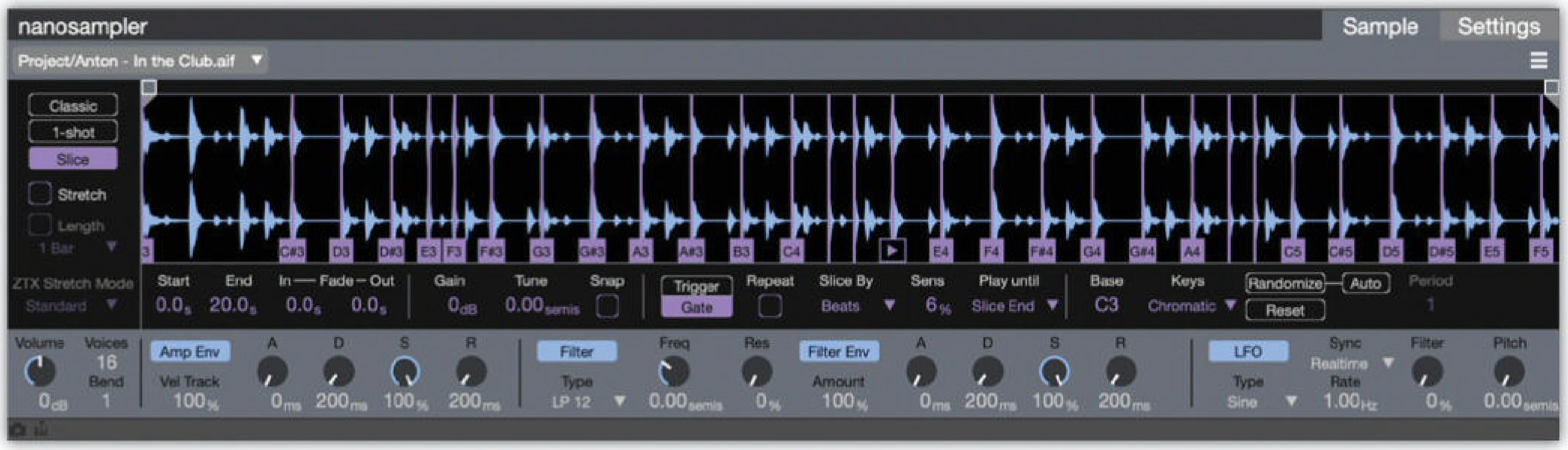

You can audition a slice by clicking on the arrow under its note name.

numerical control over quite a few key parameters, including Gain, Tune, Root Note, and more. The parameter choices vary somewhat by mode. You can also adjust the Start and End and Loop Fade settings numerically from here. Most of the fields let you click-drag them up or down, or type in a value.

Next to the Tune parameter is a button called Snap. If you activate it, Nanosampler tweaks your start and end points to keep them from creating clicks.

Do It Yourself

With all its new capabilities, Nanosampler is a flexible instrument with surprising power for a sampler that holds only one sample at a time. You can use it for everything from dance music and hip-hop production to sound design and experimental music. The following are just some examples of what you can do with it:

Make a tonal sampled instrument: Turn on Classic mode and drag and drop a tonal sample into Nanosampler's waveform display. If the sample you imported has more than one note, or a note with more than one pitch, use the Start and End sliders to limit it to only one pitch.

You can try both Trigger and Gate mode, but for tonal instruments, you'll probably prefer Gate, because its behaviour when you release a note is more like a synth or other virtual instrument.

Press the Stretch Mode button and set the ZTX Sample mode to Standard or Formant-Corrected, depending on which gives you the sound you're looking for. If you want to make the sample last for a particular duration, turn on and set the Length parameter.

Make a drum or percussion instrument: With the addition of Slice mode, you can now take your favourite drum or percussion loop and map its sounds on your MIDI keyboard.

Find a loop you like, import it into Nanosampler and turn on Slice mode. Use the Beats option to create slices from the transients. You'll probably want to use Trigger mode so that the samples ring out rather than getting cut off.

If you're slicing up a drum loop, you'll have to hunt around a bit to find the MIDI notes corresponding to the kit parts you want to trigger. If you're using a loop of a single percussion instrument, such as congas, you'll find usable notes almost everywhere.

Create vocal chops: You can make vocal chops (short vocal samples) in a couple of different ways. One is to take a vocal phrase with at least a few notes in it and drop it into Nanosampler. Turn on Slice mode, and it will create a variety of notes and syllables or words (depending on the length). You may have to adjust the boundaries of some slices to get the sounds you want. Start in Gate mode and try turning on Stretch and the Format-Corrected algorithm for ZTX.

Alternatively, you could use Classic mode to make a vocal chop instrument. Set the Start and End points to select one note of a vocal sample. Use the Loop feature if you need to smooth it out. You might want to adjust the Root parameter so that the notes you're playing match the pitches of the MIDI keyboard.

You can fortify your vocal chops or almost any samples you're triggering in Nanosampler by inserting effects after Nanosampler in its mixer channel. Compression, reverb, delay, overdrive, it all depends on the sound you're going for. Because Nanosampler doesn't have built-in effects, adding additional plug-ins can make a massive difference in how your samples sound when triggered.

Without question, Nanosampler 2 is significantly more powerful than its predecessor and opens up a huge range of creative possibilities. It's one of many good reasons to upgrade to DP11.

Adding effects after Nanosampler can add additional energy and interest to your samples.

JOHN WALDEN

We explore how Cubase's multiband tools can help you master your own mixes.

In April 2021, I described a simple, single-band signal chain for DIY mastering in Cubase. The approach followed the principles outlined by Ian Shepherd in his *SOS* Mastering Essentials video series (https://sosm.ag/mastering-essentials), and used stock plug-ins that are available to all users of Cubase 11 Pro, Artist and Elements, as well as a couple of third-party freebie metering plug-ins. As mentioned in that article, though, Cubase Pro also boasts an impressive collection of multiband processors — so in this article, I'll consider the pros and cons of using these powerful tools for our DIY mastering signal chain. To accompany the article, I've prepared a number of audio examples, which you'll find on the *SOS* website at https://sosm.ag/cubase-1121.

Join The (Multi)Band

In my previous article, I followed Ian's keep-it-simple approach and used a signal chain comprising Cubase's StudioEQ, Compressor and Limiter plug-ins. I also placed an instance of Brickwall Limiter at the end of the chain, just to catch any stray peaks, and used freebie VU and loudness meters to monitor the final level of my in-progress master.

For my multiband signal chain, I substituted StudioEQ for Frequency 2 and Compressor for Multiband Compressor, added instances of Imager (for stereo image adjustments) and Quadrafuzz 2 (for saturation), and kept Limiter and Brickwall Limiter in place. For comparing the spectral content of my master with my reference, I used Voxengo's CurveEQ, which is bundled with Cubase Pro. SuperVision provided loudness metering alongside Klanghelm's VU Meter (my favourite third-party option).

Compare The EQ

As in the April 2021 column, my first step was to use my VU meter and SuperVision's Loudness module to adjust the channel gain in the Channel Settings dialogue, aiming to get the level of my raw mix to average/peak levels around 2-3 dB below Ian's suggested final targets (-11dB on my VU meter and -10 LUFS for short-term loudness).

I then used CurveEQ to compare the overall spectral balance of my as-yet unprocessed mix and a reference track I'd chosen. Inserted on any track, this plug-in can capture an EQ spectrum of the audio that's passing through; this spectrum can then be loaded into another instance of the plug-in, allowing direct comparison between two (or more) signals. CurveEQ can also calculate an EQ curve that makes the signal (in this case, my track) match the EQ spectrum of a reference track. I did not actually use CurveEQ to apply any EQ changes — Instead, I made my EQ moves manually using Frequency 2 — but it is a useful visual guide if you wish to nudge your own track in the direction of a reference.

Voxengo's CurveEQ plug-in, bundled with Cubase Pro, allows you to compare the EQ spectrum of your mix with that of a reference track.

In this case, comparing the frequency curves encouraged me to experiment with small boosts centred at 60Hz, 2kHz and 7kHz, while also applying small cuts at 125Hz, 1kHz and above 10kHz. I also applied high- and low-pass filters to tidy up the extreme ends of the frequency range. In keeping with Ian's advice, the changes were subtle but, to my ears at least, they both added weight and brought clarity to the mix, and I retained these EQ settings as the foundation for my subsequent multiband experiments.

Two's Company

The majority of Pro 11's multiband plug-ins offer four bands of processing. The new Imager plug-in allows you to specify exactly how many bands you wish to use but in the older Multiband Compressor and Quadrafuzz 2, you cannot completely remove the unwanted bands; you can only disable them. Starting with a two-band approach, my Multiband Compressor and Quadrafuzz 2 settings are shown in the screenshot. In an ideal world, I'd have the same band configuration as in Imager, with a low band and a high band, split at around 200Hz. As it is, the two 'unwanted' bands in both plug-ins have been centred in the low-mids and made as narrow as the GUIs permit. It's a modest, pragmatic workaround, but there is therefore a narrow frequency range between about 200 and 300 Hz that isn't being processed in this configuration.

To arrive at the settings shown, I adopted the same approach as in the April 2021 article, but on each of the bands. So, for example, for the compression, I set moderate attack (20ms) and ratio (2:1) values and then simply dialled in the threshold until each band was applying a maximum of 3-4 dB of compression. For Quadrafuzz 2, I soloed each band, selected the gentle Tube mode (Tape worked equally well when I tried it) and increased the Drive control until the effect was clearly audible, before gradually dialling it back; I wanted to 'hear' the effect but without it being too obvious. I used the two Imager bands to narrow the stereo image below and gently widen it above 200Hz, and also checked for mono playback compatibility. Finally, I used identical settings for the Limiter and Brickwall Limiter plug-ins as in the April 2021 article.

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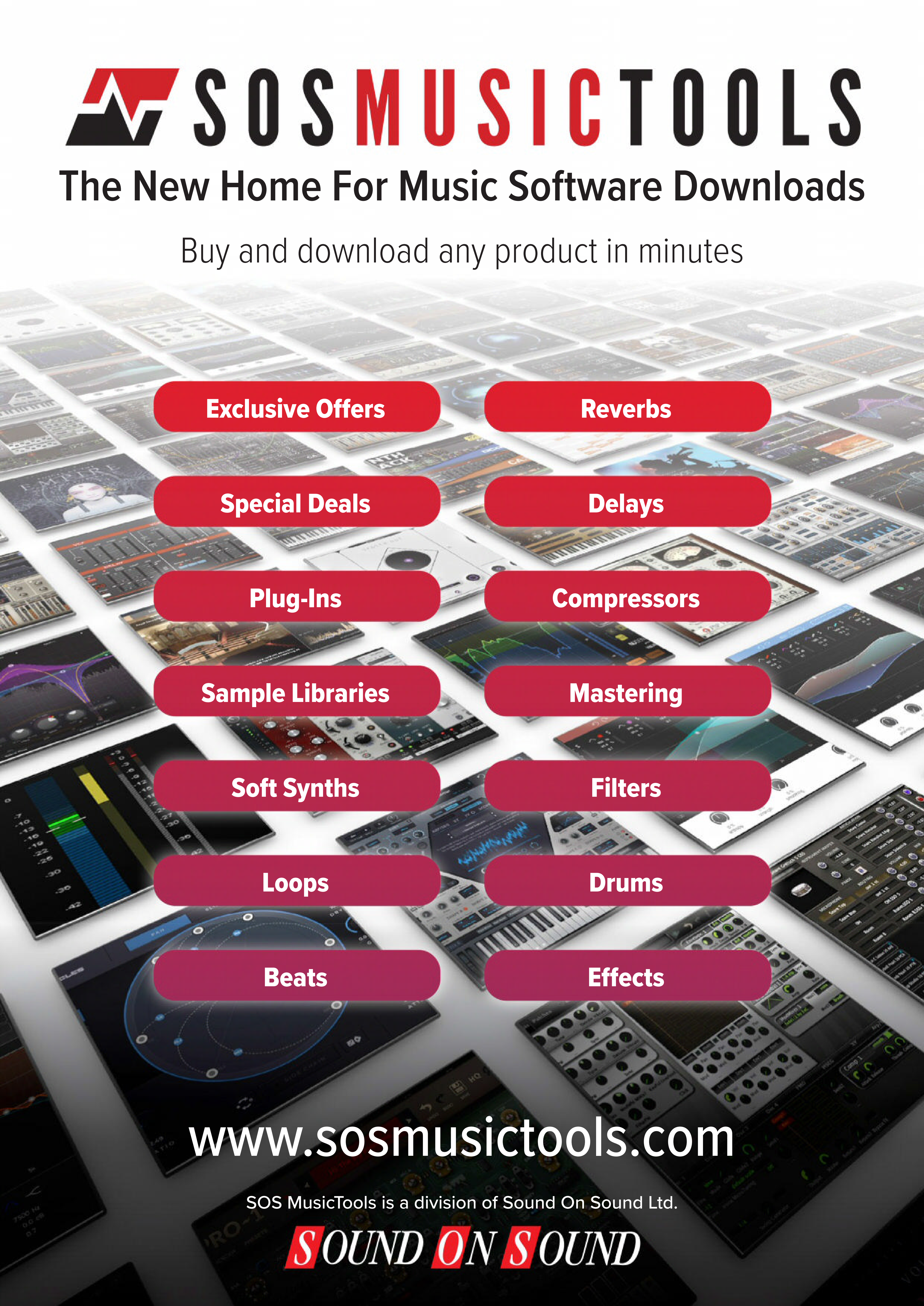
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The settings used for the key plug-ins in my two-band DIY mastering signal chain.

» In comparison to the unmastered mix, this two-band mastered version is louder, but even with the perceived levels matched it seems to have a punchier low end, less muddy mids, a crisper high end and slightly greater stereo width. The differences are subtle, which hopefully suggests the processing hasn't been overdone, but to my ears at least they seem beneficial: the two-band mastered mix has more impact.

Three, Four, Testing

If we start by sticking to the same keep-it-simple principle, moving up from two bands to three or four doesn't involve much more than deciding what crossover points to use and configuring the required settings within Multiband Compressor, Imager and Quadrafuzz 2 for the new bands. For my three-band version, I used crossovers at 200Hz and 5kHz, while for the four-band version the values were 200Hz, 2kHz and 10kHz.

Whether the differences are audible in

More bands means greater control over fine details, provided you can avoid abusing the processing power available!

a specific case or not, in principle, splitting our compression across more independent bands ought to produce a smoother overall result, as each band is focused on, and only triggered by, a narrower range of frequencies. Peaks in the low mids should not, for example, trigger compression in the high mids, or vice versa. That said, there's nothing to stop you making further adjustments in the different bands: you might, for example, experiment with higher compression ratios in your low band (for tighter control of the bass levels) or low mids (for keeping the mud at bay) if you feel your mix requires it.

However, using more bands for Imager and Quadrafuzz 2 is only really necessary if you wish to make use of different settings in the additional bands. I've done this with both plug-ins, applying gradually more stereo widening in each of the mid/high-frequency bands in Imager and, in general, applying a little more saturation in the lowest and highest bands than within the midrange in Quadrafuzz 2. Used in this conservative fashion, these extra bands feel to me like I've got just that little bit of extra control over details and the amount of 'mastering fairy dust'.

If you wish, you can easily construct alternative mastering chains using some of the other plug-ins bundled with Cubase Pro 11, but applying the same principles. For example, you could use an instance of Squasher to provide your multiband dynamics processing, or use Frequency 2's dynamic EQ options to combine multiband EQ and compression in a single plug-in. Just for fun, I've included audio examples based on both those approaches, but I'll leave a detailed discussion of those tools for another day.

Mixed Up Mixes

So, what's my own takeaway from this little experiment? First, while sensible (ie. subtle) single-band mastering in Cubase Elements and upwards can easily give a solid mix a good nudge forwards, I think the benefits of multiband mastering are obvious. This might simply be the ability to control the dynamics of your low end more firmly, without squashing your mids and highs. But the option to add different degrees of stereo width and saturation sweetening to your high-mids and highs provides useful additional control over the end result. And while you can do the same thing with third-party tools, Cubase Pro 11 users have plenty of scope on this front using just their DAW's stock plug-in collection.

Do note, though, as Ian Shepherd makes clear, that less is generally more at any single stage in your processing chain. It's easy to get lost wandering through the options presented to you when working with more bands, and thus to take your mix backwards rather than forwards. You must focus on what the material needs, not simply what your tools can do to it! Bear in mind, too, that splitting audio into frequency bands introduces phase shift which may be audible, especially when you do it repeatedly.

Finally, don't forget to audition the audio examples. As a small twist, I've randomised the order of the multiband mastered examples so that, if you want, you can audition them 'blind'. After you've decided which you think is most effective, check the small print at the end of the audio file description for the running order!

There's more to Reason's new sampler than meets the eye...

SIMON SHERBOURNE

Reason 12 is now out in the wild, with two major new devices: the Mimic performance sampler and the second-generation Combinator. I'm itching to talk about both, but let's take them one at a time and get stuck into Mimic this month.

Reason already has a few sampler type devices. NN-19 is the original sampler, which has been around since version 1.0 and is related to classic rackmount samplers like the Akai S1000. It's designed for playing back keyzone mapped samples, complementing the Redrum sample-based drum machine and Dr Rex loop player. Later on this was joined by NN-XT, which is essentially the same type of instrument as 19, but with the ability to layer samples and configure more performance controls and modulation. These samplers, supported by the multi-device patching and macro control features of Combinator, provided an engine that could happily power most traditional sample library content.

The NN devices are more like workhorses than fluid, creative samplers like, say, an MPC, or the Simpler device in Ableton Live. And that's where Mimic comes in: it's a sampler where you can quickly drop, trigger and mangle individual samples. The Grain instrument covered some of this ground in Reason 10, but as the name suggests it's focused on granular resynthesis. Mimic brings the same immediacy to regular sample playback and chopping. But while it's simple, there's a bunch of interesting and inspiring ways you can use Mimic; so I'm going to speed through the basics and then get to some ideas and tricks.

First Impressions

Mimic has a similar layout to Grain, with a full-width waveform display at the top, and various modules below for settings, modulation and effects. It's actually a fair bit simpler and more approachable, without the advanced envelope designer and modulation grid that's been a standard part of many Reason Studios instruments.

Samples can be dropped in from the browser (or desktop), or imported

Screen 1: Mimic's Slice mode and Play Thru features are the fastest way to chop samples in Reason.

from the familiar sample loader, which also features a real-time sampling button. Samples load into the eight slots, selected above the waveform. The slots are used in slightly different ways depending on the mode chosen from the top left of the panel. In Pitch and Slice modes, the slots are selected and played separately, one at a time. In Multi Slot mode, the slots are laid out on the keyboard and can be triggered simultaneously, but not pitched. The layout is repeated on each octave. Multi Pitch mode works much like a traditional sampler, with the slots layered on the keyboard where they can be zoned into note ranges.

Modes & Stretches

Samples dropped into an initialised Mimic will default to playing in Pitch mode, where you can play the sample across your keyboard. C3 will be the original pitch, but this root key can be adjusted (look just above the waveform and to the right). An auto-detected suggestion is shown here, which you can accept by clicking Set. In Multi Pitch mode the key settings are independent for each slot, as in fact are all other panel settings.

Separate from the mode is the Stretch type. The default is a traditional varispeed, which Mimic calls Tape mode. As well as following key pitch (and root note), there are independent Speed and Pitch controls, which in Tape mode are essentially doing the same thing. In all the other Stretch modes, Pitch and Speed are independent. Three time-stretch modes are provided. Advanced is a general purpose algorithm that is incredible on most material. As we'll see when we get into creative tips, it really extends the range of what you can do with a sampler. Melody is tuned for monophonic sources. Vocal is optimised for vocals (and speech) and has a formant shifter built in, plus a Fixed Pitch mode, which we'll come back to. Finally, there's a Granular mode, giving you a fairly basic granular synth.

Slicing

As the inventors of the ReCycle loop format, sample chopping/slicing is part of Reason Studios' heritage, and there's always been a REX player device in Reason. However, if you wanted to chop a sample yourself rather than import a pre-sliced file, it's not been that straightforward. You needed to edit the sample in an audio track, then export out to another device like Kong or Dr Rex (we covered this process in the April 2017 edition).

Mimic brings a simpler and more spontaneous approach with its Slice mode. This uses transient detection to automatically add slice markers, with a Sensitivity control in the Slices panel to the right of the unit. There's no real-time manual dropping of slices possible, but you can move slices directly in the waveform display.

Screen 2: The Advanced Stretch mode is so transparent you can slow samples down to a near stop.

Double-clicking a slice marker deletes it, and double-clicking an empty area adds a slice. You can also adjust the overall start and end points, which is very useful after direct sampling. Clicking and dragging up or down in the waveform display zooms for finer edit control.

Slices are assigned to keys from C1 upwards. Playback defaults to Poly, so if you slice up a drum loop into individual hits you can play the patch like a drum kit. You might want to add some velocity in the amp section, and perhaps an envelope. For re-sequencing the slices into a new pattern — MPC style — try switching to Mono Retrig mode. There's no tempo match function, so if you're using longer slices with rhythmic contents you'll need to fiddle with the Speed manually. A fantastic feature (that got special mention in last month's Chopping feature) is the Play Thru mode. With this engaged, when you trigger a slice, the sample will continue to play past the end of the slice. This is really effective for smoothly playing drum breaks without gaps.

Tips & Tricks

The rest of Mimic's sections are modulators and some processing. I'm going to leave them for you to explore so we can look at some creative Mimic tricks...

Speed Control

The Advanced time-stretch on Mimic is so transparent that it opens up some interesting possibilities. My favourite is to use Mimic like a granular synth, but without actually going granular. Samples can be slowed right down to get long, slowly evolving textures. You can even slow to a complete stop and get a short cycle that still sounds like the original sample. Add modulation to the Speed and things get even more interesting. With Speed very low or stopped, try modulating with an LFO. Playback will sweep between forward and reverse. It's like granular but much smoother.

Start Mod

Experimenting with the start position parameters yields interesting results. First there's the Global Position switch, which forces legato-played notes to trigger sample playback at the current position instead of retriggering. For synth samples that have an inherent envelope this provides a classic synth legato response. For a sampled phrase it lets you add polyphonic harmonies progressively.

The Start position can be modulated from numerous sources. With a longer, shifting sample, I tried selecting Random and Velocity as modulators. Each played notes is then tonally different, which can be dramatic or subtle depending on the sample. The Snap To Slices button can be used with this trick to quantise the Start position to slice points. Cleverly, you can set slices in Slice mode, then switch to a different mode like Pitch. The slices will then be used during Start modulation. This proved great for random percussion, or for weird chopped-up vocals.

The manual has a cool suggestion for Start mod: creating velocity zones. This is an ingenious way to very quickly sample an instrument dynamically. Record in a source played with progressively louder articulation. Then slice it into zones, and assign Velocity to Start Position with Snap To Slices engaged. Voila: instant dynamic sample playback with no MIDI data programming.

Vocals & Speech

Perhaps the most fun I had with Mimic was playing with spoken phrases and vocal lines. The Advanced Stretch and Speed controls are once again great here, but try the Vocal Stretch mode and experiment with Mono vs Poly triggering and Global Start on or off. Fixed Pitch is where the real fun is. This forces the sample to play at the pitch of the MIDI note. It's similar to the Vocal Synth mode in the Neptune Pitch Adjuster, except that you're triggering the sound instead of it being on a track, and you can freely mess with Speed at the same time. Imagine a really transparent vocoder. You can go completely bonkers with this, or try using it as a fresh take on the ubiquitous Auto-Tune gloss.

Screen 3: Live triggering and re-pitching of vocals and speech in Mimic has huge creative potential.

JULIAN RODGERS

Create custom control mappings with EuCon.

I've been using an Avid S1 for a few months now. I love it, but I suspect like most people, I use it only for some of the things it's capable of. Fader moves and automation writing are the obvious uses, but while I know how to use it for plug-in control, I rarely do. The reason for that is that, even for bread-and-butter plug-ins like EQ III, I find it quicker to locate parameters on the plug-in UI than I do on the encoders. The Achilles heel of all hardware controllers is that the unfamiliar 'new' hardware controller has to compete with the very familiar mouse/screen combination to be learned in the first place, and I'm sure I'm not alone in either being too lazy or too busy (or both) to give the controller a fair chance to compete.

Something I always loved about the ICON series of controllers was how the EQ and Dynamics sections mirrored the layout of the EQ III and Dyn III plug-ins. I thought at the time that tying such a big-ticket item as a five-figure control surface to something as ephemeral as a plug-in UI was a gamble, but I liked the concept so much that I bought McDSP's Channel G plug-in because it used the same layout to stay consistent with the ICON. In the case of EQ III, I needn't have worried: I still use it pretty much as my go-to EQ. The Dyn III plug-in on the other hand is more of a distant memory, and has been replaced by newer and better options.

While the layout of plug-in parameters has always been consistent on EuCon controllers, the recent EuCon 2021.6 release has made it possible to create custom parameter maps on the S1, S3 and Dock, using the new Custom Knob Sets feature. The advantage of being able to set up your own custom assignments is that if you make the decisions on what goes where, and also on what gets omitted, then learning the parameter assignments is completed by default.

EQ Mapping

The implementation is smart and deep. I'll illustrate this using EQ III and an S1, though of course the process can be applied to any plug-in.

There is a pressure which I'm sure I'm not alone in putting myself under when learning a hardware controller and that is to try to use it for everything. I remember when I used to use a D-Control, regularly using the surface to instantiate plug-ins. This was possible but not a very efficient process, involving as it did navigation through long lists using encoders and very truncated names. It was a lot quicker to instantiate the plug-in from Pro Tools using a mouse and keyboard. The point is that the choice between mouse/keyboard and a hardware controller isn't a binary one. You can and should use both for different tasks.

With this in mind, I decided that as EQ is the task I feel is best suited to hardware control I'd set up a Custom Knob Set for the seven-band EQ III, which rather than replacing the mouse/keyboard interface, would complement it.

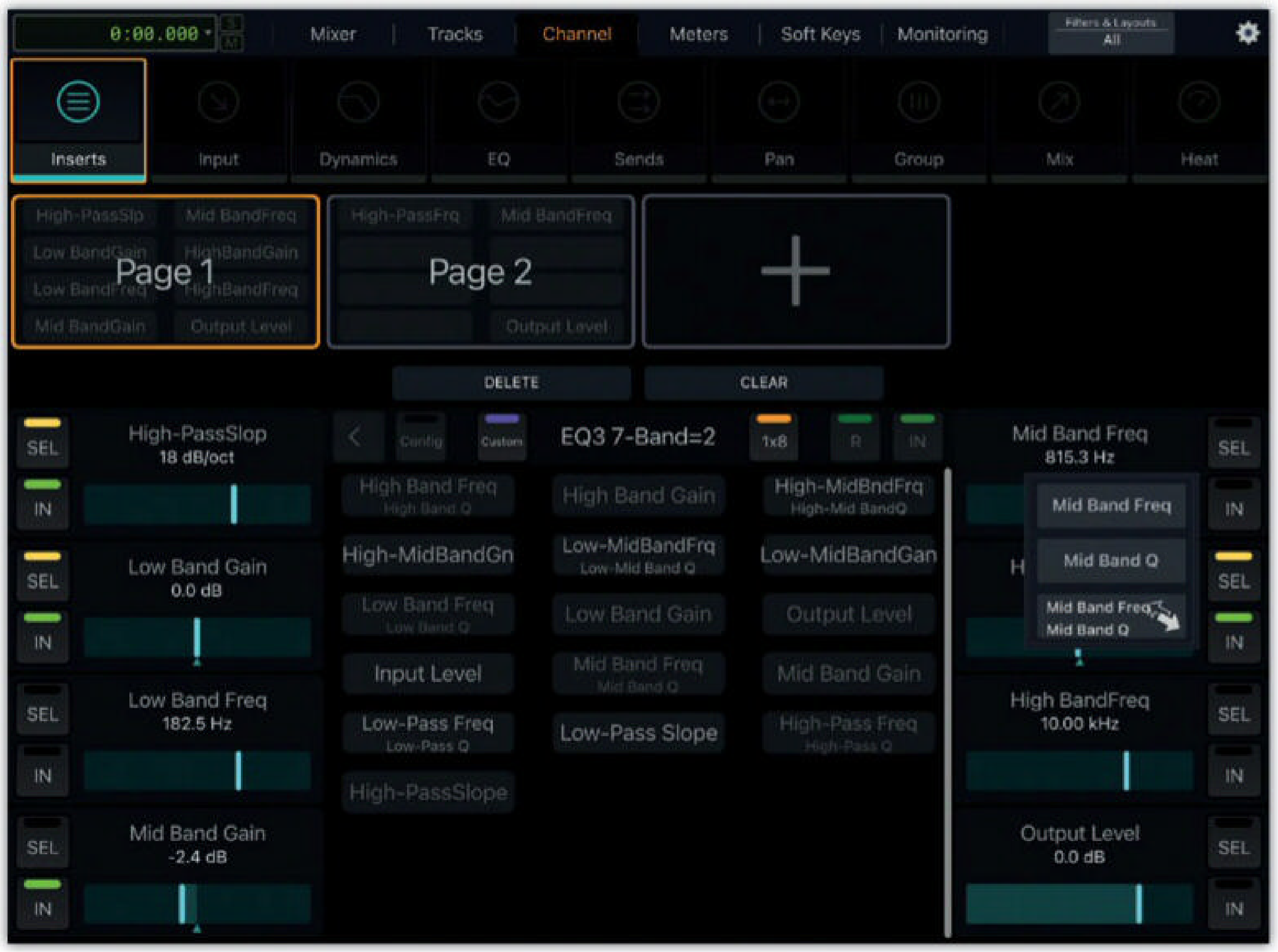
For general mixing tasks a simple three-band channel EQ, of the type found on simple analogue consoles, is all I need. Something like this:

- **Filter:** A fixed high-pass filter, switchable in and out from a button. Typically, console filters are set at 80Hz with a slope of 12dB/octave. I prefer 18dB/octave with a 75Hz turnover frequency.
- **Low Shelf:** The default settings for EQ III, 100Hz and the default Q.
- **Mid Band:** Set to 1kHz, the default setting. Actually most of the mixers I used in the console days had two mid bands but I chose to only have one in this custom map so I could have all my EQ on a single bank of eight encoders on the S1. Of course all seven bands are available in the plug-in, and even if not included in the custom map, they are still mapped in the default map. More on this later.
- **High Shelf:** The default is 6kHz, I've set it to 10.

What I want from this arrangement is for me to be able to instantiate an EQ from my S1 with my preferred settings ready to go. To do this I need to set up and save a user default and change the Setting Preference in EQ III to load my user default rather than the factory default. This is a useful and overlooked option in AAX plug-ins, and one which I first used when I finally snapped at having the 'Beneath The Waves' patch load up every time I instantiated Xpand! 2. I saved an electric piano preset as a user preset, set it as my user default, and then set the plug-in default to User Setting. Now when I instantiate Xpand! 2 I get a Rhodes rather than that ponderous synth pad...

Using the same technique I changed a couple of settings in the EQ III default patch, specifically changing the HPF to 18dB/oct and 75Hz and the frequency of the high shelf to 10kHz. With this setting saved as user default and the setting preference set to User I get my desired settings each time I load an EQ.

You can reduce your reliance on the mouse and keyboard by setting up a default set of plug-in parameters to load up every time you instantiate a plug-in.



The Avid Control app lets you drag and drop parameters to any of your available physical controls.

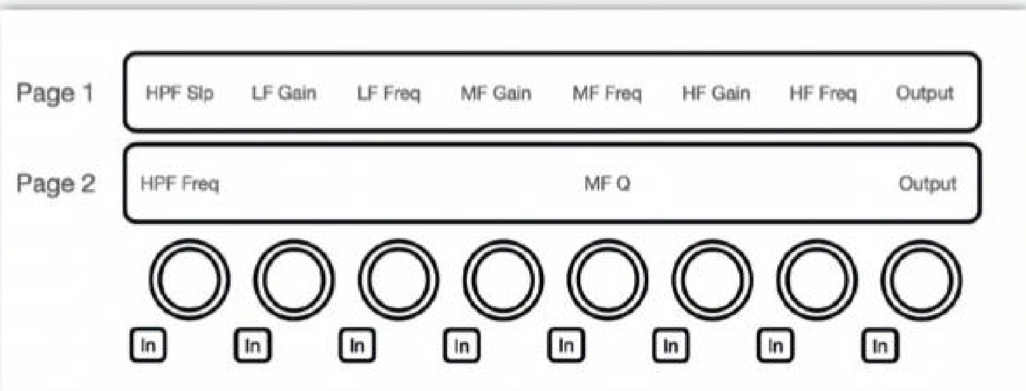
Another thing which needs to be done, if it isn't already, is it that the default EQ needs to be set to EQ III in the Pro Tools Preferences. You can set a default EQ and Dynamics plug-in in this tab and they appear at the top of the plug-ins list but if you have an S1 they also can be created from the S1.

Plug-in parameter mapping is straightforward. In the channel view in the Avid Control app you select the plug-in you wish to map and hit the Custom button. From the custom map page, parameters can be dragged and dropped to the available encoders. Multiple pages can be mapped and some parameters have multiple sub-parameters, for example Frequency and Q in the EQ. An option to choose between them is offered in the setup page.

A frustration when it comes to this example is that in spite of mapping the output to encoder 8 on both pages, there is no way to toggle the bypass on EQ III from the surface (though if the plug-in you're mapping has a power switch as part of the UI, you can assign that). Another limitation is that, in the default map, although the buttons and the encoders control different parameters, the function of the button is tied to the function of the encoder. In the case of the HPF, to assign filter in/out to the button on encoder 1 I had to assign HPF slope to the encoder. There is no way to assign Frequency to the encoder and In/Out to the button on the same layer.

Custom & Default Maps

Custom Maps are an alternative to the default maps which have always existed. These are created by the plug-in manufacturers and, while largely consistent between plug-ins of the same type, the flexibility offered by custom maps allows so many more options. For example, in the Custom Map I set up for EQ III I chose to have a limited set of controls mapped so that almost everything I wanted was available from a single set of eight encoders. I then built a second page of assignments which offer a couple of extra options, specifically the mid band's Q and the HPF's turnover frequency. The Output is mapped to encoder 8 on both pages, so it is always available for level matching.



The author's Custom Map for Avid's EQ III plug-in.

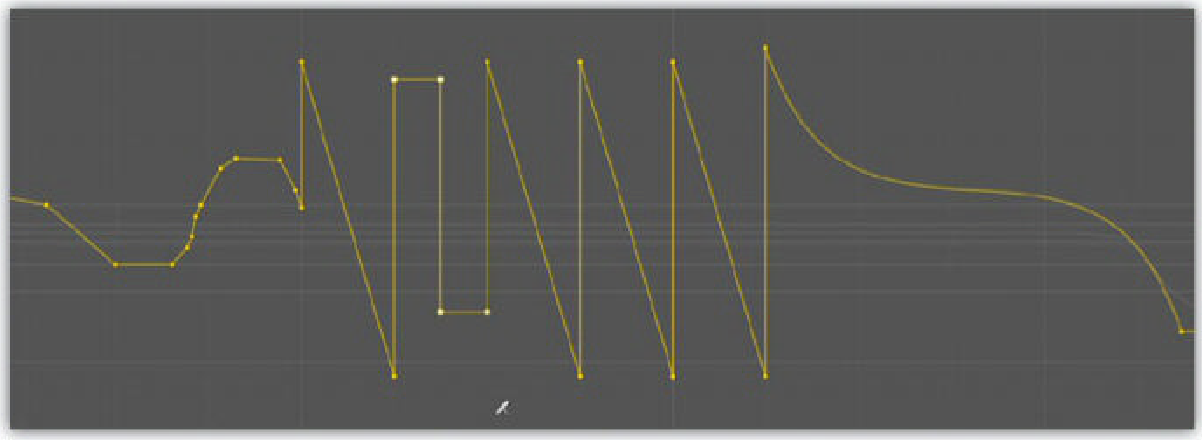
I could have mapped a second mid band but, returning to my earlier point, the plug-in UI is always available for more detailed work, and for EQ and Dynamics there are two knob mappings available on an S1 courtesy of the dedicated EQ and Dyn buttons.

All inserts in a session are available via the Inserts button. Select a track, page through the inserts slots and press the encoder to spill out the parameters across the 8 encoders. Unless a custom map has been set up this will be the default assignment map as set by the plug-in manufacturer. If the insert is an EQ or a dynamics plug-in then it is also accessible via the EQ or Dyn buttons, and a different custom map can be set up for this mode too. In this way, for EQ and dynamic plug-ins, it is possible to access either the default map, or up to two custom knob assignments. Because of this added flexibility I'm perfectly comfortable with my choice to have such a reduced number of parameters mapped in my custom knob assignment. Fewer controls make it easy to navigate, but if I need more control I can still have it via the default map, which is still available.

There has been a lot of progress in the development of EuCon, and while there are still occasional surprising gaps in the provision of features, these are being addressed and EuCon is still far and away the best way to control Pro Tools from hardware. This EQ example is only one of the options, another obvious choice is building custom assignments for complex plug-ins with large numbers of parameters, such as reverbs. I do look forward to the Bypass button in the plug-in header being assignable though...

ROBIN VINCENT

Make your tracks come to life with automation.



You can input automation in a number of ways: by adjusting a parameter in real time, using the freehand draw tool, or by selecting wave shapes.

Automation helps you realise your creative vision by baking in those movements, changes and layering that you couldn't manage to pull off in performance. But more than that, it gives you an opportunity to resuscitate a dying mix and breathe new life into your tracks. It's what DAWs were born to do. So let us charge up the defibrillator, stand well back and unleash the tools of automation.

What Is Automation?

Automation is the process of recording or writing in the movements of knobs, sliders and other parameters onto a track so that when the track is played back all those movements happen as if you're moving them by hand. It's the magical sense of watching the faders on the console dance about, or hearing virtual instruments modulate, or effects being thrown around. Pretty much anything that has a value can be automated in Studio One. It keeps things fluid, it moves more faders than one mouse or many fingers could possibly manage and it's essential for creating mixes with space and vitality.

There are two main ways of automating a parameter: you can record the movement of that knob, fader or other control in real time, or you can draw changes in with the Paint tool.

Let's automate a channel's volume: the fader in the mixing console. We'll do the real-time fader movement recording version first. Find the channel you want to automate in the Console. At the bottom of the fader, you'll see a box that says Auto: Off. Click on that and select Write. (There are some other options there which we'll come to later.) The word Write is now lit up in red and the channel is ready to record any movements you make to the fader. You don't have to put your project into Record for this; automation recording can be done during playback. Set your project playing and start moving the fader. You'll see the movements being written as lines and nodes over the top of the audio or MIDI clips in the arrange window.

Shortcut tip: to switch between seeing the automation data and seeing the clips on the track, press A on your keyboard to turn Automation view on or off.

If you then rewind to the beginning and press Play, you can watch in confusion as your automation is overwritten with a flat line. This is a very common mishap that happens because the channel fader is still in Write mode. So, once you've finished recording automation it is vital that you go back to the track and swap Write for Read, which is helpfully highlighted in green rather than red.

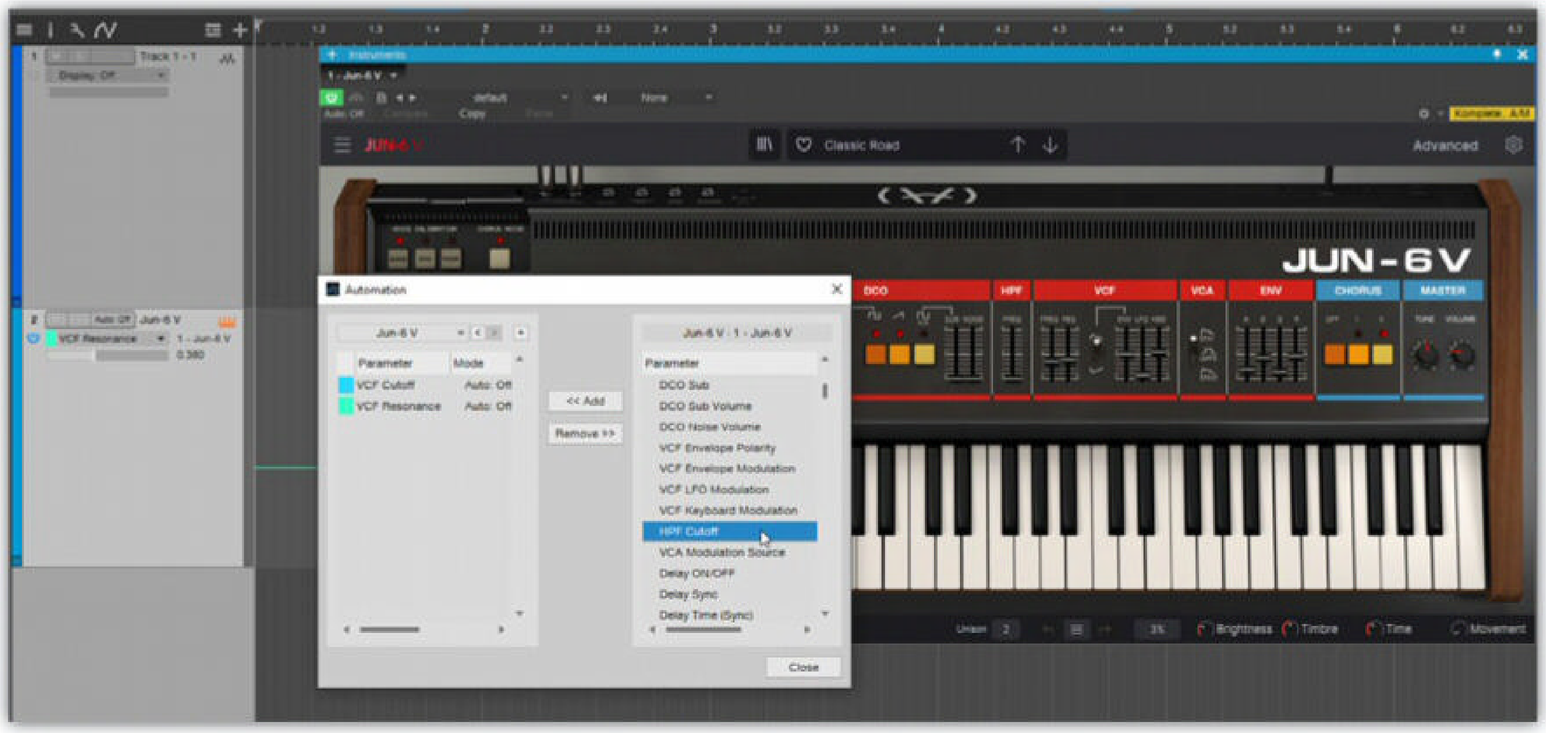
Pro tip: Write mode is dangerous and has a habit of inadvertently erasing previous automation data when trying to record a new parameter. See below for why you should use Latch and Touch modes in most cases.

After going back and re-recording your fader automation and then switching your channel to Read, you can now watch as the fader moves by itself, following exactly the movements you recorded. That's it, that's exactly how automation recording works for any parameter of the mixing console, plug-in or a software instrument.

Writing Automation

To get a perhaps more precise and considered movement you can draw the automation directly on the track. Select the track in the arrange window and press A to switch to automation mode, or click the zig-zag icon at the top of the track list to do the same thing. If you haven't done any automating up to now, it will say Display: Off. Click on that and you'll see a list of the currently available automatable parameters. For an audio track this usually means Volume and Pan. Choose Volume. The clips on the track will grey out and you'll see a single blue line representing the current static volume.

You have



To make a plug-in parameter available for automation, you need to drag it into the left-hand column of the Add/Remove... window. Alternatively, if you automate a parameter using the mouse in real time, it will be added to the list of automatable parameters automatically.

a couple of choices here. Hover your mouse over the line and it turns into a hand with a pointy finger. You can now drag that line up and down to change the volume, and in doing so it creates a node where your mouse grabbed the line. Release the mouse and pull on the line again and the first node will remain static while the new node you've just created can be moved, with a straight line connecting the two. More clicks create more nodes, and you can generate a mountain range of automation lines. Alternatively, you can grab the Paint tool and draw freehand whatever shape or movement you want. This is also how you can edit automation after you've recorded it, whichever way you did it initially.

Adding Parameters

If you try writing some automation for an Instrument track, you'll find no readily available parameters, so you'll have to add some. This is also how adding plug-in parameters on audio tracks works.

In automation mode click on where it says Display: Off and select Add/Remove... You'll be presented with a window that lists all the parameters of the currently selected instrument or plug-in that's loaded on that track. Select the parameters you want and add them to the left column. They'll now be available to automate.

There's a faster way though, which is to move the parameter you want to automate while in Write mode. It then automatically gets added to the list of automatable parameters. Often this is far easier than scrolling through a list of synth parameters looking for whatever the manufacturer named that knob you want to move. To enable Write automation on an instrument you'll find Auto: Off on the top left under the Activate button. Set the project playing and move the knobs or sliders on the synth, and they'll get recorded and added to the list for editing. You can do this with multiple parameters and it all gets recorded. Don't forget to put it back to Read before you play back your automation.

Editing Automation

We don't always want straight lines between all the nodes, and Studio One gives us the option to use curves instead. If you hover your mouse on the line between two nodes, another node appears; when dragged, this will produce a concave or convex curve between those two points. If you hold Alt/Option it will give you an S-curve to play with.

If precise shapes are your thing, you can use the different wave shapes available under the Paint Tool to throw in sine waves, sawtooths and so on. You can also select a range of nodes and move all the selected automation at once, perhaps increasing the intensity of the parameter without changing the performance.

While seeing all the automation superimposed onto the track is rather nice, it's often helpful to be able to see all the automation displayed on individual lanes. This is always available by clicking the tiny zig-zag icon on the bottom left of the track header.

Advanced Automation

You can generate automation lanes quickly using the Alt/Option+A command, which creates a lane for the last-clicked parameter. With Studio One instruments you can also right-click the knob and select Edit Automation; it will add a green dot to it to show that it's being automated. For this to work, the Recently Touched option needs to be ticked in the Control Link Box that sits in the tool bar. This is a super-quick way of creating lanes for multiple parameters.

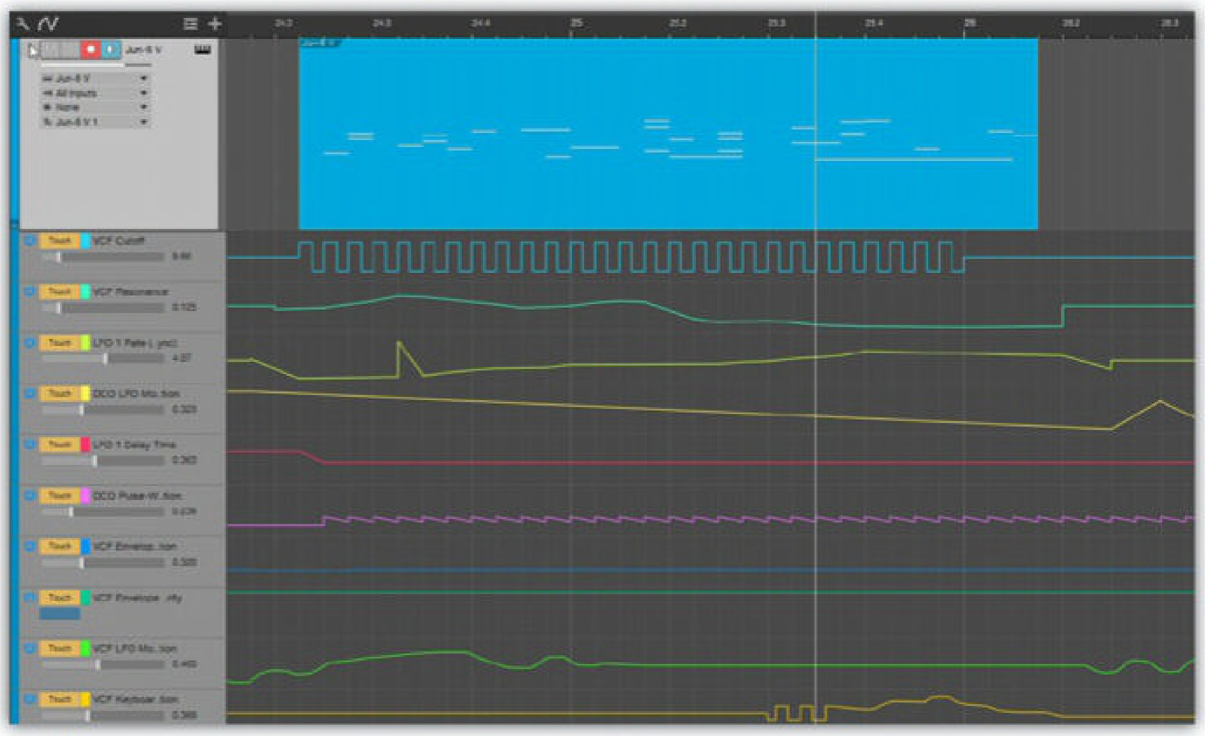
Studio One also can decouple automation lanes from the track, allowing you to drag them to their own track and put them together in folders for better organisation. You can drag automation lanes from the Control Link box. Do you see the hand waving between the 'A' and the current parameter? Grab that and drag that parameter's automation onto a new track.

You can use this feature to create an automation lane in the part rather than the track. If you open the piano-roll editor and drag the parameter into the Edit window it becomes an automation lane under the notes, like velocity. It doesn't take the track automation with it because automation within parts is kind of its own thing. The brilliant result is that when you draw in the automation for the part, it stays with the part: move, or copy and paste the part, and the automation moves with it.

Other Modes

Along with Read and Write we also have Latch and Touch automation modes, which are a lot less volatile. These are like combinations of Read and Write. While playing back, Latch mode will read and act on all the existing automation until you click on the parameter, at which point it will start writing and continue writing as if it was in Write mode. Touch mode will do the same, but when you release the mouse it will switch back to Read again. So, if you just want to make a change and not replace everything that's already there, Touch mode is the most forgiving. Touch is certainly the best mode to be in if you are looping a section and don't want it to be overwritten on the next loop. Latch is useful for picking up where you left off.

Both these modes are essential if you are automating additional parameters on playback, because being in Write mode will erase all of the automation you've got going on so far. Write mode is great for making lots of adjustments in one pass, but use it with care and never leave it on.



Clicking on the tiny zig-zag icon in the track header will display all active automation lanes for that track.

Managing your levels will help you get the most out of your plug-ins.

STEPHEN BENNETT

In the dim and distant days of analogue audio production, the limitations in the signal-to-noise ratio of contemporary equipment meant that engineers had to make sure that audio passing through each piece of equipment was as 'hot' as possible. This usually meant aiming for the VU meter bouncing around the 0dBu level in mixers, effects and tape machines. But that was until engineers found out that some equipment sounded wonderful when you exceeded these sensible levels! Things are very different in the digital realm, and it's taken quite a while for us to really understand what the 'best practice' might be when recording and mixing in the brave new world of 'in-the-box' production. Understanding how you might deal with gain at different stages of the recording, mixing and mastering processes is crucial — especially if you want to break any 'rules' that may exist!

No Gain, No Pain

It's important to be clear about the relationship between analogue (dBu) and digital (dBFS) levels, as these are not the same thing at all. While you can push an input signal into an analogue mixer to well over 0dBu until its electronic headroom is reached, if you exceed 0dBFS with an analogue-to-digital (A-D) converter, you'll get digital distortion — which is very nasty and usually unwelcome. There is no headroom over 0dBFS in the digital world and the signal is basically shorn off, or 'clipped'. If you want to play about with this kind of distortion, it's better to record cleanly and add clipping in post, where you have a chance to change things if they go wrong.

The very best A-D converters have a dynamic range (ie. the difference between the loudest and quietest signals they can represent) of around 120dB, but 24-bit digital recordings have a dynamic range of 144dB (16-bit is 96dB), so there's no need to keep your meters close to maximum any more to reduce noise levels. Giving yourself plenty of headroom when recording will give you greater flexibility when mixing.

Screen 1: Logic's Gain and Level Meter plug-ins are extremely handy for keeping an eye on your levels at all stages of the signal path.

At the other end of the signal chain, you should never push levels right up to 0dBFS (or above) at your stereo output channel, as this will clip the digital-to-analogue (D-A) converter. Perhaps perversely, some people appear to like the sound when this happens, but even if that's the case, you need to be aware that some playback systems, especially CDs, can have problems playing back audio that clips at 0dBFS. And if you're sending a mix out for mastering, you'll want to ensure your final mix peaks several dB below 0dBFS, to give the mastering engineer plenty of headroom to work with.

Gain Control To Major Tom

Once you have your nicely recorded audio, you'll want to mix it into a stereo or multichannel file. Some people like to keep all the levels of each channel peaking at or below the -18dBFS mark. One reason you might want to do this is that it makes it less likely you're going to overload the stereo output. It also can make sense if you're using virtual recreations of effects processors. The best of these emulate every part of analogue circuitry, including the input and output gain stages, which means that they will expect a certain amount of headroom. Peaking at -18dBFS will often achieve this, but there's no universal rule — you'd need to read each individual plug-in manual to find out if any specification is given.

If you're using no plug-ins on a channel, or are relaxed about your meter levels, you can run individual channels into the red with no audible ill effects — the internal headroom of Logic is, to all intents and purposes, infinite. But for analogue-modelled plug-ins, it's wise to treat 0dBFS as the equivalent of the hardware's ultimate headroom. Many plug-ins have input and output gain controls and meters, but for those that don't, I usually put a Gain plug-in (and Level Meter) before and after the software so I can control the input and output levels, as in Screen 1.

With software such as UA's 1176 plug-in, I'd feed it -18dBFS then use the 1176's input knob to adjust the level going into the compressor, but with the excellent Logic Compressor plug-in in Vintage mode (Screen 2), I'd use that plug-in's own input control to set the level into the compressor at -18dBFS for uncoloured compression, and use the same control to 'push' the input level up closer to zero on the plug-in's VU meter. »

Screen 2: Logic's Compressor is one of many plug-ins that responds differently according to the level you feed it.

I then use the output level knob to bring the level back to around -18dBFS on the channel.

The sonic effect of working with different input levels is quite noticeable in the UAD and Logic compressors, as well as in many of the best-quality modelled plug-ins. The UAD plug-in also has a variable 'headroom' level, from 4 to 28 dB, so you can experiment with how the input level affects the distortion and compression characteristics with differing levels of available virtual headroom.

Saturation

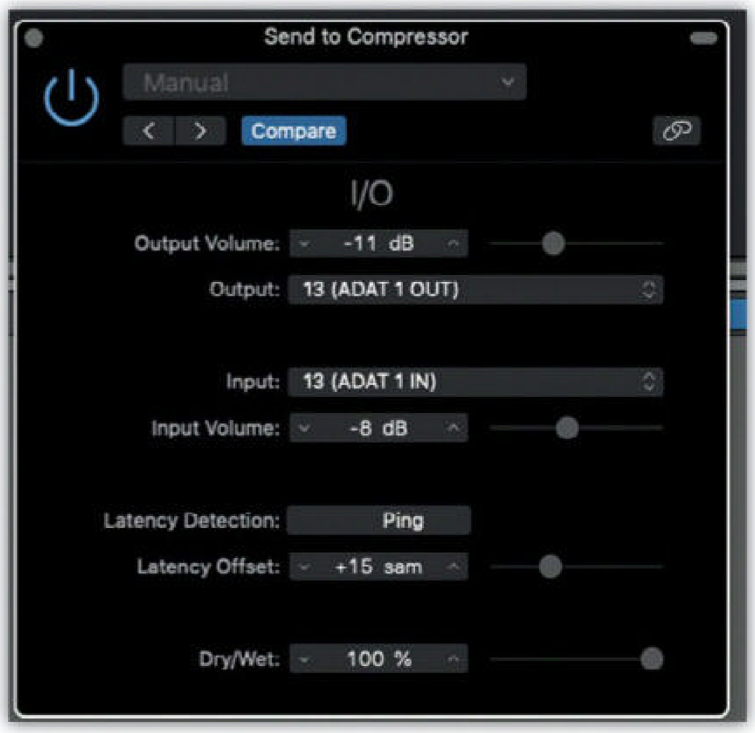
Distortion and saturation plug-ins are particularly sensitive to the input levels fed to them. Logic's Overdrive has a simple Gain parameter that increases the harmonic content of a signal passed through it, but it responds quite differently if you send it a signal peaking at -18dBFS or one pushing (or exceeding) the 0dBFS maximum.

Logic's Vintage EQs have no input controls, so the characteristic of any saturation effect is entirely dependent on the level of the signal fed to it — clean at or around -18dBFS, and more coloured as you push the level towards 0dBFS. The Tube EQ is modelled on thermionic valve circuits, and using a Gain plug-in before the EQ to increase the input level nicely emulates the effect of 'pushing' a valve hardware unit to saturation, but don't forget to add a Gain plug-in after the EQ to bring the output level back down to -18dBFS. Doing this is important as we humans usually perceive louder to be 'better', so you need to audition any processing without gain changes. Try doing this with all the EQ controls flat to hear how the audio is changed just by being passed through the plug-in. I find DDMF's affordable Plugindoctor (https://ddmf.eu/plugindoctor) useful in helping me visualise how software is affecting my audio, but you can get similar information using a mixture of Logic's meters and the free Voxengo Span plug-in (www.voxengo.com/product/span).

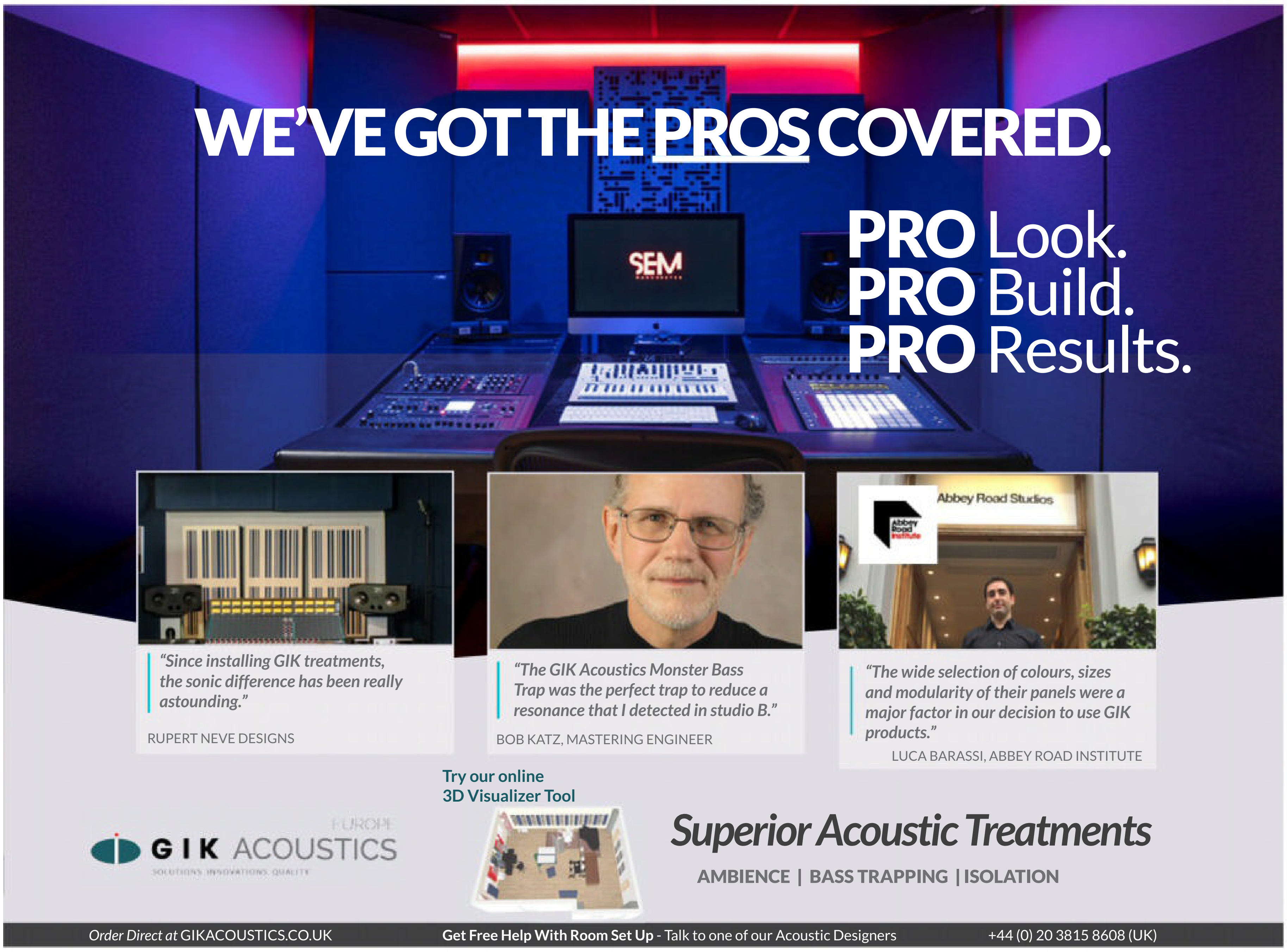
If you're working in a hybrid fashion, mixing digital processing with hardware, you also need to take care with the levels you are sending to your physical outputs. The I/O plug-in (Screen 3) helpfully has input and output level controls to help make sure you're sending the right amount of audio to your hardware to best make use of its particular properties. I often use Logic's Test Oscillator to send a -18dBFS sine wave or pink noise signal into my hardware to make sure my input levels are appropriate, and a Level meter to monitor the level coming back to the DAW after processing. I find Logic's level meters invaluable here, as they can be placed anywhere in the signal chain to monitor gain changes. You can drag the meter's limit line from -12 to -18 dBFS and treat this as the equivalent of an analogue reference level such as +4dBu. Many hardware processors have input and output gain meters, and the signal they need to be fed can depend on several factors — which were expertly covered by Matt Houghton and Hugh Robjohns in *SOS* September 2013: www.soundonsound.com/techniques/gain-staging-your-daw-software.

Breaking The Rules

Virtual effects have got much better over the years at modelling the nonlinear characteristics of analogue circuitry, and many respond in almost the same way as their hardware counterparts — as long as you remember that 0dBFS in the digital world is not the same as 'zero' in the analogue world! But you don't need to obsess about getting levels exactly right. Gain staging in Logic can help make processor emulations sound closer to their hardware inspirations, or can be used as a creative effect. At the end of the signal chain, we are often trying to get the maximum level possible to compete with other productions, but at the recording and mixing stages, careful gain staging and a little thought about the levels going to your plug-ins can make or break a track.



Screen 3: The I/O plug-in lets you adjust the output and input volume, which is invaluable when sending audio out from Logic to be processed by external hardware.



The Advertisers Index

SOS WHY I LOVE...

THE FUTURE

JYOTI MISHRA

One of the things that most irks me is when someone blithely says, “Oh, everything’s been done, there’s nothing new to discover.” This, from a member of a civilisation that still hasn’t established a colony on its moon, let alone another planet.

I’m pretty sure when the first two humans accidentally discovered vocal harmony, they patted themselves on the back and said, “Well, Glargh, that’s it. What else can there be after that? Pass the mammoth, please.”

It’s a twisted chrono-hubris that assumes that we, us, here and now, just happen to live at the pinnacle of all human creativity. Everything’s been done? No it bloody hasn’t. Stop being lazy, get off your postmodern posterior and engage your brain.

So, to get you going, seeing as I appear to be playing Toffler in a sea of maudlin Baudrillards, here are some things I would like to see in The Future...

When I used to record on a Fostex X15, reverse delay and reverb were simple effects to create. You just flipped the tape over, bunged the effect on the source, recorded to a new track and then flipped the tape back the right way. Wind forward to DAWs and you would think that there’s a plug-in that does that for you. But, dear reader, I have Googled ’til I’ve drooped and I haven’t found one. You can do it, but it’s a faff. Now, my genius is exceeded only by my laziness; as if I have time to do all that. Give me a plug-in that scrolls through the entirety of the track and then yields proper reversed reverb and delay. And I don’t mean an envelope simulating it, I mean the plug-in does all that reversal wotsit inside itself and then plays the result.

We are overflowing with plug-ins that mimic vintage greats. But could someone, somewhere, please make me a Roland JX-3P plug-in that can listen to my old cassettes of saved patches and sequences and load them? Surely it can’t be that difficult for a DAW to pass audio to a plug-in which then reads it? Oh, and while you’re at it, please do the same for my TR-707, Jupiter 8 and MC-202. Ta!

I want a physical modelling plug-in but... more inclusive? I want it to model the instrument, the player, their body, their life history, their current mental state and maybe even the room they’re in. So, if I give the plug-in a middle eight to compose and enter ‘Morbidly obese middle-aged Indian synth-botherer with a dodgy knee and only the vaguest grasp of modes playing a Rogue with scratchy pots’, then I can go and have a cup of tea, and my solo will be, as we say, *le kiss du chef.*

In the immortal words of Robert Browning: “Grow old along with me. The best is yet to be...”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