electronic MUSICIAN

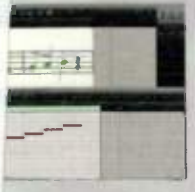
RECORD • PRODUCE • PERFORM

MIX FASTER, MIX BETTER!

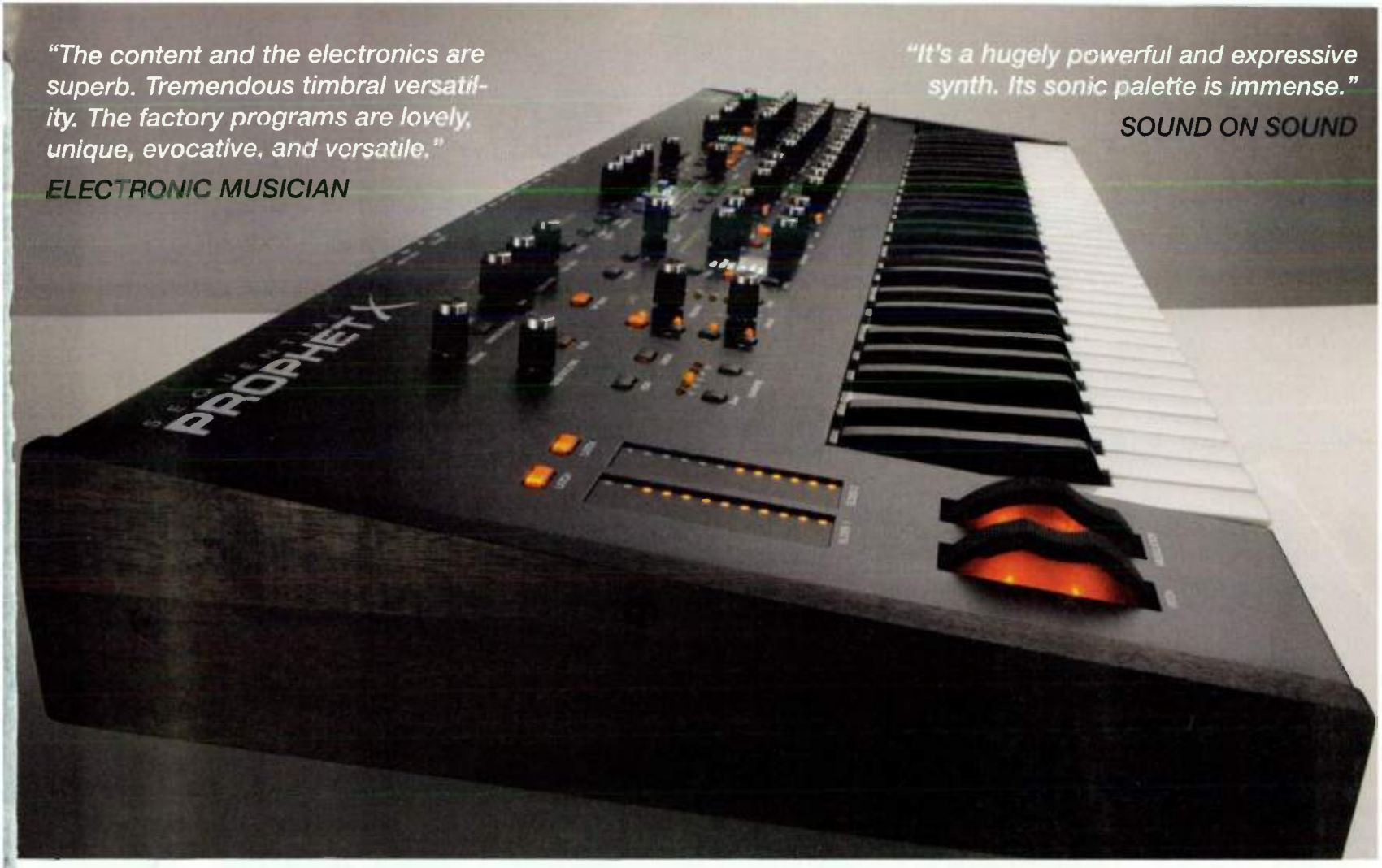
THE SPEEDY GUIDE TO PRO-SOUNDING TRACKS

Yamaha CP8
Full In-Depth T

Easy Guide
Signatures



More!



PROPHET XL

HYBRID SYNTHESIZER

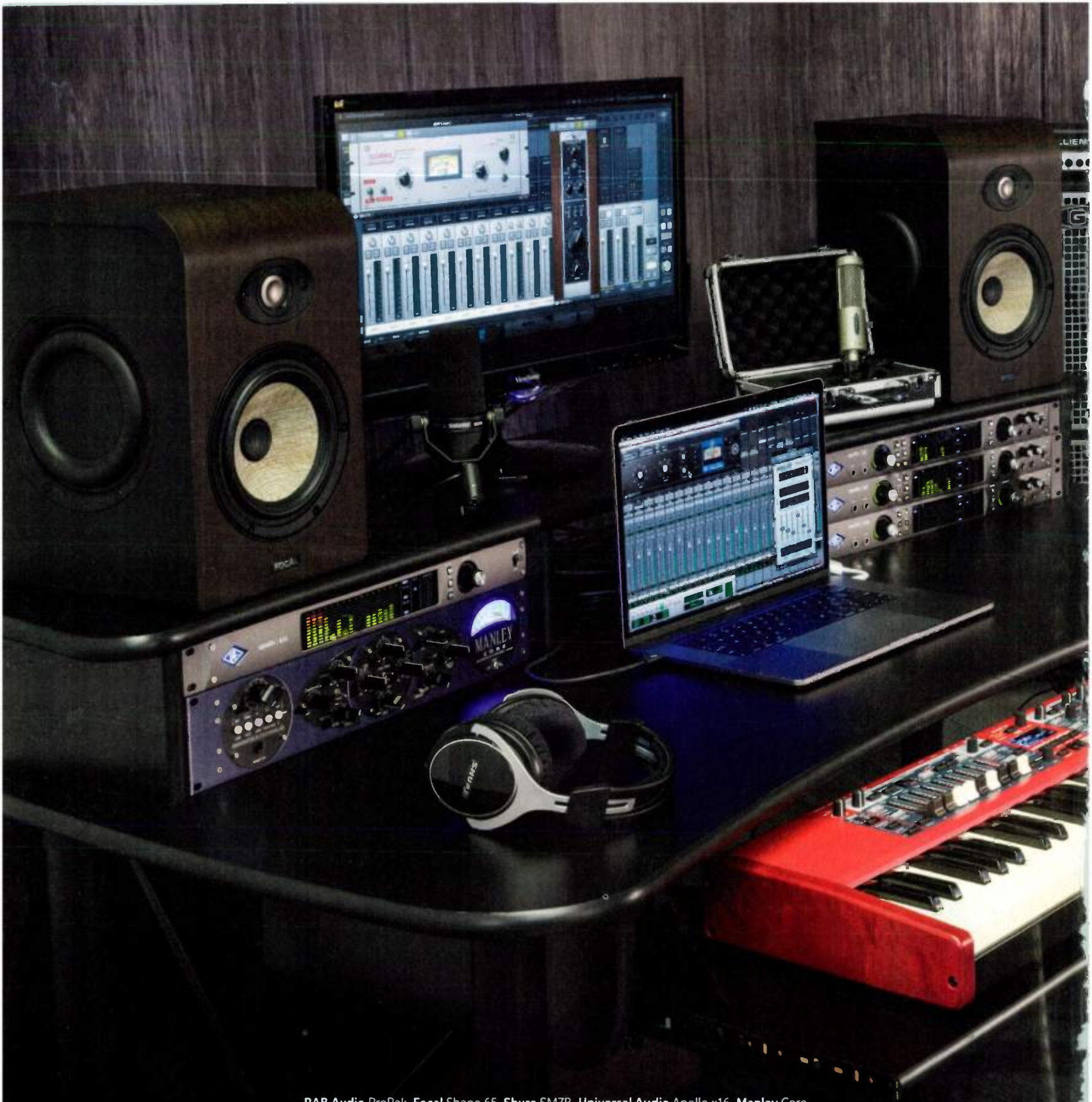
SAMPLES + SYNTHESIS

SAMPLES BY 8DIO

76-KEY SEMI-WEIGHTED ACTION · FULL SYNTHESIS ENGINE · 150 GB OF SAMPLES · 50 GB USER SAMPLE IMPORT

SEQUENTIAL®

www.sequential.com



RAB Audio ProRak, **Focal** Shape 65, **Shure** SM7B, **Universal Audio** Apollo x16, **Manley** Core, **Shure** SRH1540, **Universal Audio** Apollo x6, Apollo x8, Apollo x8p, **Royer** R10, **Nord** Electro 6D, **Gallien-Krueger** 1001RB-II, **Gallien-Krueger** Neo 412,

Keyboard MAGAZINE

ARTHUR BAKER

ARTURIA PIGMENTS

GEAR

HOW TO

Electronic Musician (ISSN 0884-4720) is published monthly by Future PLC, 28 East 28th Street, 12th floor, New York, NY 10016. Periodicals Postage Paid at New York, NY, and at additional mailing offices. Canada Post: Publications Mail Agreement #40612608. Canada Returns to be sent to Bleuchip International, P.O. Box 25542, London, ON N6C 6B2. POSTMASTER: Send address changes to Electronic Musician, P.O. Box 232, Lowell, MA 01853.

XTRAX STEMS 2

One Song In. Three Stems Out.

Automatically Separate Any Song into Vocals, Drums and Remaining Music.

Powered by brand new algorithms based on artificial intelligence, XTRAX STEMS 2 offers faster, cleaner stem separations, backing tracks and a cappellas at the same low price. Created for DJs, music producers and remix artists, XTRAX STEMS allows users to separate any song into its vocal, drum, and remaining music components for easy sampling, remixing and unmatched control during live sets.

audionamix.com/june-em

Audionamix
SEPARATE 2 CREATE

Editor's Note

Mastering the mix

Mixing tracks can be a tricky business. Perhaps more than any other aspect of music making, it really blurs the lines between technical skill and creativity. Sure, there are rules you can follow — knowing what frequencies certain track elements should sit at, understanding how to set up gain staging — but there are no clear right and wrong answers when it comes to how a track should be mixed, and getting it right requires plenty of practice, patience and a clear creative vision. What might seem right for one track isn't always going to work for another. The rules applied to a radio-friendly pop song, for example, can't be carried over to a bass-heavy dance track.

That being said, there are things you can do to help yourself along, from tips for speeding up your mixdown workflow to tried-and-tested techniques that can be a quick route to getting sounds into the right ballpark. It's this sort of practical advice we're focusing on in this month's cover feature. While we might not be able to tell you exactly how to mix your next track, our tips and techniques can hopefully make the task a little less daunting.

For a more purely creative burst of inspiration, check out this issue's Masterclass feature, in which we dig into the capabilities of Pigments, the first original softsynth from French brand Arturia. With a combination of virtual analog and wavetable engines, plus lots of modulation and some great effects, it's a killer tool for sound design. Dive in on page 52, as we show you everything you need to know to get the most out of its well-equipped synth engine.

However you're making music this month, we hope you find inspiration inside this issue!

SI TRUSS
EDITOR

electronic MUSICIAN

Vol. 35 No. 6 JUNE 2019

www.emusician.com

FOLLOW US
twitter.com/EM_Magazine
facebook.com/ElectronicMusicianMagazine
instagram.com/electronicmusicianmag

CONTENT
Editor
Si Truss simon.truss@futurenet.com
Content Director, Music
Scott Rowley scott.rowley@futurenet.com
Production Editor
Alice Pattillo alice.pattillo@futurenet.com
Art Editor
Steve Dawson steve.dawson@futurenet.com
Editor *keyboardmag.com*
Jon Regen kbwebeditor@gmail.com

Editors at Large: Geary Yelton, Mike Levine, Francis Preve, James Russell
Contributors: Jono Buchanan, Francis Preve, Jerry Kovarsky, Michael Ross, Dave Clews, James Russell, Ronan Macdonald, Scot Solida, Danny Turner, Simon Arblaster, Ben Rogerson

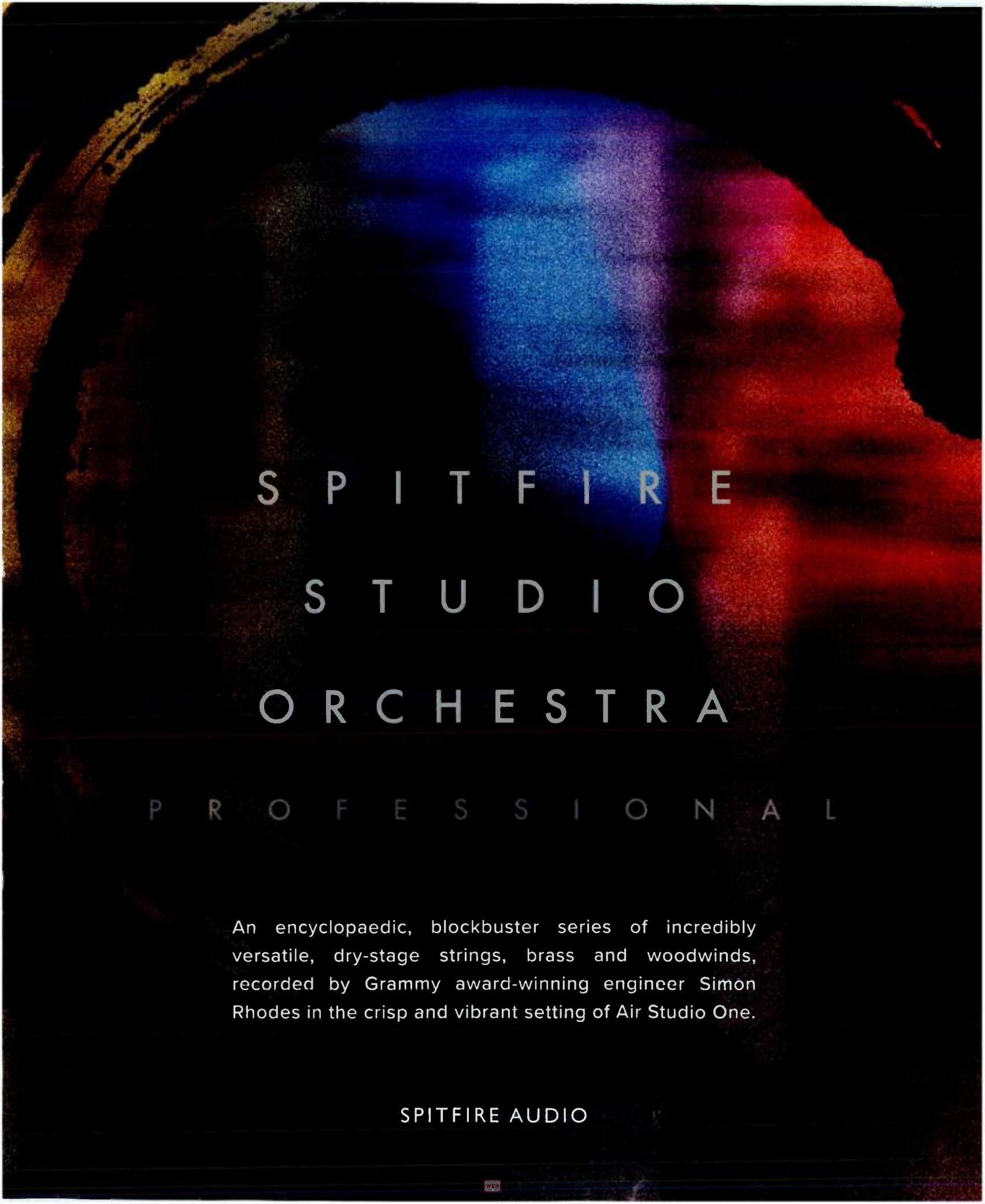
FUTURE MUSIC GROUP BUSINESS
Chief Revenue Officer
Luke Edson luke.edson@futurenet.com
Advertising Director
Jonathan Brudner jonathan.brudner@futurenet.com
Advertising Director
Mari Deetz mari.deetz@futurenet.com
Advertising Director
Jason Perl jason.perl@futurenet.com
Advertising Director
Scott Sciacca scott.sciacca@futurenet.com

MANAGEMENT
Chief Content Officer
Aaron Asadi
Commercial Finance Director
Dan Jotcham

REPRINTS AND PERMISSIONS
For article reprints please contact our reprint coordinator at Wright's Reprints: 877.652.5295

Please direct all subscription orders, inquiries, and address changes to: 888.266.5828, outside the US 978.667.0364, emmag@computerfulfillment.com

Back Issues are available for $10 each at 800.289.9919, 978.667.0364, emmag@computerfulfillment.com


SPITFIRE
STUDIO
ORCHESTRA
PROFESSIONAL
An encyclopaedic, blockbuster series of incredibly versatile, dry-stage strings, brass and woodwinds, recorded by Grammy award-winning engineer Simon Rhodes in the crisp and vibrant setting of Air Studio One.
SPITFIRE AUDIO

SSL SIX COULD BE THE COMPACT CONSOLE YOUR STUDIO HAS BEEN WAITING FOR

SSL has released the SiX, which it's proudly calling the 'Ultimate Desktop Mixer'. Now there's a chance to get a slice of that SSL sound in your DAW at a fraction of the space and cost of a full console. The mixer features two recording channels with SuperAnalogTM mic pres, a one knob version of the SSL Channel Compressor, two-band Channel EQ, inserts and 100mm faders.

You also get a two-knob version of the G-Series Bus Compressor on the main mix bus and the Listen Mic Compressor on the talkback channel. SiX also includes a mixdown mode, which is a 12-channel summing system that promises analogue detail, depth and width to your mixes.

"With over 30 years' involvement in the design of SSL consoles, when developing the concept of SiX, I really thought hard on what our users appreciate about our larger consoles; what helps their workflow and delivers quality results for them," says Niall Feldman, SSL Director of New Products.

"The big challenge then was how to deliver those values and features in a compact product. Working with a great team and focusing on audio quality, workflow and flexibility, the resultant SiX mixer is one of our proudest achievements."

SSL SiX is available now priced at $1,499 plus tax.

By James Rusell
As one of Electronic Musician's cadre of Editors At Large, James is responsible for keeping his finger on the pulse of the music software world, reporting on the latest developments in plugins and DAWs. He also takes a more irreverent look at music software as co-host of Appetite For Production Podcast, and is often to be found creeping about on Twitter: @rusty_jam

SOFT OPINIONS

25 new guitars in one... plugin?

Guitars seem to be at the forefront of my mind this month. Not only have I just taken mine to the tech to give it a new lease of life, but I've been seeing many guitar-focused pieces of software popping up from all around. Conspiracy? Confirmation bias? Whatever it is, the point remains: this month has seen Positive Grid update its Bias FX software to version 2, and I've seen at least three new 'virtual guitars' hit the news in their own ways.

But the most interesting axe-themed software package arrived in the form of a plugin from Blue Cat Audio. I've been enjoying the furrow that this developer has ploughing in recent years, with a particular focus on plugin-hosting plugins. But like I was saying, the latest string on Blue Cat's bow is a bit more... coiled.

Blue Cat Audio's Re-Guitar plugin

Ever wanted to swap your smooth Strat for a chunkier Les Paul but didn't have one lying around? Or wanted to trade your bitey Ibanez for a sweeter Telecaster tone, or even for an acoustic guitar, with the touch of a button? That's exactly the intention behind Re-guitar

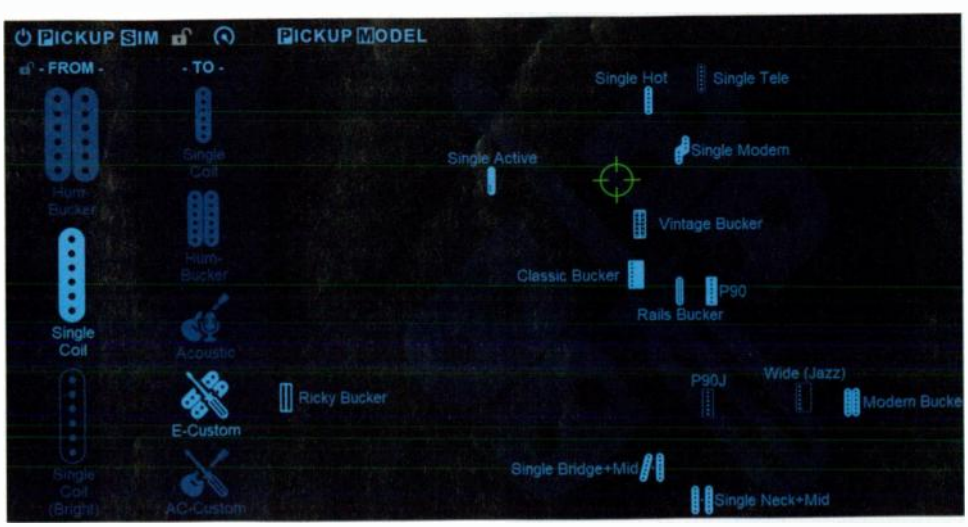
The idea: select the pickup type that's producing your input signal – Humbucker, Single Coil or Single Coil (Bright) – then select the destination pickup category (Single, Hum, Acoustic...) and the exact type of pickup from that category. There's a comprehensive selection that can take you on a tonal journey without needed to set foot in a guitar shop.

Each pickup model has its own Volume and Tone controls, even for the various acoustic models and acoustic simulator pedals, and there's a Hollow Body section for adjusting the thickness if you're emulating that kind of guitar.

The E-Custom and AC-Custom settings look like very interesting sonic destinations to visit. These two pages splay out a bunch of models of pickups or acoustic guitar bodies, plotted in a 2D plane. The idea is for you to move the cursor between the models in order to morph between them. You might want, say, to land your tone close to that of a Mini Jumbo acoustic body, but stretch it out with some Modern Dreadnaught flavouring, or a tinge of Acoustic Simulator Pedal.

Soapbar soapbox

Let's get down to the nitty gritty: Re-Guitar probably won't be 100.0% perfect – especially in the estimation of guitarists who can tell the difference between two guitars of the exact same type – but it's probably going to be very good at what it does, and will surely serve most of the needs of the wannabe tone hound who doesn't have thousands to spend on a variety of guitars [Hi].



Re-Guitar's custom models give you a navigable 2D space to blend between pickups

Re-Guitar brings to mind those plugins that claim to ape different types of studio monitors on your master bus: you can get the frequency response just right, but you can't emulate, say, the transient response, overshoot, damping or whatnot of any other studio monitor simply by changing your input signal.

Neither will Re-Guitar do away with other inherent characteristics of your pickups. Case in point, I recently asked my guitar tech if there was anything that can be done about my Fender Jaguar's inherent buzzing. "I can file down the pickup claws, but then it would sound like a Strat." This plugin could do something similar, but I expect the buzz will remain!

As long as you don't expect the earth from it, I think that opening your mind to a processor like this can only ever help you in your eternal search for new sonic horizons. And don't forget, that's why we all got into this in the first place, isn't it?

Re-Guitar is available as a VST/AU/AAX plugin from bluecataudio.com, and is also operable as part of Blue Cat's Axiom plugin. It costs $99, but may still be on its introductory discount ($69) period when you read this.

THIS MONTH IN SOFTWARE

The NAMM hangover has ended, and new experiences of virtual tools that you've never dreamed of await for the months ahead

Softube's Parallels summons up classic oscillators easily

Until recently, Softube had mostly been known for its hardware emulation talents. With a plugin back-catalog made up mostly of classic hardware effects, as well as several projects done in collaboration with other companies – Universal Audio, Propellerhead and more – you'd be forgiven for wondering when exactly the Swedish software developer was going to get around to making a classic synth.

Actually, the company's roster wasn't completely devoid of instruments; the Heartbeat drum synth has been knocking out a pulse for a few years, and the recent(ish) launch of Modular – a complete Eurorack setup emulated for use in your DAW – gave a few indications of where things might be heading.

Enter Parallels, Softube's new release, and the company's emulation talented are being put to even better use, with a synthesizer that apes a collection of classic and sought-after units alongside a gathering of new inspirational source signals.

Parallels' Sources are the true heart of the synth – you could think of them as 'oscillators' in the classic sense, but rather each is a 15-second sample of sound which evolves over time. Sources included physically modeled entries, environmental samples, chords, and of course those analog and digital synth sounds.

Each Source has its own amplitude envelope, rate of movement through the source recording, and controls for octave, pitch and vibrato, and there's a large Mix knob to set the balance between the two.

Naturally, the signal progresses through filters (aka 'Shapers' in Softube speak), while modulation is on tap. The effects section gives you Distortion, Chorus, Flanger, Delay and Reverb. Given Softube's effects-modeling history, these should certainly be no slouches either.


Parallels brings to classic synths and more back to life

Parallels is available now as a VST, VST3, AU and AAX plugin, from softube.com, for the introductory discount price of $129. It's usually $149.

James Russell

Tracktion Waveform 10 is now available

After its announcement at NAMM in January, Waveform 10 has recently become available. You may not be very familiar with Waveform, but this top-tier DAW has had a colorful history. Starting out as Tracktion, the software was taken under the wing of Mackie, then later re-acquired and re-established under the Tracktion brand again.

Older versions of Tracktion are regularly made free, with the full version of Tracktion 7 (the version that was released in 2016) being currently available to anyone and everyone right now for no money. Based on past behaviour, we can expect Tracktion to remove the price tag from version 8 – the first of the series dubbed Waveform – sometime within the next six months or so.

The best DAW you've never heard of? Waveform 10 adds 62 new plugins with a focus on its Rack system

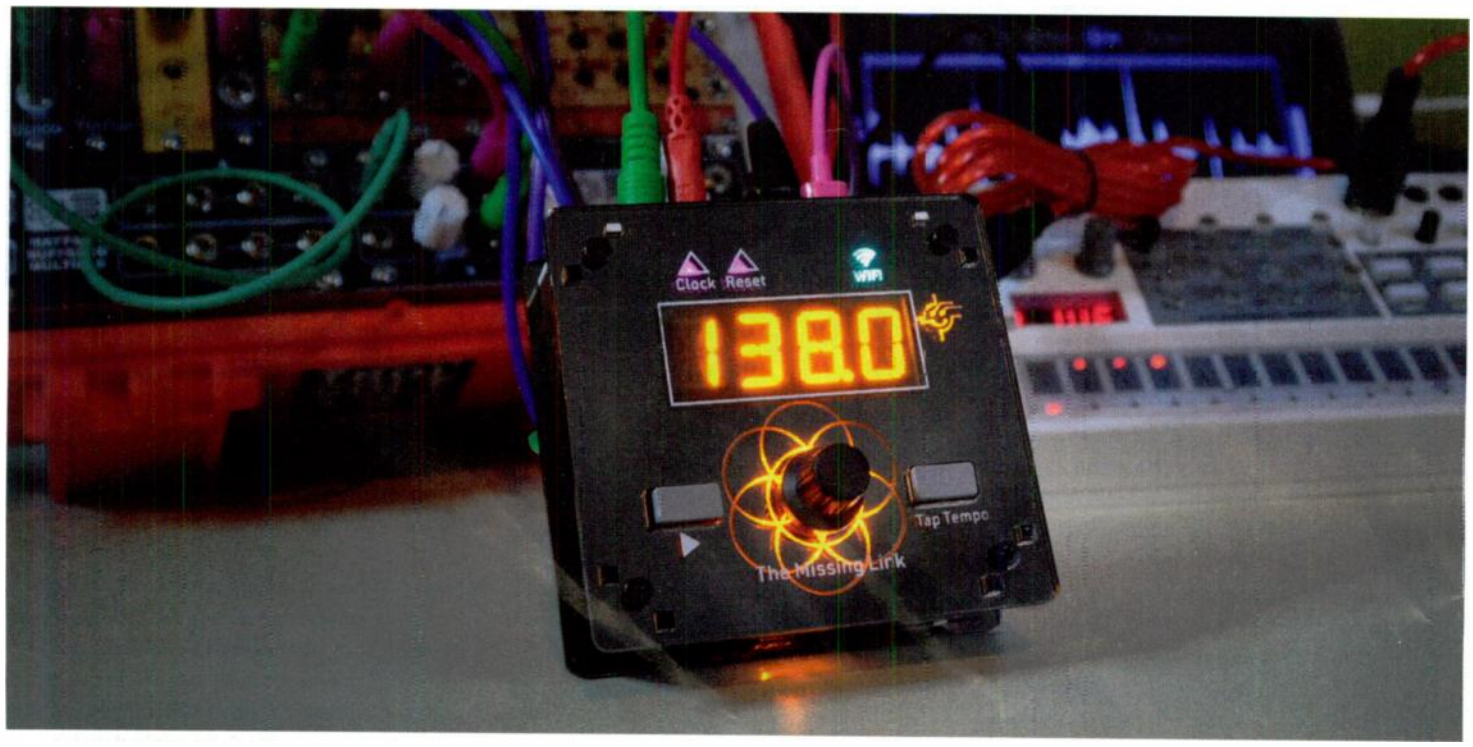
In version 10, Tracktion is pumping up Waveform's modular processing offering. The DAW has actually had its own plugin 'rack' system since day one, and it's most likely this that inspires one of version 10's key new features: 62 additional built-in plugins, created by AirWindows, covering the gamut of effects processing. Previously, just a few devices were available with Waveform, although some other Tracktion-made and Tracktion-distributed offerings also came with it.

Not only do these 62 plugins bulk up Waveform's value to users, they also increase the value of its rack system, being simple, to-the-point processors that would be equally at home as part of a channel strip or a rack setup.

Waveform 10 users also get a version of Auto-Tune Access, meaning that the DAW scores the hat-trick of all three major pitch treatment solutions: Auto-Tune, Melodyne and Elastique Pro.

The in-built MultiSampler – a previously cheerful, straightforward sampler device – has been improved in a few areas, allowing keyboard control of sample selection, layer mute and solo, and the ability to sample from "the retrospective record buffer"; seemingly akin to Ableton's Capture but for audio. The MultiSampler can also record its input from other apps running on your system.

Waveform 10's Actions panel is something inspired by other software, but this is the first time I've seen it used in a DAW context. This left-hand panel contains entries based on actions related to what you've currently selected, and allows you to favorite certain



The Missing Link brings Ableton Link to your hardware setup

The best thing about Ableton Link is that it's so simple and, for the most part, fuss-free. The worst thing about Ableton Link, though, is that it isn't implemented in much hardware, especially modular synths, sequencers, samplers and drum machines. Well, until now, that is...

Circuit Happy has launched a new piece of hardware designed to bring Ableton Link syncing to anything with a Clock/MIDI connection: The Missing Link. After the first look at last year's Knobcon event in Chicago, the folk behind The Missing Link have been beavering away at a software update, which now sees the device support MIDI clock.

The hardware itself is essentially a desktop modular clock with wireless integration. WiFi set up seems pretty painless. If The Missing Link can't find a WiFi network to connect to, it will just go into Access Point Mode, then all you need to do is connect directly to the device to configure it with your network.

The unit sends clock and reset information out via the 3.5mm ports, allowing you to directly connect your modular unit of choice. To connect via MIDI, you will need a USB OTG adapter; this can then connect to any class-compliant MIDI interface which may or may not be built-in to your hardware instrument, so another device may be required there. However, once you're all set up, you'll be able to connect two units wirelessly and incorporate any Ableton Link-enabled iOS app into the mix.

The Missing Link is available now priced at $200.

Novation adds more new features to Circuit

Novation has updated its affordable groovebox Circuit, adding four new features coming for version 1.8. Living up to its claimed commitment to the ongoing development of the Circuit, the UK brand has added more features that it says "are sure to benefit every producer's creative workflow."

The new features include non-quantized recording, which lets users live input synth and drum tracks directly to the sequencer's microsteps (ie, letting hits fall between the main 'steps' of the sequencer). Microsteps can also now be edited using the sequencer grid. Each step gets six available microsteps, allowing for syncopated lines and rapid-fire trills.

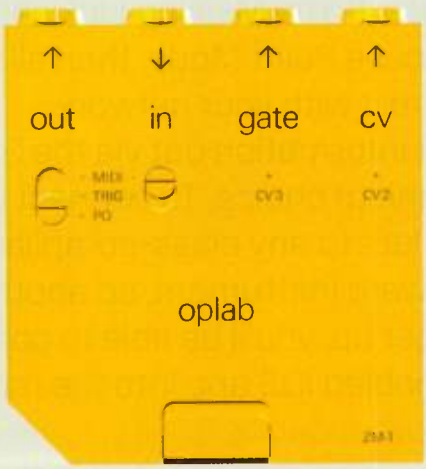
Elsewhere, the update adds per-note velocity tracking for the two synth engines, meaning that notes that share a sequencer step can now each have different velocity levels. Finally, version 1.8 also introduces assignable MIDI channels. This will be welcome news to anyone who regularly uses Circuit's (excellent) sequencer to control external hardware.

The update is free to all users, and can be grabbed now via the Components app.

Teenage Engineering releases the Oplab module, a CV and MIDI add-on for the OP-Z

After its first reveal at this year's NAMM show, we can now welcome the Oplab module, the first hardware upgrade for its compact OP-Z synth, sampler and sequencer.

Not wanting to just stick with software updates for this latest instrument, Teenage Engineering equipped the OP-Z with a small hardware module expansion slot, accessed by removing the rear panel. This first module adds three CV outs and one Gate out, as well as options for PO sync and MIDI over 3.5mm via the included DIN adapter. This significantly expands on the sync capabilities of the base OP-Z, which can send MIDI over its USB connection, but lacks the necessary I/O to control external hardware without a computer in between. This is sure to please all those who are looking to fully unlock the capabilities of the instrument's fun, well-equipped sequencer.



Those with a good memory may remember the original Oplab, the now discontinued cardboard-mounted, I\O Swiss Army Knife from five years ago. While the product may be dead, the name lives on.

There's no word on what other modules are in the pipeline. The Oplab is available for $149 and can be purchased from the Teenage Engineering website.

Get custom control of your hardware with Joué's re-Connect setup

Joué is a customizable controller made from wood and built around a pressure-sensitive surface. It can be adapted by placing modules onto it, in the form of piano keys, a guitar-style fretboard, drum pads or 3D control surfaces.

With the release of their new re-Connect cable accessory, however, the controller now looks even more interesting, as it allows use of Joué with any hardware synth without the need for a computer in between. The cable is powered and sends native MIDI messages without latency, so it can be hooked up for direct control of any MIDI-equipped hardware.

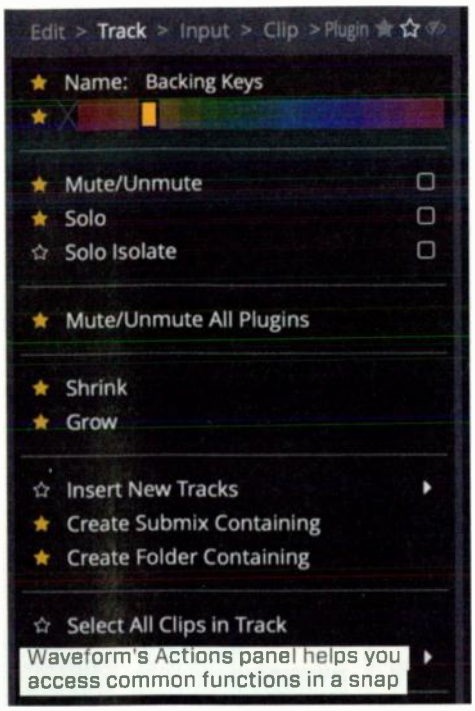
You can pick up the new re-Connect MIDI cable for 39 Euros, or as part of the re-Connect bundle, which includes the cable, Joué board and GrandClavier overlay for 477 Euros.

Twisted Electrons unveils the Deton8 sample-powered drum machine

Twisted Electrons has announced their new instrument: the Deton8, a diddy desktop drum machine that's sure to pop some heads.

The Deton8 is part hybrid sampler and part drum synth, with lo-fi 8-bit samples and wavetable synthesis at your disposal. Deton8 holds 16 patterns of 1-16 steps each and eight voices, including a triangle wavetable synth part that can be shaped into a square wave. On top of that, you're afforded a whole bunch of performative effects, which include eight different stutter modes and more.

Currently, there is no sampler editor or uploader, but the folks behind Twisted Electrons say they're working tirelessly to have that implemented soon.



actions you take regularly.

Tracks in Waveform can now be grouped, and tracks or single clips can now be edited within their own windows, muting and soloing can be done to multiple tracks by dragging , and when Waveform 10 crashes, it automatically relaunches and reopens the project, making things just that bit less frustrating.

Waveform 10 is available in three editions with varying content from $119 to $499. Upgrade pricing from Tracktion 7 upwards costs between $69 and $369. Find out more at tracktion.com

James Russell

BIAS FX 2 looks like a tonehound's playground, and a vintage guitar shop, too

Positive Grid's BIAS FX suite is about to hit version 2. This is set to deliver a re-engineered DSP engine, new amps, pedals and racks, advanced Fuzz, Time and Harmonizer modellers, and a Guitar Match system that's designed to make one guitar sound like another.

The amp engine is the same as the one from Amp 2, one of Positive Grid's other bluechip releases, and is said to be supremely responsive and bursting with rich tone and dynamics. Your stock of pedals, meanwhile, can be added to by downloading custom models from the ToneCloud, and if you go for the top-line BIAS FX 2 Elite version, the BIAS Pedal software comes included.

The Guitar Match feature, meanwhile, looks similar to one of the options in Blue Cat Audio's new Re-Guitar, in that it enables you to emulate the pickups, body type and body thickness of a range of classic guitars. The eye-catching premise is that you can make your humble guitar sound like any classic model you fancy.

Elsewhere, it's very much a case of 'more, more, more'; with so many effects at your disposal, this could be a tonehound's paradise. There are Impulse Response-modeled Celestian cabs, too, and it's been made easier to configure your virtual pedalboard setups and switch between them. There's also a built-in loop recorder, so you can capture those moments of inspiration when they strike.

BIAS FX 2 launches on March 21 for PC and Mac in VST/AU/AAX and standalone formats. It's available for pre-order now, with prices starting at $99 for the standard version, $199 for the Professional edition and $299 for the Elite. Find out more at positivegrid.com.

MusicRadar

PSP Audioware HertzRider offers modulated frequency shifting

It's a while since PSP Audioware's heyday of frequent product releases, so it's nice to see the developer bringing a few more plugins to market. HertzRider is a frequency shifter with a few tricks up its sleeve.

We all remember the difference between frequency shifting and pitchshifting, right? Pitchshifting tends to multiply a signal, keeping it at musically-intelligible logarithmic settings, whereas frequency *shifting* is purely addition or subtraction, enabling small tweaks that bear little relation to the octave you're in, making the result vary wildly depending on the material.

Frequency shifting, then, can be a good effect to throw onto non-harmonic sources like drums, or applied in small amounts to gently thicken a sound by just a few cents. HertzRider takes this concept and, er, rides with it, adding and LFO and Spread parameter to help animate the pitchshifting effect. The left and right sides can be shifted separately, and there's a giant Phase strip control in the middle, too.

The result is a plugin that sounds equally at home while mixing as it done for more creative, sound-design duties, and one which PSP says provides "Frequency shifting on steroids". Doping's not illegal in music production, is it?

James Russell

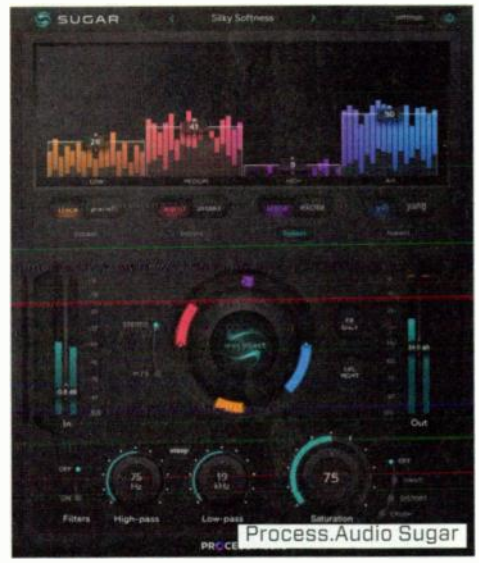
Process.Audio's Sugar aims to sweeten your sound with top producers' processes

This "audio sweetener plugin" comes as a new venture from the team behind puremix.net, an online learning site that's seen involvement from some of the biggest names in production (think Dupont, Scheps, et Alge). For their first plugin at Process.Audio, we're not sure exactly which producers they've enlisted, but the idea seems to have been to distill their go-to mix tricks – and hence their years of production experience – into this one product.

Put simply, you could get away with calling Sugar a 'multiband enhancement' plugin. Four bands are applied with varying amounts, and each is switchable between two modes: Thick or Punch for the bass band, for example, or Yin and Yang for the topmost Air band. One large, central 'Wet Inject' knob seems to apply the whole setup to greater or lesser amounts.

Following that come high- and low-pass filters, and a Saturation control with Drive, Distort and Crush modes. The plugin costs $149 from processing.audio but is free to Puremix Pro subscribers.

James Russell



New Apple iMac promises to be twice as fast as the previous model

The new iMac features updated processors, with the 21.5" model starting with an 8th-generation quad-core and also offering, for the first time, 6-core Intel chipsets.

The 27" iMac will, for the first time, feature up to 9th-generation 6-core and 8-core processors. Figures for the 21.5" iMac show a 60% improvement in performance, where as the 27" iMac looks to be delivering up to 2.4 times faster performance when testing soft synths in a DAW environment.

Prices for the new 21.5" iMac with Retina 4K display start at $1299, while the new 27" iMac with Retina 5K display starts at $1799. If you're looking to save a bit of money, the non-Retina display iMac configuration continues to be available for $1099.

MusicRadar

MEET YOUR SESSION GUITARIST
STRUM GS-2

200+ Acoustic and electric guitars **700+** Strumming patterns with your own chord progressions **100+** Styles

TAKE IT FURTHER WITH EXPANSION SOUND PACKS

NKS NATIVE KONTROL STANDARD

WINDOWS · MAC OS X 32-/64-BIT
VST · AU · RTAS · AAX NATIVE · NKS · STANDALONE

www.applied-acoustics.com

A|A|S
Applied Acoustics Systems

MIX FASTER, MIX BETTER!

Turn off your synth, power down your amp and turn on your listeners... Time to mix it up and make some magic

From the seed of inspiration, through the many hours of programing, producing, recording, tracking, editing and arranging, you're finally ready to enter the phase which separates your carefully crafted new track from its audience: the mix. These days, it's easy to see the mix phase as being an overlap of many of the processes listed above, as we all tend to make 'balancing' decisions about the sounds we reach for as we work. But there's also no doubt that there is a distinct phase where we turn off our synths, power down our guitar amps and mute the output of our modular systems to concentrate exclusively on the parts we've committed to. And at this point, the fun really starts. There simply is no end to the assorted ways in which it's possible to sculpt and craft a mix; the process can be so involved that it can be easy to lose sight of the eventual goal. So let's clarify this a mix is designed to present a track to its listeners in the most musically-captivating way possible. So, to that end, let's get on with learning some crucial techniques.

Mapping your mix

When done well, mixing sounds effortless. For proof of this, tune into your favorite radio station or streaming service. You'll immediately be greeted with numerous tracks which will almost certainly leave you with the impression that, in a sculpted, sinuous way, each one provides the only possible representation of their musical ingredients, such is the clear and obvious balance struck by each track's mix engineer. But, in truth, a million unconscious decisions will have gone into the mixing of that track. Indeed, in another mix engineer's hands, the track might sound very different and, potentially, just as good. So we learn an essential lesson, which is that for all of the suggestions made in any article you might read about mixing (including this one), there is no 'one fixed way' of mixing a record.

Saying that, there are two hugely important considerations. The first is that your job as the mix engineer is to guide the listener from one moment to the next in the most musically satisfying way possible.

The second is that familiarity with the processes on which mix engineers are reliant is essential; you need to know about sidechain compression and dynamic EQ, and you need to be able to identify the sibilant frequencies in your singer's voice so that you know exactly which ones to attenuate. You need to know about stereo widening techniques (and when to use them) and you also need to know which parts you should address when bass frequencies become conflated.

We aren't interested in learning these techniques for their own sakes; we want to know how they work and what they can do so that when we're working on our own tracks, we can respond immediately to the issues raised by the elements of our tracks as musically as possible. In doing so, we're resolving problems with the most appropriate techniques in our creative lockers.

So remember that it doesn't matter which DAW or version of an EQ plugin you reach for. The techniques we're exploring aren't about operating systems or computer platforms; we just want to help you make great-sounding records. (And we could fill the entire issue exploring many more techniques.) Yet with some creative thought, you can apply the contents of this article to other parts of your mix. For example, once you address frequency overload at the bottom end, apply the same technique around your vocal part, to ensure that there's clarity in the mix towards the top end too.

Mix toughening with parallel processing

We tend to think of auxiliaries as places for adding spatial effects. But if your mixes lack weight, they can be used for other types of effect too – here's how

Our first understanding of auxiliary channels is usually that they're places to set up effects which can be shared across a number of different tracks. If you want to add reverb to your synths, lead vocal and electric piano parts, set up your chosen 'space' on Auxiliary 1 and route each sound accordingly. Easy. But what auxiliaries really do is to provide you with a place where any effect can be added to run alongside — in parallel with — the original sound. That's where the term 'parallel processing' comes from, and it's the secret lying behind all of your favorite tracks. Through the following steps, we'll see how parallel compression and parallel distortion can be used to quickly add beef to some basic drum parts.

Start with individual beat elements programed from a variety of Battery 4 kits, with a bassline from the Monark Ensemble for Reaktor 6 and a pluck synth part from Omnisphere 2. The ear goes to the bassline rather than the drum parts, which are being outgunned.

Set up a new auxiliary to which every drum part is routed. Set up FabFilter's Pro-C 2 as a parallel processor. Set low Threshold and a high ratio (10:1), with quick Attack and Release times. Then blend the parallel compression mix 'under' the original parts for added power.

Add a parallel distortion channel for the drums and bass on Aux 3. Set up UAD's Thermionic Culture Vulture, after which, place an EQ to scoop unnecessary low end. You'll hear four bars without the parallel distortion and four with distortion added. There's more grit now.

DOs & DON'Ts

1 If you open an EQ plugin and discover that every band you've created is a 'boost', you're really just turning overall volume up. As a rule, most mix engineers use EQ to reduce volume for individual bands more than they use it to boost.

2 Of all of the tools most ignored by DAW-based producers, the pan dial is too often overlooked. Place each sound in its own space and you'll find your mixes will benefit enormously.

3 Insert an EQ after every auxiliary effect you add. It's too much of a gamble to hope or assume the frequency response of a reverb will be a perfect match for your track. Be ready to scoop out unnecessary content from effects channels as well as instruments and audio tracks.

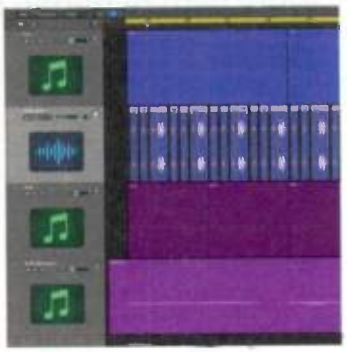
4 EQ backing vocals differently, so that they don't dominate a lead vocal. Turning the volume of backing vocals down helps, but making their tonal footprint 'thinner' than the lead often does a more effective job.

Bass overlap and build-up

Overloading bass is probably the easiest mix mistake you can make. Here are some techniques to refine your bottom end

01

Here's a mix containing four elements, all of which contribute to the bottom end in some way or other. As a result, the bass is completely overloaded and you can even hear phasing as the parts all fight for space.



02

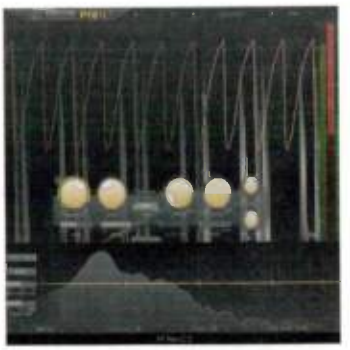
We solo the kick and drum loop. We use FabFilter's Pro-Q 3 to find out exactly where the two sounds' frequencies fight one another. We then place a low-cut filter (24dB/octave) on the drum loop at 167Hz to remove the overlapping frequencies to make the kick the dominant drum part.

03

Here's how the bassline sounds with the kick. The delay is a problem. It's echoing the whole frequency spectrum, leading to unnecessary low-end build-up. We need to filter out low-end content and reduce the delay Feedback, so that there are fewer repeats per note.

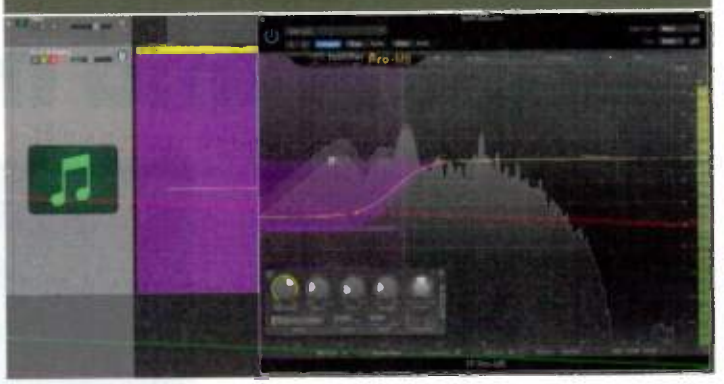
04

We adjust Waves' H-Delay to make those changes but also put a compressor on the bassline. We set its sidechain input to monitor the kick drum, so that whenever the kick and bass play together, the bassline 'ducks'. This reduces bottom-end build-up when the two parts coincide.



05

We could repeat the same trick with the synth sequence line but instead we use a multiband compressor. We set a single band in the problem frequency area, so that when these frequencies are detected, they duck, keeping them under control.



06

Here's the final mix. Suddenly, the phasing issues have gone and there's even room to turn the volume of certain parts upwards. Remember, the key is to identify which sound you want to have dominate the bottom end, and adjust other parts to this one accordingly.

How to...

Manage gain structure

Before your mix gets underway, consider gain structure. When mix engineers work with analog mixing desks, they use the gain dial as a 'sensitivity control' for the potential level of each channel. In a DAW, we don't tend to have a dedicated gain control and, worse, third-party plugin makers often try to impress us by making 'loud' presets. Add two together and your mix output will overload; even before you've added the rest of your instruments. Either insert a gain plugin at the top of each channel or build headroom into your mixes by dropping the level of all plugins by 6-8dB before you make any other mix choices.

How to...

Exercise bass control with high-pass filtering

It's tempting to think of filtering as a 'special effect' process, good only for radical shaping, like teasing a synth line in using a low-pass filter, from a distant rumble to a searing lead over 16 bars. But high-pass filters can be useful for less 'showy' mix elements, particularly if you have parts adding to the bottom end of a mix and want to do sculpting. Say you've got a pad sound with plenty of bottom end. Setting a high-pass filter with a 6dB/octave slope below 250Hz, will ease out unwanted content a bit. More radical contouring? Increase the slope to 12dB/octave and raise the cutoff point.

How to...

Avoid confusing mixing with mastering

Many processes which used to be separate now overlap; arranging and producing, engineering and recording — the lines between these blur. The most recent 'pair' to have become conflated are mixing and mastering but we urge you to see these separately. Slapping an 'output chain' on the 2-bus of your project as your ears grow tired is almost irresistible and, of course, checking how your mix 'might' sound at mastering stage won't hurt. But if your mix sounds hugely unbalanced when you remove this chain, get rid of it and get the mix right first. Mastering isn't there to save a bad mix; it's there to enhance it.

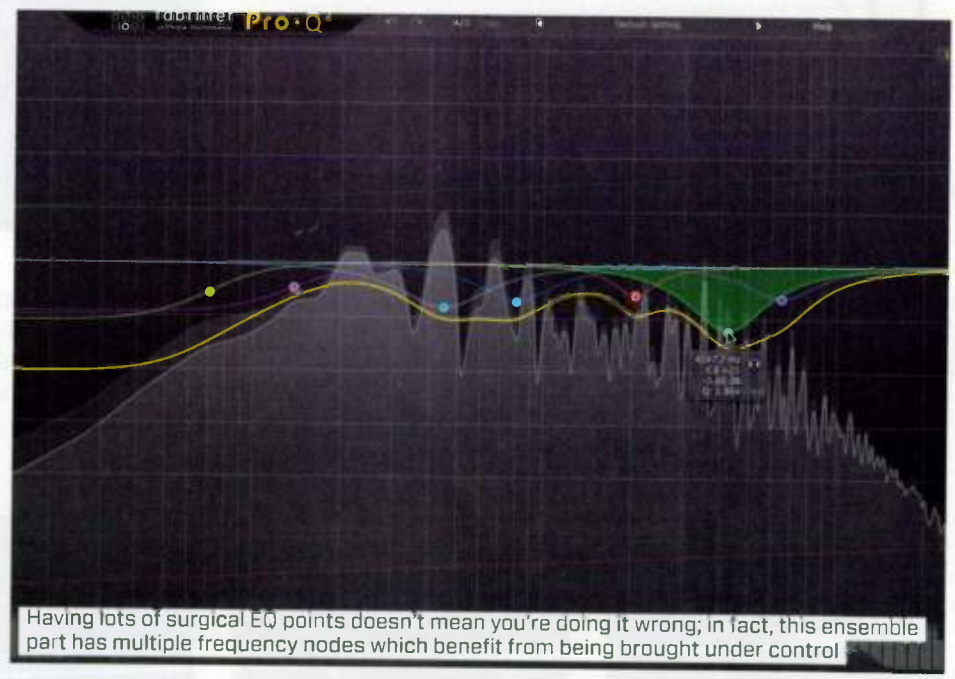
How to...

Work with dynamic EQ

Setting an EQ band to tame unwanted frequency content is common but what if the 'problem frequency' changes in volume through your track? You may want to attenuate level a lot as the volume of this frequency builds, whereas just a little attenuation will work fine when the same frequency is less prominent. Try a dynamic EQ. They let you apply frequency-specific compression, so that attenuation is adjusted 'dynamically'. Singer becomes 'screamy' around 2kHz in the chorus but sounds great in the verse? Set up that band/threshold so that volume only ducks when input level gets too hot.

Surgical and musical EQ

Without question, EQ is one of the mix engineer's most powerful weapons but there are two 'approaches' to EQing a sound and understanding these is crucial. At one end of the spectrum is 'musical' EQing, which usually involves a broad brushstroke approach to adding a desirable musical sheen to a sound. This might be a high shelf filter boost to add 'air' to a sound above 5kHz, for example, or a general bass scoop to avoid unwanted mix 'flab'. The more surgical approach is needed whenever a sound contains smaller, more problematic frequencies which need radical taming. A good way to practice working with surgical EQ is to create a part which contains a 'bell' like sound. Bells tend to contain loud, harsh frequencies, with large frequency gaps between harmonics, meaning you can get 'into' a sound and tame unpleasant spikes. Vocals also frequently benefit from identifying narrow, 'over-loud' spikes and tackling these to avoid the vocal becoming screechy, particularly when singers perform loudly.



Having lots of surgical EQ points doesn't mean you're doing it wrong; in fact, this ensemble part has multiple frequency nodes which benefit from being brought under control

All kinds of stereo considerations

From the humble pan dial to auto-panning and stereo-widening, considering sounds in the stereo field is a huge part of mixing

01

Here we can see the starting point for our track. It contains two tom parts which have been placed into a Track Stack (which controls their volume and tone together), a main beat loop and a separate hi-hat. Plus, there's a synth lead, a synth pad, a piano part and a bassline.

02

Pan-wise, everything is centrally placed, so we pan the two tom parts, with one (-50) to the left and the other (+50) to the right. So long as the levels match, pairing sounds like this across the stereo field will add mix weight and leave space in the middle.

03

We want more dynamic panning on the hi-hat sound, so we use SoundToys' PanMan to get this part moving from side to side across one bar. We can select how quickly it moves from left to right and how 'hard' it jumps, which controls how much time it will be spending in the middle.

04

Next we add a delay to the lead synth part. We switch Waves H-Delay into Ping-Pong mode, so that the added echoes bounce from side to side. This is effective as the original sound remains centrally panned, while the delays occupy the sides of the mix.

05

The synth pad could be widened too. We add Stereo Widening to this via Brainworx' UAD bx_digital V3 plugin. This single dial spreads upper frequency content left and right, smearing this airy sound across the stereo field. This makes the synth pad much more of a canvas for the other mix elements.

06

We can also enhance the stereo image of the entire mix, which we're doing via the Imager module in iZotope's Ozone 7. We make the bass band (below 167Hz) more mono compatible, but increase the stereo image in a band between 2kHz and 10kHz, and the band above 10kHz too.

Advanced compressor and EQ controls explained

Auto Makeup

As compression triggers gain reduction, we usually set a Makeup Gain parameter to re-level the signal. Auto Makeup makes this happen for you, often informed by the average gain reduction that's going on.

Feedback/Feedforward

Every compressor (or other dynamics device) consists of a detector and amp in a circuit. The detector measures the signal, with the measurement is used to control the amp. On a circuit level, the detector's input can be taken from before (feedforward) or after (feedback) the amp, with its output still controlling the amp in each case.

Hold

A control to supplement a compressor's attack and release times, in effect 'holding' the gain reduction in the same place after the signal has breached the threshold. With a quick release, a spiky signal might trigger then de-trigger gain reduction in a small space of time. Having a Hold control lets you keep that gain reduction held for longer.

Auto Release

This compression function may still take your selected release time into account, but adapts it slightly depending on the material at hand. The release may be, for example, instantaneously changed for short, transient parts of the signal or for signals with extra bass.

Hysteresis

If a signal is dancing around a single threshold, the dynamics process may switch on and off erratically. Seen on good gates and some compressors, a Hysteresis control lets you set two thresholds – one for 'on', one for 'off'.

Knee

A knee control can make the compressor's ratio vary with the signal's distance from the threshold. In a traditional, very angular 'hard knee' compression graph, as soon as the threshold is breached, the ratio is applied fully. With a 'softer' knee setting, the angle of the compression graph will be smoother, so, as the signal approaches the threshold, a lower ratio than the one you've set is applied. As the signal gets higher, the actual ratio you've set will be applied.

Linear-Phase

A traditional analogue or digital EQ has a side-effect: the frequencies around a band's cutoff are phase-shifted, which can cause problems. A linear-phase EQ (or EQ band) results in no phase changes to the signal, although there's a trade-off, as these filter models introduce some ringing, which may not be desirable on transient-heavy material. For this reason, it's worth having both types of EQ in your arsenal.

Minimal-Phase

An alternative to a Linear-Phase EQ, in which a compromise is struck between the phaseshifting of a traditional EQ and the ringing of a linear-phase EQ.

Lookahead

Setting up a short attack time in a compressor helps reduce transients, but won't fully work on extreme signals. You need some 'negative attack time'. Lookahead does just this, reading the signal ahead of time and making gain reduction trigger earlier than the signal actually passes through: useful in mastering to prevent clipping, and in sidechaining to help move a signal aside before the sidechained signal plays.

Tilt & Tilt Shelf

This EQ band type is a bit like a seesaw, giving you a cutoff frequency, with everything below that frequency being reduced or boosted, while everything above that frequency reacts in the opposite way. A 'Tilt Shelf' band effectively employs opposing low and high shelving EQs that meet at a single cutoff frequency.

Proportional Q

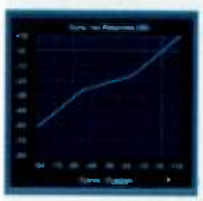
A type of EQ band that's more typical of an analogue-style EQ. With this, the Q of the filter band will be increased as its gain (whether positive or negative) is increased. Some analogue-style bands may change the Gain/Q interaction differently for cuts than for boosts.

Zero Latency

A 'normal' filter band in a digital EQ, causing phase changes to the audio as a natural consequence of its filtering. For the alternatives, see Linear-Phase and Minimal-Phase filter types.

Range

Think of this control as the threshold above which compression will stay the same. If your Range shows on your input/output response graph, it appears as a 45-degree line after the ratio. In effect, this means your compression is limited to a 'range' a certain amount below 0dBFS, and anything louder than that range will see the exact same amount of gain reduction, regardless of how high its level over the threshold. This can be useful for letting the highest peaks of an audio signal pass through unscathed, depending on your attack/release timings.



Stereo Link

By definition, in stereo audio, left and right channels carry different info, but when compressing audio, we just dial in one set of controls. By default, the most basic theoretical compressor is triggered by any of the right/left channels, but changes the gain of both at the same time. To gain more independence, 'unlink' the two channels, effectively doubling the compressor into two – on the left and right channels – with the same controls to set both. Which to use? If you don't want a loud sound coming from the left to trigger compression over the tiny details on the right, you may be after an unlinked response. On the other hand, an effect reacting to both channels in the same way will have a more 'gluing' outcome.

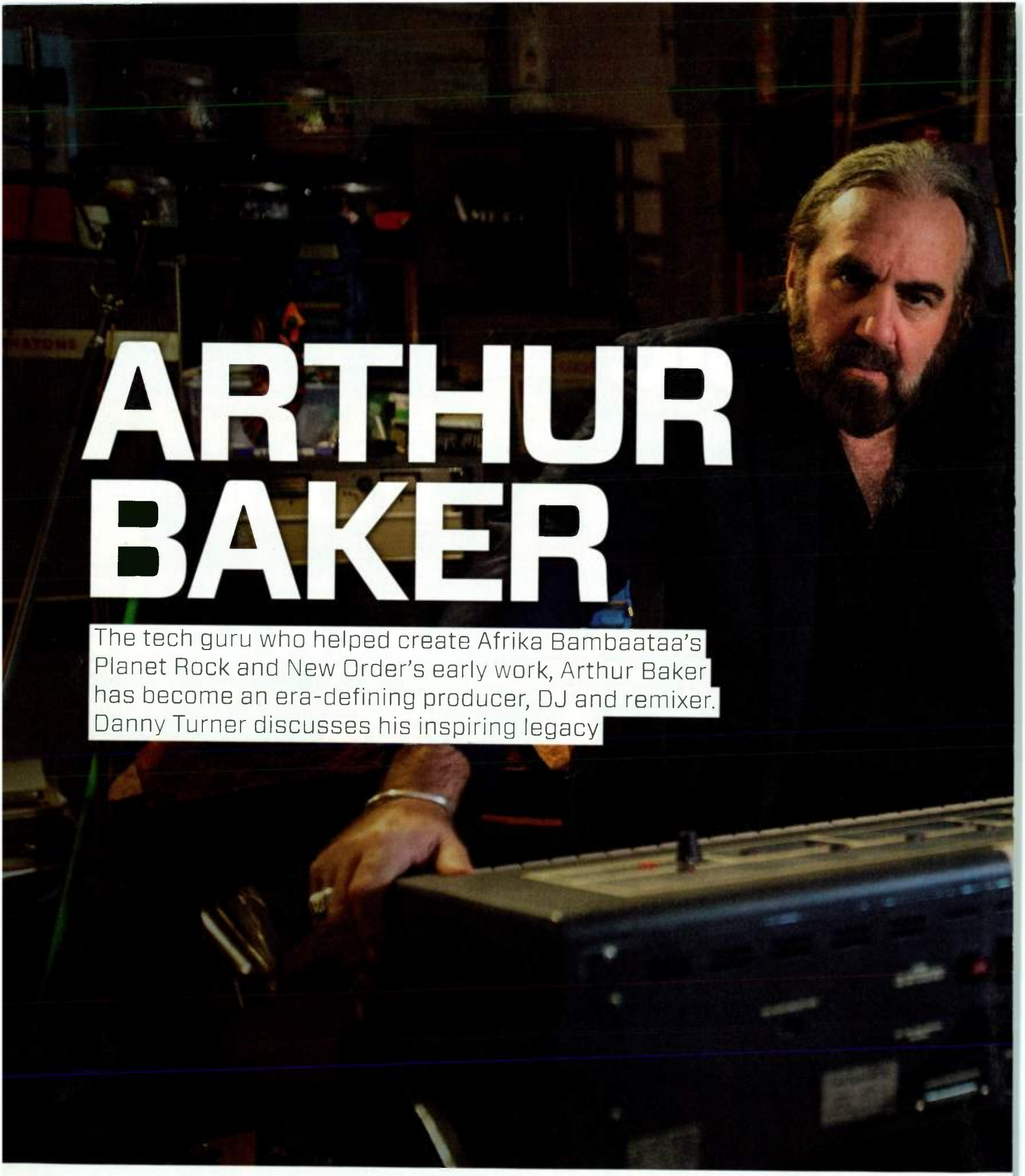
Make great music on your PC or Mac!

Computer Music is the magazine for musicians with a PC or Mac. It's packed with tutorials, videos, samples and exclusive software to help you make great music now!

www.computermusic.co.uk

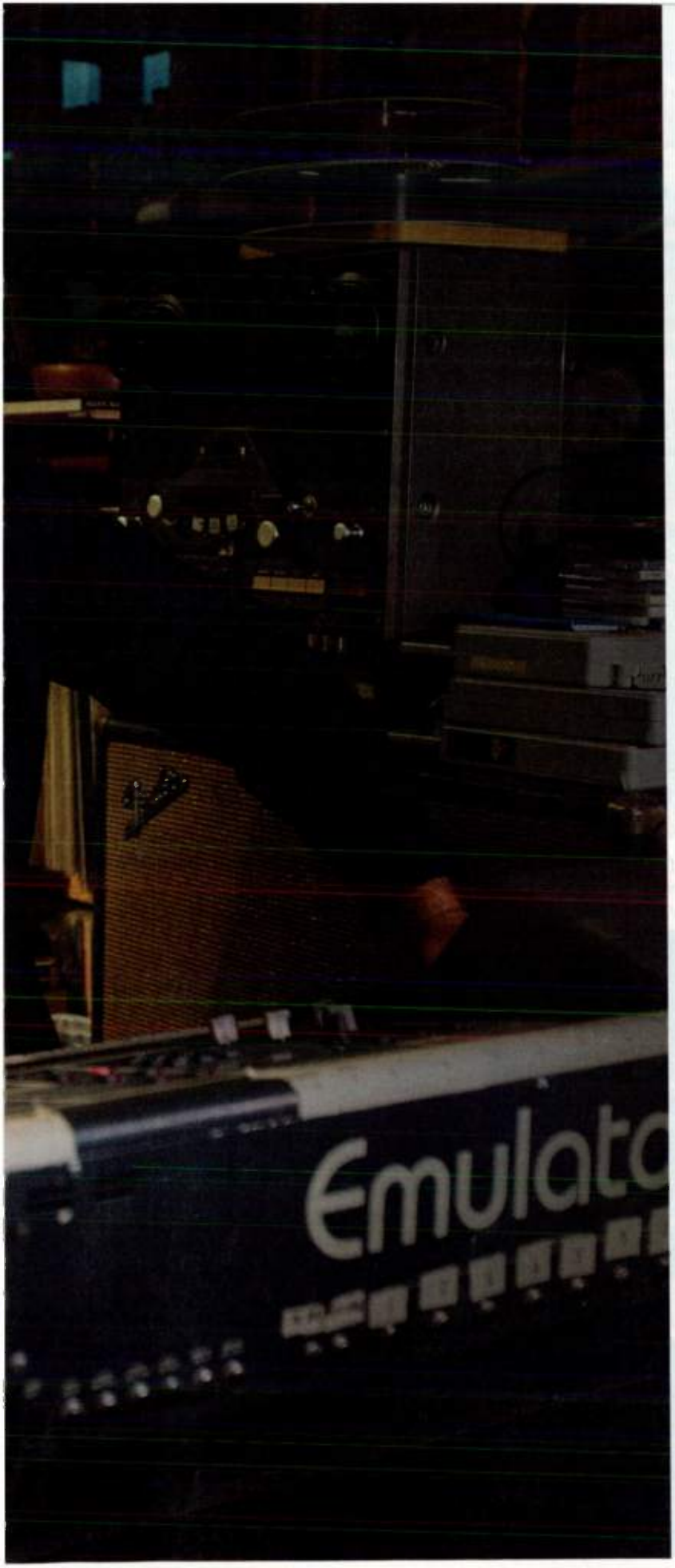
Available digitally on these devices

Available on the App Store · GET THE MAGAZINE ON Google play · Available at amazon · nook · zinio™



ARTHUR BAKER

The tech guru who helped create Afrika Bambaataa's Planet Rock and New Order's early work, Arthur Baker has become an era-defining producer, DJ and remixer. Danny Turner discusses his inspiring legacy



By Danny Turner
Danny Turner is a musician and journalist based in the UK

By his own admission, Boston native Arthur Baker wasn't a born DJ. More interested in the means behind the music, he studied engineering at Intermedia Studios and began releasing disco hits under various pseudonyms with some success. After several false starts and setbacks, Baker relocated to New York and began producing for Tom Silverman at Tommy Boy Records, where he played a key role in seminal '80s tracks by pioneers Afrika Bambaataa (*Planet Rock*) and New Order (*Blue Monday/Confusion*).

Within a few years, Baker was hot property. His studio project, Rockers Revenge, hit No.1 on the US dance chart with *Walking on Sunshine*, while remixes followed for the likes of Bruce Springsteen, The Rolling Stones and Fleetwood Mac. Entering the '90s and beyond, Baker has continued to stay ahead of the production curve while keeping the electro flame burning through documentaries like *808*.

You were initially a DJ, yet you didn't always respond well to the reaction you got...
I wasn't really a great DJ technically and was fairly impatient as a kid. If I played a track that I knew was good and it didn't get a reaction, I'd chuck it onto the dancefloor, smash it or throw it into the crowd. The club I mostly did that in was called Rasheed's in Northampton, which was a college town. But this was around '75 or '76, when it wasn't even legal for me to be playing at a club.

Sounds like an expensive hobby?
Back then there were 45s and albums but there weren't any 12" records yet, so I had to stop doing

proven myself as a producer, and I got the big remixes after I did *Planet Rock*, the Freeez track *I.O.U.* and New Order's *Confusion*. The labels thought I was creative and would give them a dance sound, so I never had a remix turned down or sent back. Sometimes the bands might question things. When I did a mix for Fleetwood Mac's *Big Love*, Lindsey Buckingham didn't get the change to the bassline, but he let it go. With The Stones' *Too Much Blood* I took Bill Wyman off and replaced him with Brian Rock — a reggae bass player, but didn't get much kickback. The people I was dealing with didn't know the market at all, so they trusted me.

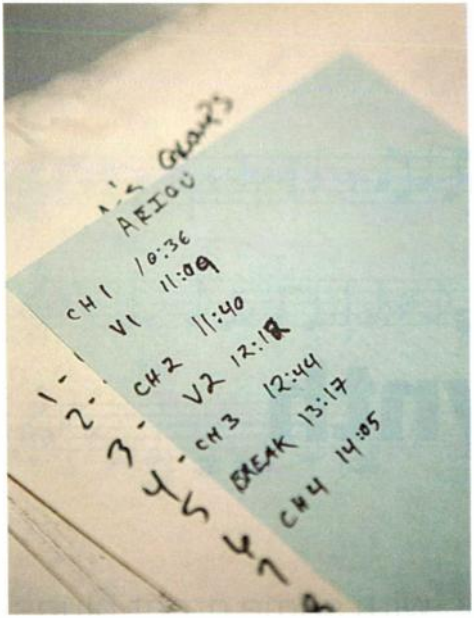
Did you ever get knocked back?
The first time I had a kickback was from someone who knew a little bit. In England, some labels would have these dance department guys and I did a remix for a label guy called Pete Edge, who's had an incredible career. I heard the song and thought it would be a hit, so I went on a TV show called *The Tube* with Paula Yates and said I was doing this remix and it's going to be a huge hit. The record was *Living In A Box* by Living In A Box, which did great, but I put on a new bassline and dropped the drums out. Edge said you have to put the drums back in because people can't dance without a kick drum. I said, no man, go to Paradise Garage and watch Larry Levan play a track with no kick drum and see how people love that — but he made me take it out.

What's your approach to remixing these days?
All the best records I made always had live musicians on top of the programing, which is something I'm going back to now. Even if you chop and move stuff around, there's something special about having live musicians on your tracks. But these days you have to dump a lot of music because that's what people want — you have to take risks. Right now I'm limiting myself. I did a really old-school, dirty house record recently and it's only got seven things in it. If you listen to the old house records, people used to love that minimalism. Production is always a learning process. As long as I've been doing it, you can always learn from listening to what people are playing or watching a crowd; and you can relearn things that you knew but forgot about.

The '80s was obviously a prolific period for you, but you seemed to change tangent after that?
In the '90s, I moved to London and did a lot of records with Brit Pop acts and some mixes for New Order. I was music supervisor on the documentary *Listen Up: The Lives Of Quincy Jones* and did the *Fried Green Tomatoes* soundtrack. Then I started DJing again and had an XFM show called *Baker's Dozen*. I did some recording for Perfecto and worked with Utah Saints, Rennie Pilgrem and breakbeat stuff.

What about now?
The last six or seven years I've been producing documentaries. One called *Finding The Funk* for VH-1 and the five-year 808 project because I wanted to document the music I was involved with. I think that's important and it was important to me. I didn't do it as an ego thing; I just wanted to document all the other guys from back in the day that did interesting things but didn't get the attention; people like Strafe, Man Parrish and Hank Shocklee.

You're also back with Rockers Revenge now?
The last few years I've been making a lot of records and collaborating with people like Steve Lawler and The Martinez Brothers, but right now I'm doing this record with Rockers Revenge and a movie about them. Through that, I've



reconnected with a lot of the people I worked with back in the '80s. For years I've been sitting at my laptop making my own music. I've taught myself programing and use Logic, but getting back together with Rockers Revenge — a group of friends I hadn't seen for 30 years — was crazy. It's like we were never apart. Not only are we making new music, but the documentary is much more of a human interest story. If I get it right, it's going to be very special.

At what point did you possess your own studio and use that for your production/remix work?
I got my studio around 1983 I think. I went from having no money and no studio to literally having an SSL desk. It was called Shakedown Sound, and for three years everyone worked there from Rick Rubin and David Morales to the Latin Rascals. First I had an MCI reel-to-reel tape recorder, then a Studer tape machine, then the SLL E-Series console. I had all the synthesizers. In 1985, I did the Hall & Oates album, *Big Bam Boom* and they gave me their Synclavier, which we actually didn't use much. I had Akai delay units, MXR delays and a rack of PCM42s and Neve EQs — there was a lot of AMS stuff too.

Didn't you have an Emulator at one point?
For sure, the Emulator was relatively inexpensive compared to the Fairlight. I had my own studio but also worked at Unique Studios. The guy that owned it was called Bobby Nathan, and he literally bought every piece of gear that came out. He had the E-MU Emulator I and would sample James Brown shouts. He was actually the first guy to sample in that way; going through records and sampling shouts and comedy bits. When I did *I.O.U.* by Freeez, we used his Emulator to sample the vocals. It was an important piece of gear, as was the AMS delay unit, because you could grab a snare sample and trigger it off of tape.

Would you listen to other innovative records and try to find out what gear they were using?
I started using the Oberheim DMX because I liked some of the records that Trevor Horn was doing. That was a drum machine that sounded like real drums, whereas the 808 sounded nothing like real drums. The Rockers Revenge track *Walking On Sunshine* was all DMX drum machine. I remember when Bobby Nathan came back from NAMM and told me that Roland had discontinued the 808 and made a better one called the 909. But it wasn't really successful at the time; by 1986 it was already cheap.

What did you think of Roland's 808 reissues?
The latest 808 that Roland did — after the horrible one with all the fucking lights and shit — was pretty damn close to the original. The only bizarre thing is how they quarter-tuned the cow bells down. If I have a gig just round the corner, I'll take the old 808, but if I'm taking a flight I'll take the little one.

Did you embrace digital technology when that started to come out?
I had an analog SSL desk and moved that shit a couple of times, which is nuts. I moved it to one studio, but the guy I partnered with was a nightmare. Then I bought a loft in Jersey City and put my studio in there. I really wanted to get Larry Blackman of Cameo in the studio and knew an engineer that worked with him. He told Larry that I'd used a Sony 32-track digital recorder, so they rented one and got it up the elevator. Unfortunately, I had a doorway in my studio and it wouldn't make it through, so I went out, got a sledgehammer and knocked the fucking frame off it to get it in. Larry worked on it there for a couple of hours; then he said, "I'm not sure about this, I've lost the vibe."

What about softsynths?
Yeah, of course man! To be honest, I never got my head around things like Reason. I tried and had all those things for sure, but when I first started making records with software I used ACID Music Studio. I made some amazing things with that, but it wasn't for Mac so I had to buy a PC. To me, ACID was revolutionary, it was the predecessor to Ableton Live I guess; it was easy to loop but had no MIDI. At Shakedown we got into Logic, and I've stuck with that the whole time — I never got into Pro Tools.

A lot of artists are going back to having hardware setups, perhaps as a reaction to software. Do you empathize with that?
When I see people with studios that have 30 synths, I always think they must have a lot of time on their hands. Listen, I kept a lot of my synths — I never sold one. The only ones I got rid of were the ones that got stolen. It's really cool if you have the synths and the time to use them, but what's more worthwhile, in my opinion, is having a really good player. I recently used a Juno-60 again, which is great. The string sounds on that are just like the The Pet Shop Boys. You don't have to work much with them to get those great sounds! ■

ARTHUR'S GO-TO GEAR

ROLAND TR-808
"What really needs to be said about this? Go and watch my 808 documentary on iTunes. The original box is still the greatest. While people have often tried to recreate them as plugins, the true sound from this monster is very near unbeatable."

THE MINIMOOG
"The first synth I used in a studio back in Boston in the mid-'70s. Phat as fuck, the bass on this bad boy has driven funk and disco from the '70s right up until today. White noise handclaps and stacked synth horns also got plenty of use on my early tracks."

ROLAND JUNO-60
"The first polyphonic synth I purchased was the Juno-60, closely followed by the Oberheim OB-8. The strings on the Juno defined early electro. Listen to those strings today and it's instant New Order and Pet Shop Boys."

OBERHEIM OB-8
"Right now I'm fixing up mine so I can use it on my next session."

LOGIC
"At Shakedown we got into Logic, and I've stuck with that the whole time — I never got into Pro Tools."

EASY GUIDE

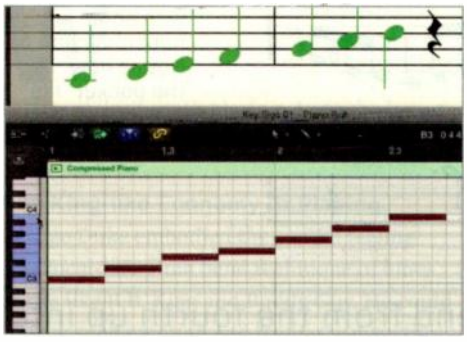
Key signatures

Every tune's got at least one, so shouldn't you know how they work? Our resident theory buff will enlighten you

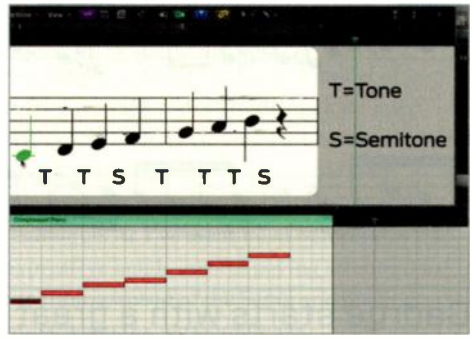
If you've ever tried to read a piece of sheet music, you'll know how important the key signature is. For the most part, every piece of music (in Western tonal music, at least) has a key that represents its tonal center. The key signature is a symbol found at the beginning of the stave, between the clef and the time signature, that lets the player know at a glance what key the piece should be played in. Easily identifiable once learnt, the symbol shows which notes in the piece need to be altered in pitch – in other words, sharpened or flattened – to make the piece conform to that key. In practice, one sharp or flat placed on a line in the key signature implies that every note on that line will be modified, and the sharp/flat symbol needn't be repeated.

You could be forgiven for thinking that this kind of thing is only relevant to people who read sheet music, but being able to identify the key of a piece goes much further than that. One example is Logic's use of Apple Loops, which not only conform to the tempo of a piece, but to the key as well, so if you have a good idea of the key of your track, you'll be able to choose loops that work better because their original key is closer to that of your project. Confused? Well, hopefully you won't be after following the steps below.

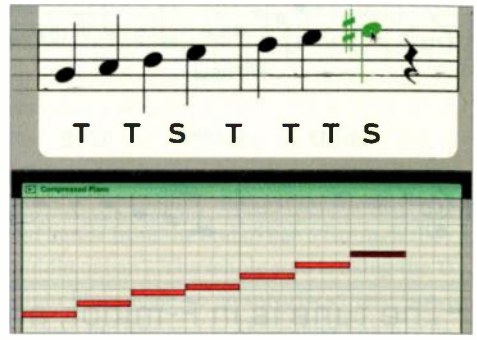
Step by step Unlocking key signatures



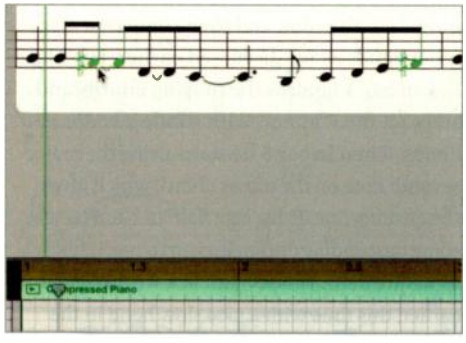
01 Time to meet our old friend the C major scale once again – C, D, E, F, G, A and B. This scale is unique amongst major scales because it's the only one that has no sharps or flats in it – every note of the C major scale is one of the white notes on the piano keyboard. Because there are no sharps or flats, it doesn't need a key signature.



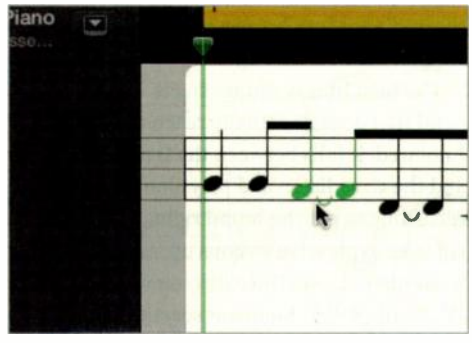
02 What makes the scale a major scale is that it obeys a particular pattern of intervals between the notes that defines it as a major scale. This pattern, shown here, is T-T-S-T-T-T-S, where T represents an interval between notes of a whole tone, and S represents an interval of a semitone.



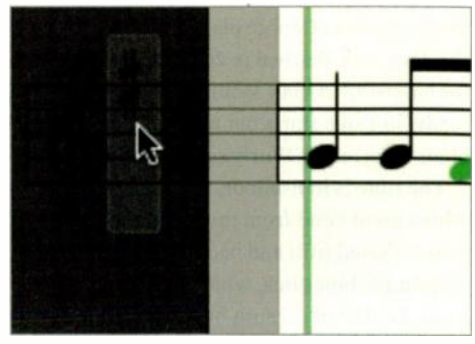
03 Now let's look at a different major scale – G major this time. You can build a G major scale by applying that same pattern of intervals we've just seen, but starting on the note G instead of C. Having done this, we can immediately see one major difference – we've ended up with one sharp note in the scale, the seventh degree is an F#.



04 So, here's a bit of music in the key of G major. If we look at the score, we can see that, as you'd expect with a piece in this key, it contains a fair few F# notes. With all those sharp symbols, it looks untidy and confusing. What if there was a simple way to tell the player that the key of the piece is G major, so therefore every F in the piece has to be played as an F sharp?



05 This is where the key signature comes in! By removing the sharp symbols from the notes themselves and placing just one at the start of the piece, in between the treble clef and the time signature, the player knows that the key is G major and that all of the Fs are to be played as F#s (unless marked otherwise).



06 So this single sharp symbol at the beginning of the tune is the key signature, and its purpose is to indicate to the player the main key the music is in. It's written so that the inside square of the sharp sign is directly over the note's position on the stave, and it shows that sharps are to be applied to every instance of the note, in all octaves.

Recommended listening

Beyonce, "Love On Top"
Every tune has at least one key signature, so I've picked a couple that have loads! "Love On Top" runs through a selection of them during the outro choruses.

bit.ly/BeyoLOT

Dam'nco, "French Kiss"
This track not only features outstanding musicianship but also key changes aplenty from drummer Damien Schmitt's jazz fusion ensemble.

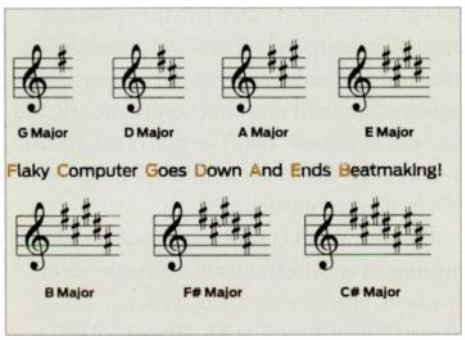
bit.ly/damncoFK

Pro tips

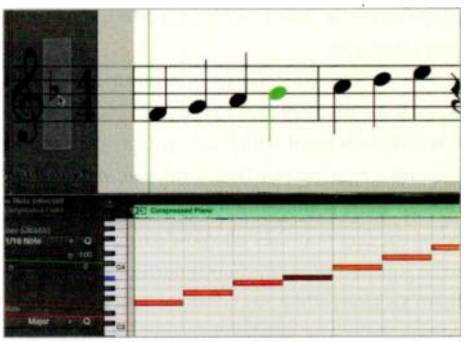
Quick ID Technique
To quickly identify a key with a sharp key signature, just figure out the last sharp note in the key signature, the rightmost one, then go one note higher than that to get the key. For keys with flat key signatures, just identify the second-to-last flat from the right in the key signature to get the name of the key. This flat trick has one exception: F major has only one flat (Bb), so you just have to remember that one!

Bass clefs
Key signatures on bass clefs work just the same as those on treble clefs, they just appear in a different vertical position on the stave as dictated by the fact that it's a different clef. Whereas the lines on the stave in front of a treble clef, from the bottom to the top, read E G B D F, the lines on a stave in front of a bass clef read G B D F A. So for the key of G major in the bass clef, the sharp will appear on the fourth line from the bottom.

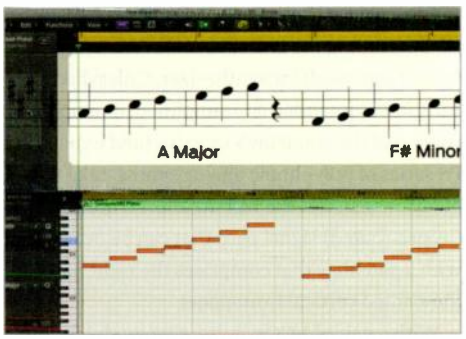
By Dave Clews
In his 25 year career, Dave has engineered, programed and played keyboards for numerous artists including George Michael and Tina Turner



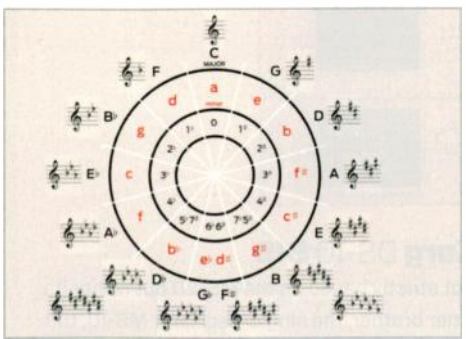
07 Here are the key signatures for G major, D major, A major, E major, B major, F# major and C# major. The order that the sharps appear in the signature is always the same — F#, C#, G#, D#, A#, E# and B#. To remember this, you can use my personal mnemonic — Flaky Computer Goes Down And Ends Beatmaking!



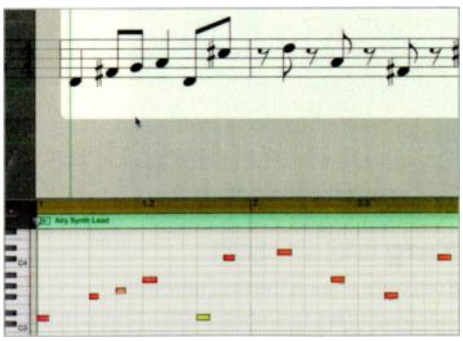
08 But that's not all the available keys, I hear you cry... what about the others? Well, certain keys contain flat notes, rather than sharps. Take F major, for instance. It contains one flat — Bb — so the key signature for the key of F major also contains one flat sign, centered on the B. You have two flats (Bb and Eb) for the key of Bb major, three (Bb, Eb and Ab) for Eb major, and so on.



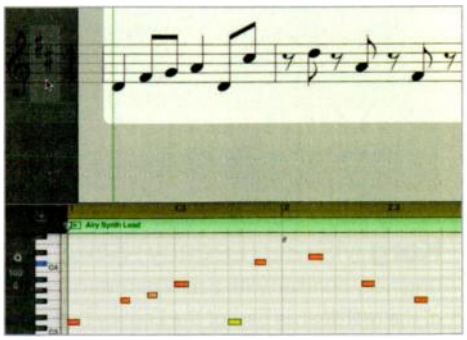
09 So far, we've only looked at signatures for major keys, but what about minor keys? Each major key shares the same key signature as its relative minor key. So for example, A major and F# minor both use three sharps. To find the relative minor of a major key, just count three semitones down the keyboard from the major key's root note to find the root of the minor key.



10 Alternatively, you could consult the circle of fifths. Key signatures are an important ingredient in the circle, radiating out from the top center position in both directions, adding either a sharp or flat to the signature with each key, Here's a diagram, showing the shared signatures for both major and minor keys.



11 Finally, here's a quick quiz to see if you've been paying attention! Check out the lead synth line in this piece. There's no key signature at the moment, so let's look at the accidentals in the melody and consult the circle of fifths to try and work out what key it's in. We've got several F#s and a scattering of C#s. What scale contains these notes?



12 If you guessed D major, congratulations! If we select this as the key in our DAW's project settings, the two sharps appear magically in the correct position at the beginning of the stave, and all the accidentals on the individual notes disappear. We've correctly identified the key!

BLAST FROM THE PAST

Korg MS-20

A semimodular with a big sound, this synth icon laid dormant until the time was right for a comeback

TECH SPECS

Years produced
1978 - 1982
Original sale value
$750
Current price
$1000-$1500
Number made
Approximately 20,000

While other big-name manufacturers steadfastly resist delving into their own pasts, Japanese instrument manufacturer Korg has heeded the pleas of electronic musicians everywhere. The company's recent hardware offerings certainly have a vintage vibe to them, even drawing on retro products from rival companies. Korg was also among the first to embrace the idea of virtual recreations of its own classic kit, producing a stream of legendary Korg instruments recreated for both desktop and mobile devices.

One ancient Korg stands proudly over them all, demanding the most attention: the legendary MS-20 semimodular synthesizer. Unleashed in 1978, it was intended as a simplified, monophonic version of the company's massive (and expensive) PS series of polyphonic analog synths. Like those bulky behemoths, the MS-20 offers a normalized (ie, pre-routed) signal path that can be overridden with a patchbay that comprises the right half of the slanted vertical front panel.

With a pair of VCOs, noise, ring modulation, LFO, and a pair of envelope generators, this diminutive dynamo offered enough to impress even without that modular-style patch field. Add to that a pair of 12dB filters (high- and low-pass), each with rip-snortin' resonance (called "peak" here), and you have an instrument with a decidedly edgy character. The MS-20 really shone in the hands of purveyors of industrial and aggressive dance styles, and experimentalists embraced its ability to bypass the standard signal path to create unconventional timbres – Aphex Twin (Richard James) and The Legendary Pink Dots' Phil Knight are longtime proponents, the latter still hauling his out on tour year after year.

Savvy readers may have noticed that the MS-20 bears some resemblance to the ARP 2600 and the EMS VCS3, both of which are vintage icons in their own right. While they don't sound the same, they all share some common features, not least being the fact that they all allow external signals to be shunted through their innards, making them an interesting candidate for processing, say, guitars, drums or vocals. This external processing is one of the MS-20's signature tricks, allowing sounds to run through its characterful filters.

Something of a dinosaur even at the time of its release, the MS-20 sold respectfully well for a few years and then vanished from the catalogue. Yet enough were made to keep it in the public eye, increasing desire and secondhand prices among aficionados of audio esoterica. When that desire coalesced into a frenzy, Korg heeded the call. You can now buy a new "mini" version of the MS-20, or even a full-sized kit in addition to Korg's own official plugin emulations.

And why wouldn't you? The MS-20 is an ideal instrument on which to learn the arcane art of synthesis, but more than that, it has a sound all its own. Bright, brutal, brash – the sound of this brilliant blast from the past translates perfectly into today's musical styles. Let's hope it sticks around this time! ■

Three great emulations

Korg MS-20 (Legacy) $50

Originally bundled with a mini hardware replica of the real deal as part of the Legacy Collection, you can now get Korg's own virtual VST/AU/RTAS version of the MS-20 on its own. Fully patchable and with modern niceties like unison and MIDI sync, this is the real deal by the company that invented the thing.
www.korg.com

Korg iMS-20 $21

Korg were among the first hardware manufacturers to embrace iOS – first with their stunning iElectribe and then with this, a recreation of the MS-20 that throws in a virtual version of the SQ-10 analog sequencer. Additionally, there's a wicked MS-20-based drum machine. Awesome fun on the fly.
www.korg.com

Korg DS-10 $19

Not strictly based on the MS-20 but rather its littler brother, the single-oscillator MS-10, the DS-10 has the distinction of packing an awful lot of power into what amounts to a plaything, the handheld Nintendo DS game system. The DS-10 offers convincing sound, built-in sequencing and X/Y control.
www.xseedgames.com

ALL ABOUT THE BASS
EVERY FOUR WEEKS

NEW ISSUE
ON SALE NOW

Order online at **www.myfavoritemagazines.com**

Yamaha

CP88

$2,499

yamaha.com

By Ben Rogerson

Ben is the Group Content Manager of our sister website MusicRadar, and a piano and keyboard player of more than 30 years

Strengths

+ A versatile, high-quality soundset for the gigging musician
+ A straightforward and logical interface makes the CP88 very easy and rewarding to use
+ Natural wood, graded hammer-action keyboard is a joy to play

Limitations

- The organ and synth sections are lacking in comparison to some of its rivals
- It can be difficult to see where the controls for each section begin and end on the panel
- We'd have liked to have seen a matching stand

Yamaha's new stage piano was one of the sleeper hits of NAMM 2019, but it faces stiff competition from all sides

The clue's in the name. A stage piano is something that you're going to be seen with and is, you might argue, a reflection of the kind of player you want to be seen as. We wouldn't go quite as far as to say that it's an extension of who you are — we'll leave that kind of histrionic talk to the guitarists — but there's certainly a perception issue here.

This is where Yamaha has been presented with a problem in recent years. Yes, it has produced some fine and functional stage pianos. but other manufacturers — notably Nord — have gained a reputation for building the live keyboard instruments du jour. Go to a festival or watch any kind of live music television and chances are you'll see one of their bright red beauties popping up — and this represents the kind of branding that money can't buy.

It's into this climate that Yamaha is launching its new CP pianos, the name being a reminder that Yamaha itself has a pretty strong pedigree in the electronic piano market. Anyone who's heard a Phil Collins record will be familiar with the seductively synthetic tone of the classic CP pianos, the sounds of which have come into favor again in recent years.

There are two models in the new range — the CP73 and the CP88 - and it's the larger of the two that we'll be considering here. Aside from the number of keys (73 and 88, in case you hadn't worked that out) the pianos are functionally very similar, though there is some difference in the keybed, which we'll get to later.

The CP88 weighs in at 41lb, which is pretty reasonable for a piano of this size. The box is big and awkward, as you'd expect, but it's just about a one-person job to move it, and I managed to lift the CP onto a stand without much trouble. As such, it ticks an important box for gigging players — it's reasonably portable — though it would have been nice to see a matching stand included.

Build quality is impressive, too. As I've already mentioned, Yamaha has been making instruments like this for many a long year, so it knows a thing or two about putting them together. The case is made from aluminium, which helps to keep the weight down, and the CP feels durable and well set for life on the road. Another box ticked, then.

First impressions of the interface are that it's simple, clean and well thought out, with a couple of subtle touches immediately catching your eye. Firstly, there are the yellow, red and green rocker switches that are used to scroll through the instrument variations in the Piano, Electric Piano and Sub sections — a nod (I assume) to Yamaha's classic CS-80 synth, and another reminder of the company's illustrious past. You'll also notice the metal switches that are used to turn each of the aforementioned sections, and also the master Delay and Reverb, on or off. There's definitely a vintage vibe here, and the CP88 wears it well.

One slight criticism is that, when you need to make hands-on tweaks at speed, the subdued, one-color interface can make it a little difficult to see where one section ends and another begins,

and this problem is exacerbated in low-light conditions. I like the subtle styling, and the lights round the knobs make assessing current level settings easy, but a more obvious visual separation of sections would've been welcome.

That said, the CP88's control setup is actually one of its key selling points, giving you one-to-one access to all of its key features. There's no menu diving here, with each dial, switch and button being clearly labeled. Assigning keyboard splits and adjusting octaves is all very self-explanatory, and you may never even need the manual. You know where you are with the CP right away.

To the sounds, then, and the obvious place to start is with the acoustic grands. Yamaha has gone all-in here, giving you emulations of not just its own CFX and S700 models, but also a Bösendorfer Imperial 290. Like all of the sounds here, they're powered by Yamaha's AWM2 sampling engine, and they cover all the necessary grand piano ground. The CFX and Bösendorfer are great if you want a big 'classical' sound, while the S700 works particularly well in a pop/jazz setting, cutting through nicely in a mix. As with all the sound sections, there are dedicated Volume and Tone knobs and dedicated effects — these vary from section to section — and you can switch on the Damper Resonance function for realism.

It must be said that the playability of all these pianos is helped by the superb keybed. This is Yamaha's Natural Wood, Graded Hammer, triple-sensor keybed, which also boasts Synthetic Ebony and Ivory Key Tops (the CP73's keyboard has an electric piano-style 'balanced action'). It looks and feels fabulous, responding superbly and handling fast runs with ease. There's something extremely rewarding about the way this handles, almost to the point that it makes you feel like a better player than you actually are.

> "You may never even need the manual. You know where you are right away"

There's more acoustic piano goodness in the uprights section, which features Yamaha's U1 and SU7 models. As you'd expect, there's an excellent CP80 emulation here, too, and the Pianos section is rounded-off by some layered piano/strings sounds for big ballad moments.

Move to your right and you'll find the E. Piano area, which gives you three flavors of Rhodes (76 Rd, 75 Rd and 73 Rd), Warm and Bright Wurlies, Clavs, a harpsichord and — of course — several varieties of DX piano. These can all be clean and polite or filthy and funky — a Drive control and some excellent modulation effects are particularly useful when you need to dial in the dirt. Even if you can't play like Stevie Wonder in his '70s pomp, you'll have the sounds in your arsenal to make you wish that you could.

The final sound section, Sub, covers everything else, including pads, strings, a variety of organs, chromatic percussion and a selection of synth sounds. These are all perfectly serviceable, if not showstoppers. Again, the effects are the ones you want to see — a rotary speaker emulation, distortion, tremolo and chorus/flanger.

Speaking of effects, the Master Delay is worth a special mention, giving you a lovely warm analog variant and a crisp digital algorithm. For both this and the Reverb, you can adjust the send levels individually for each sound section. A master EQ, which works globally, completes the control panel.

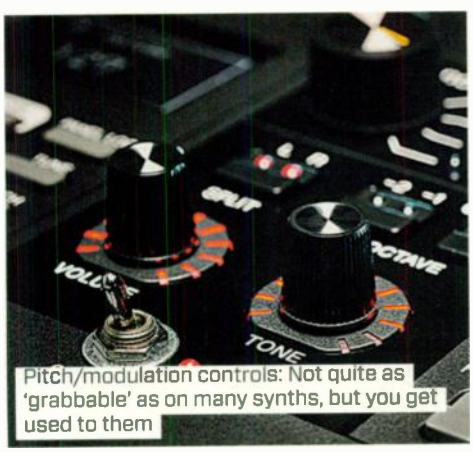
The CP88 is a genuine one-stop stage keyboard solution, but if you do want to bring a second keyboard into the mix, the L/Mono and R audio 1/4-inch jacks give you an easy way of doing it, potentially eliminating the need for a mixer. A Gain knob enables you to adjust the volume.

Let's talk about (Live) Sets

You can use each of the CP88's sound generation sections on its own, but if you want to create monster patches that combine Piano, Electric Piano and Sub tones — as well as the Delay and Reverb effects — Live Sets are the way to go.

These are accessed via the panel of buttons to the left-hand side of the interface; there are eight selector buttons, plus a further two to let you navigate the 20 pages. You can also zip through the stored Live Sets quickly by turning the push button encoder. A wide range of Live Sets comes included, with some featuring just one sound, and others a full stack. These do a fine job of demoing the CP88's capabilities, and can be adapted to taste (you can see which sounds are coming from each section via the display). Storing, copying and moving Live Sets is simple.

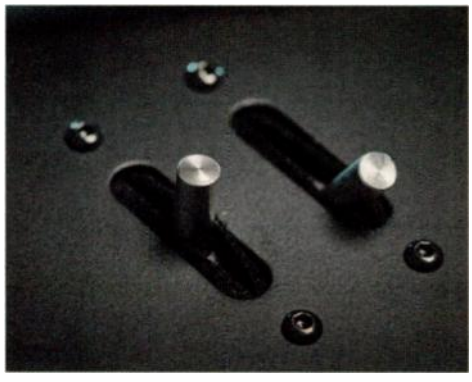
What's more, there's Seamless Sound Switching between them, meaning that you can select a different Live Set while the previous one continues to be heard. This is another useful performance feature, and will come in particularly handy when you need a fast patch change between songs.



Pitch/modulation controls: Not quite as 'grabbable' as on many synths, but you get used to them


Yamaha's Natural Wood, Graded Hammer, triple-sensor keybed puts the CP88 in quintessential electric piano territory

Sound and effect interface: Totally self-explanatory, all round. No need for the manual



Other connectivity options include balanced XLR L/R outputs, unbalanced 1/4-inch outputs, and a 1/4-inch headphones socket. You can plug in a sustain pedal and a further assignable footswitch, with two further Foot Controller inputs letting you control Expression, Wah, etc. MIDI I/O comes on standard ports and over USB. A further USB Device port means you can load and save data to a flash drive.

As I implied at the start, the CP88 has some serious competition, and you could argue that, when set against the Nord Stage 3, it does fall short in some areas. It can't rival the Nord's A1 Synth Engine and user sample import, and the organ section isn't as comprehensive, either. If you simply must have drawbars — physical or digital — the CP isn't for you. But there's something about the CP88 that just makes you want to play. It passes the Marie Kondo test in that it sparks joy when you sit down in front of it, and that has everything to do with the one control per function interface and logical layout. By paring things down to the essentials, Yamaha has made less feel like more.

The CP88 can be had for over a grand less than an 88-note Nord Stage 3 — and some hundred dollars less than the Nord Piano 4, too. Yes, there are other options to consider — Korg's GrandStage 88 or Roland's RD-2000 — but the CP is right there in the mix. ■

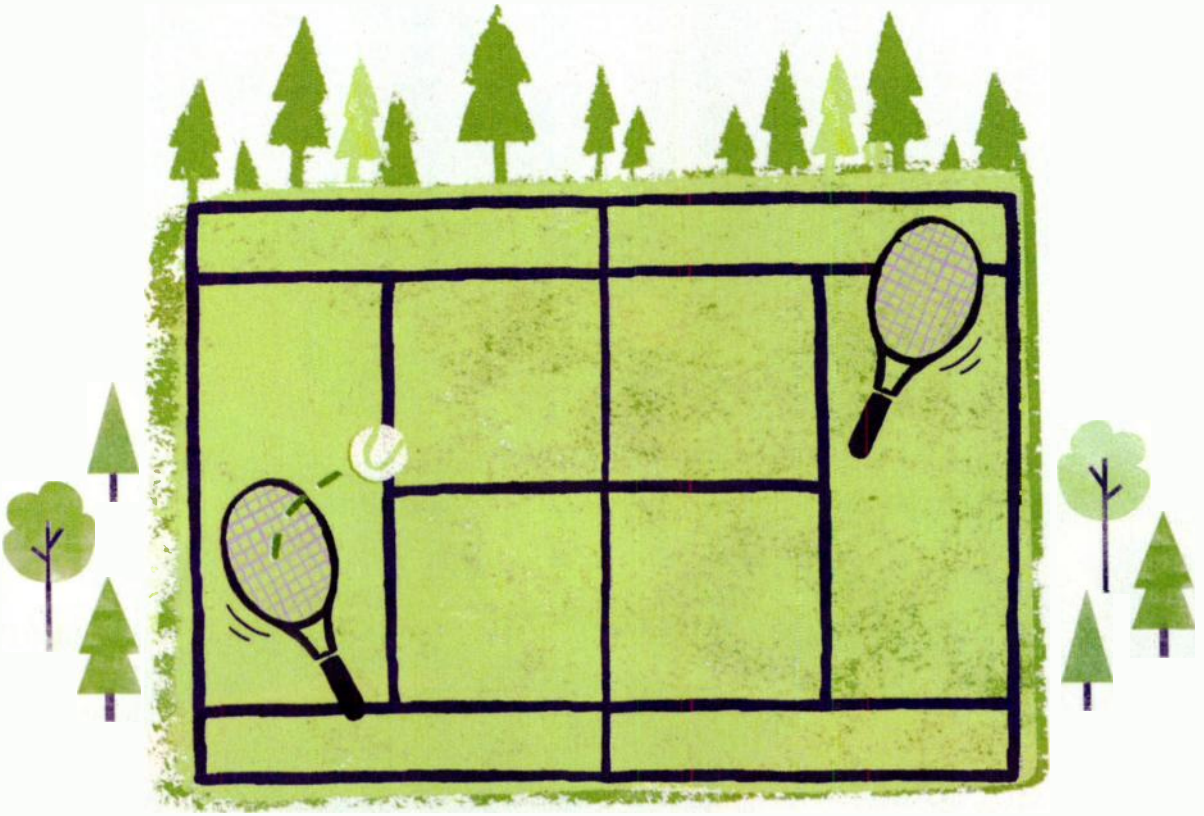
THE ALTERNATIVES

NORD
Stage 3
$4,499
The latest version of Nord's high-end gigging keyboard is a belter. You can now load up to 2GB of sounds from the excellent Nord Piano Library, and you get both the excellent Nord Lead A1 Synth Engine and Nord C2D Organ Engines built in (there are physical drawbars on the Compact model and digital ones on the HP76 and 88 variants). The Rolls-Royce of stage pianos, but it's priced accordingly.
nordkeyboards.com

KORG
GrandStage 88
$2,199
Priced similarly to the CP88, the larger of Korg's two GrandStages also has an intuitive, performance-friendly interface, and comes with an excellent set of sounds that are derived from the Kronos synth engine. A well-built, versatile instrument, it has the added bonus of an included stand.
korg.com

22 THOUSAND

TENNIS COURTS EVERY DAY!



Did you know that U.S. forests, which provide wood for making paper and many other products, have been growing every day, equal in area to more than 22,000 tennis courts!

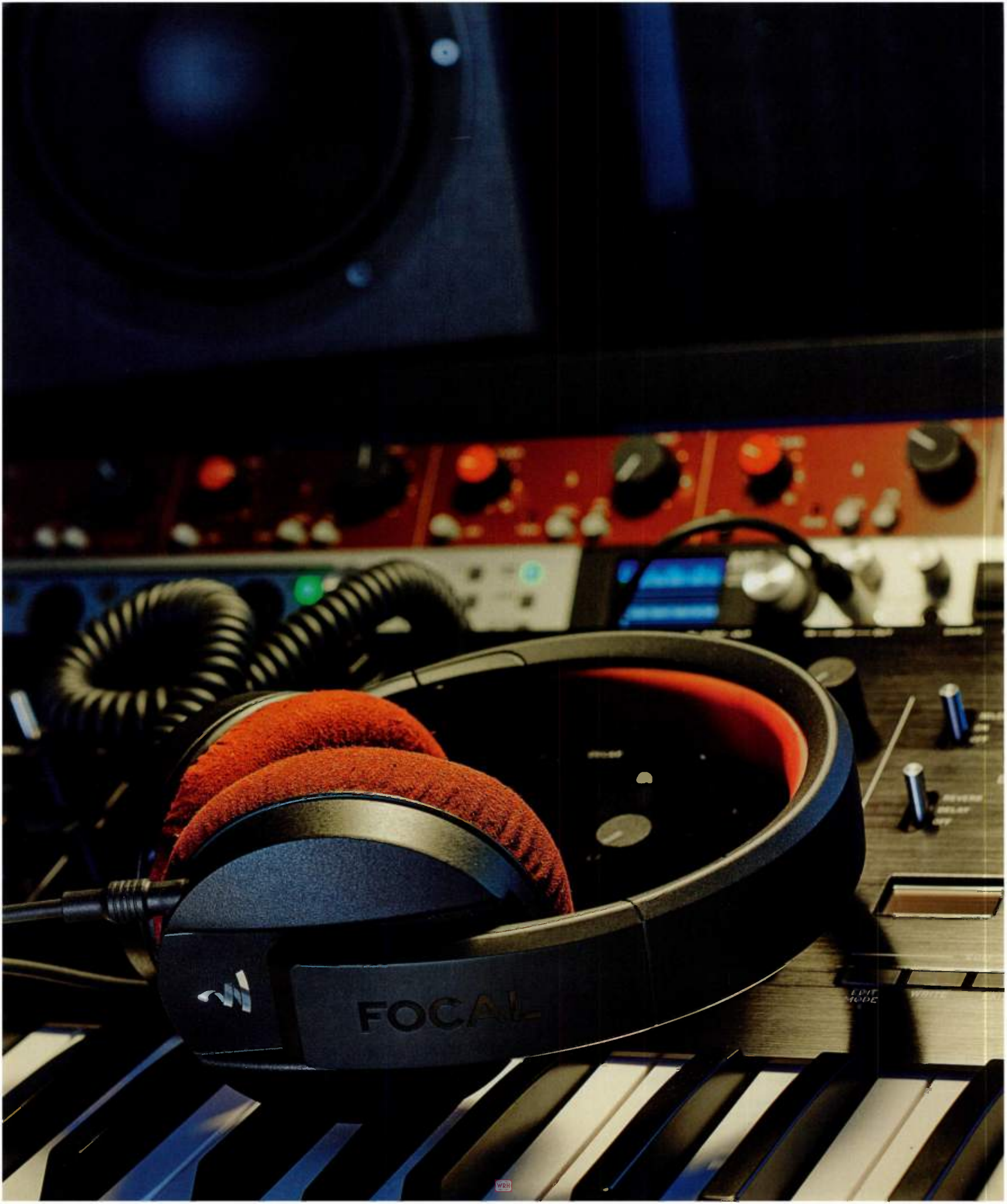
Love magazines? You'll love them even more knowing they're made from natural, renewable and recyclable wood.

UNFAO, Global Forest Resources Assessment 2005-2015

Two Sides is a global initiative promoting the responsible use of print and paper which, when sourced from certified or sustainably managed forests, is a uniquely powerful and natural communications medium.

Discover more great #PaperFacts at **twosidesna.org**

TWO SIDES


FOCAL

Focal Listen Professional

$299
focal.com

By Simon Arblaster
Simon Arblaster is music technology Content Editor for our sister website MusicRadar

Strengths

+ Sturdy and comfortable fit
+ Solid sound reproduction at low volumes

Limitations

- Silicon cushion on the headband, though comfortable, will attract sweat

Focal's Spirit Pro headphones were excellent. Does the spirit of those forebears reside within this new set?

The Listen Professional headphones from Focal represent the only closed-back phones in the French company's 'professional' range, replacing the now defunct Spirit Pros. I was really impressed by the all-round performance of the Spirit Pro phones back then, with, perhaps, the only misgiving being the fit. So we were pleased to see that the design of the Listen Pros has changed somewhat when compared to its predecessor, but of course the real question is, how do they feel and how do they sound?

Fresh out of the box and we find the LPs neatly encapsulated within a rigid case, which also includes a 5m coiled cable and a 1.4m cable, with omni-directional microphone and remote control. Of course, the obligatory jack size adapter is present, as is a handy little elasticated strap for your iLok dongle, which is a thoughtful touch.

The design differs from the Spirit Pros quite significantly. The headband and hinge looks to be more robust than before and will withstand any twisting, or bending. The folding mechanism differs from the Spirit too, with one cup folding after the other, not at the same time with the earcups pushing against one another, which could damage cones.

The overall fit feels like a marked improvement on the Spirits; these are some of the more comfortable phones we've tried. This is largely to do with the memory foam earcups, which provide the right density for both comfort and isolation. The headband grip is firm, yet reassuring. The low weight and cup-profile ensures long hours of monitoring and mixing are possible and you wouldn't bat an eyelid at wearing them out and about. We're loathe to use the word 'stylish'... but we just did.

Being a closed-back design means you're not going to get the most expansive soundstage that an open-backed design can provide, but next to its closest rivals, the Listen Pros are honest, direct and the soundstage is very pleasing. Those memory foam earcups aren't just for comfort and isolation, as Focal has deployed certain materials as acoustic treatment, which further enhances the level of clarity and detail that these headphones provide, leaving you with no unwanted frequencies bouncing around, getting in the way. Reproduction across the entire spectrum is transparent, with plenty of excitement down the low end – the lower mids and bass are both tight and lusciously rich.

The Focal Listen Professionals, are exactly what they say on the tin; ideal for both listening and pro duties. As a result, we'd put them firmly in the all-rounder category. Overall, the fit is snug, but not too tight that long periods in the mixing saddle are unbearable. In fact far from it, the longest period of time we had them on continuously was for around three hours – with little discomfort or need for a break. While many engineers and audio specialists may prefer open-backed designs for their lack of fatigue, the closed Listen Pros perform admirably in this regard. Highly recommended. ■

4ms POD

$55-$175
4mscompany.com

By Gino Robair
Former EM editor Gino Robair has written two books on music technology, scored music for film and TV, and recorded with Tom Waits and Thurston Moore, among others

Strengths

+ Lightweight
+ Well built
+ Inexpensive
+ DIY friendly

Limitations

- Shallow case does not support all Eurorack modules

The lightest, most portable Eurorack cases available

Eurorack developer 4ms has made it easier than ever to get into modular synthesis. Housed in anodized aluminum, their Pod enclosures are lighter and shallower than other portable cases, and they are available in four sizes based on HP width — 20, 26, 32, and 60 — in powered ($99, $109, $119, $175, respectively) or unpowered ($55, $60, $65, $85) versions.

Most Pod enclosures weigh under 8 ounces, with the powered versions adding an extra ounce (not including external power supply). Each Pod comes with stainless steel M3 screws and adhesive rubber feet for the bottom of the case.

The powered Pod 60 has four internal power headers whereas the other three models have only two. If you want to plug in additional modules, attach a multi-power cable ($6) to one of the headers. The internal headers are keyed to keep you from connecting modules incorrectly.

Powered Pods require a power brick (sold separately). They support any universal laptop supply with a 2.1mm positive tip that provides 15-20V, but I recommend the 4ms 45W Power Brick ($15), which gives you 3 Amps and automatically handles 100V-240V (50-60Hz) input — perfect for international travel. Moreover, the powered Pods have a secondary power jack for daisy-chaining other powered Pods from the same Power Brick using the company's 2.1mm barrel-connector cable (available in two lengths; $15 and $16). A light inside the case indicates that it is receiving power.

With the unpowered Pods, you can use the 4ms Row Power module ($119-$155) and a flying bus cable or distribution board to connect your modules. I have a Nono Rover 1.60 portable case that uses a Row Power system and it has worked flawlessly. Row Power modules can also be daisy chained using a barrel connector.

Even packed with modules, the Pods are remarkably light and fit easily into a carryon bag (though you'll need to improvise a case to protect any knobs and switches on the modules, themselves). Fully loaded, my Pod 60 weighed 2 lbs., providing the easiest Eurorack travel experience I've had to date.

An important aspect of using the Pods is finding modules that fit the 1.34" (34mm) depth of the case. Recent 4ms modules fit without issue, but many Eurorack modules have circuit boards that exceeded the depth. You must also take into consideration space for the power cable attached to each module: These ribbon cables need to fit between the module's boards and the bottom of the Pod. (As this went to press, 4ms announced the powered Pod40-X [$145] and Pod48-X [$155] which are 2" [51mm] deep.)

As I wanted to bring my 4ms Spectral Multiband Resonator on tour along with my Pod 60 setup, I bought a powered Pod 26 (4ms's SMR module is exactly 26HP). Then, I powered it from the Pod 60 using a barrel cable — very convenient.

In addition to being compact and travel-friendly, the 4ms Pods provide a low-cost way to get started in modular synthesis — especially the powered models. Overall, anyone interested in adding Eurorack to their tabletop setup should give the 4ms Pods a serious look. ■

Erica Synths

Black Hole DSP 2

$2,499

ericasynths.iv

Strengths

+ Manual and CV control over most parameters
+ Easy to save and recall presets
+ Crush
+ Wet/dry control

Limitations

- Higher noise floor than studio-based processors.

By Gino Robair

Stereo multi-effects processing for Eurorack

Part of the company's Black Series, the Erica Synths Black Hole DSP 2 (BHD2) is a stereo Eurorack module that provides 24 time- and pitch-based effects (eight more than the original) as well as some useful new features. Like the original, the BHD2 gives you manual and CV control over three parameters for each effect, as well as Patch selection.

The new Crush feature allows you to alter the sampling rate of the DSP from 48kHz down to 1kHz. Beyond adding distortion-like artifacts, Crush often changes the behavior of an effect in surprising ways, and having voltage control over it greatly increases this module's usefulness.

Designed for live performance, the BHD2's user interface is straightforward. Keep turning the Patch encoder until the number of the effect you want is shown in the LCD, then press it to load the patch.

The levels for each of the three parameters in a patch can be saved and recalled manually or with CV. A simple visual system helps you reset the knob for each parameter if you want to.

You save settings by pressing and holding the Patch encoder until you see SA in the LCD. To recall the saved settings, select the effect number and press the encoder twice so that RC is displayed. Then, turn each Parameter control until you see the dot in the LCD flash: That's the original level of that parameter's knob.

The BHD2 includes 9 delays, 7 reverbs (3 with shimmery pitchshift parameters), a chorus, a flanger, a phaser, the Ripper bit-crusher/distortion, a dual pitchshifter, two filter-based freeze effects, and a three-oscillator Drone Bank, which is not dependent on an input signal. Although different aspects of each effect are assigned to the three parameter controls, the majority of them assign Feedback or Pre-delay to parameter 1 and Delay Time or Reverb Size to parameter 2.

Typically, Parameter 3 controls the signature part of an effect (eg, Tone, Shimmer Amount, filter amount, etc.). For example, with the LP and HP Freezer effects, parameter 3 sets the cutoff frequency for the filter, whereas parameter 1 "freezes" a sound when its knob is turned past the 12-o'clock position, and parameter 2 determines how much time is in the buffer for the frozen sound (10ms-500ms). A chart showing parameter assignments is included, but it all becomes intuitive as you work with the module.

The input level for both channels is controlled by a single attenuator, and an LED shows input clipping. (The Left input is normaled to the Right input for mono signals.) The output level is handled by an analog wet/dry circuit, which can be set manually as well as voltage controlled—another handy modulation point.

Though its output is a little noisier than a studio-based processor, overall, the BHD2 feels well-built and its UI is easy to remember. Most importantly, the effects are as musically useful as they are exciting, and in combination with the Crush parameter, the Black Hole DSP 2 is a clear stand-out as a multieffects module within the Eurorack scene. ■

Synapse Audio

Dune 3

$179

synapse-audio.com

By Ronan Macdonald

Strengths

+ Wavetable editing!
+ Dual filter with new models and effects
+ Sweeping effects improvements
+ Swarm stacking mode

Limitations

- $79 upgrade is a little pricey
- FM is still very rudimentary
- Wavetable Editor is a little laggy

This member of the exclusive supersynth club looks and sounds better than ever with its latest full version

The second version of Synapse Audio's Dune synth was a landmark virtual instrument, turning what had previously been a decent virtual analog/wavetable synth into a mainroom-filling powerhouse. While it's apparent at first glance that Dune 3 (VST/AU/AAX) doesn't mark quite as sweeping and profound a revision as its predecessor did, there's still plenty of good, meaningful new stuff here for experienced users and newcomers alike to sink their teeth into. Before we get to all that, though, a quick recap is in order.

A three-oscillator synth with superb built-in effects, arpeggiation and sequencing, Dune's two headline features are the ability to independently set Oscillators 1 and 2 to Virtual Analog, Wavetable or (rather basic) FM mode, and the incredible number of unison voices it's capable of generating: up to 8320 when running 32 stacked oscillators each in the Oscillator 1 and 2 blocks, and eight global unison voices, at 16-voice polyphony. The oscillators feed into a zero-delay feedback filter with various onboard distortion and second-stage filtering effects, and the overall sound of the thing is big, beefy, glamorous and supremely versatile. In one sense, it's a workhorse, delivering the full spectrum of 'static' and sequenced sounds, and in another, it's a character instrument with a sound very much its own.

Fundamentals summarized, then, let's move on to what's new in Dune 3...

Under the 'table

Dune 2 included a solid array of prefab wavetables, and not only does v3 almost double their number to 47, but it also introduces a full-on Wavetable Editor. Here, the currently selected waveform in the wavetable can be shaped freehand or using Line and Segment drawing tools, and manipulated in terms of harmonic (partial) volume and phase in the Additive Editor mode. Samples can be converted to wavetables, either split into a specified number of waveforms, or spread across as many as their length dictates, up to the maximum of 256. Selecting the Morph function creates a smooth transition through the wavetable from first wave to last, and other waveform operations include DC offset removal, normalizing, inversion and reversal. There's also a Formula field for generating waveforms via mathematical function input, and wavetables can be exported in WAV or Dune 3's WT format.

Also new for the oscillators is the Swarm stack type. This is a supersaw with random pitch modulation of each oscillator in the stack, yielding a dense, wildly animated signal that sounds wicked on its own and makes for a viable alternative to high counts in other stack types when system resources are running low.

Fresh filter

Another major architectural upgrade has been implemented in the filter section, which now

houses two filters and an insert effect. The filters are blended with the Balance knob, and can be routed in parallel or series, with the effect inserted before, in between or after them when in serial mode.

There are also a ton of extra filter modes – including low-pass emulations of the Roland JP-08 and Alpha Juno, and CEM 3372, as well as a 36/60dB Brickwall, the 12/24dB Moog ladder from Dune's sister synth The Legend, and a Deep Notch – plus new Formant, filter, Vowel filter, Phaser and Hard Foldback filter effects.

While we're on a frequency-shaping tip, Dune 3's EQ effect expands on the simple three-band (low/high shelves, parametric mid) design of old with the addition of a second parametric band, 12/24/36/48dB low- and high-cut filters, and a graphical editor for node-dragging adjustment of the shelving and parametric bands. Other changes in the effects department are summarized in Epic effects.

That's the standout improvements in Dune 3 covered, but a few of the less attention-grabbing tweaks are worth mentioning, too. For starters, you can now switch between two independent arpeggiators, and toggle playback with a footpedal or other controller via the new Arp Hold modulation target. Then there's the Alternate mod source, for jumping target parameters between positive and negative values; a pink noise option for the noise generator; presets for the MSEGs; four GUI resolutions, from tiny to enormous; unlimited undo/redo, even across patch changes; and a brickwall limiter on the output.

Epic effects

The EQ may have seen the most obvious improvement among Dune's effects for v3, but every one of the synth's seven signal processing modules has been blessed with new algorithms and additional modes.

The Delay gets six further types, taking the total to 11. Among them, L-C-R alternates echoes between left, right and centre, Filtered incorporates a filter, and Swing lets you dial in offset for every other echo.

The Reverb module's new Shimmer Hall and Shimmer Room algorithms are able to employ pitchshifting for strange harmonic ambiences, while Cathedral and Studio Room provide users with extreme spatial options at both ends of the size scale.

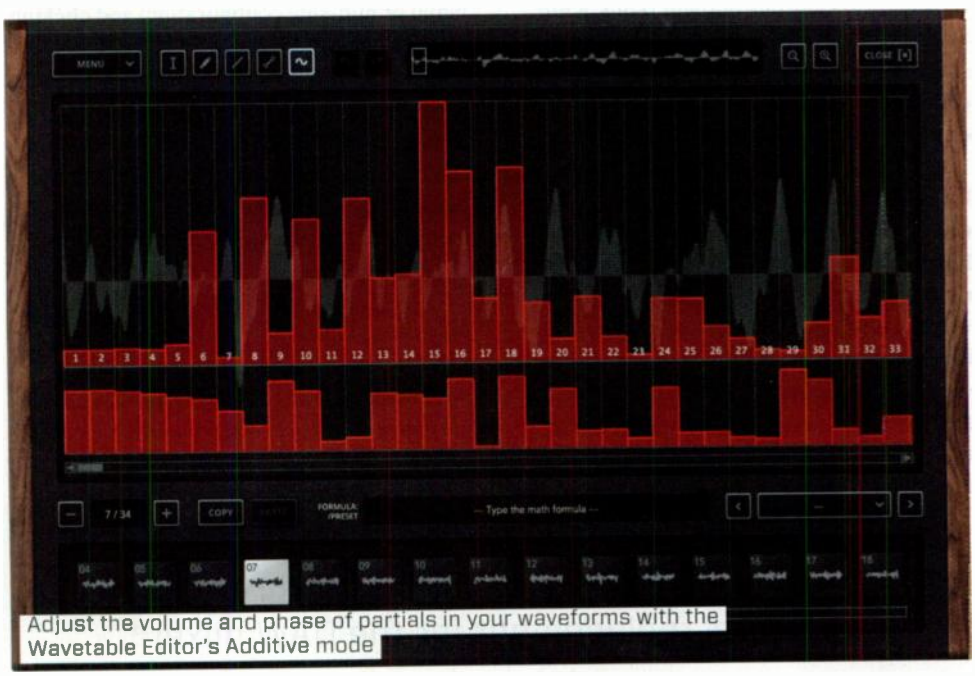
The Chorus and Phaser effects are bolstered by a range of new Roland-inspired models – specifically, the Dimension-D and JP-8000 (Chorus and Flanger, with Rate and filter parameters), and the Boss PH-2 Super Phaser pedal.

Unusual in a synth, the Opto Compressor mode is ripe for experimentation with its languid attack. And the already well-stocked Distortion module can now serve up Fuzz Face and 'Dirt' stompbox flavours, and a tasty alternative to bitcrushing with the Dynacrush algorithm.

Sand storm

While, as suggested earlier, Dune 3 could never have been as radical an overhaul of Dune 2 as Dune 2 was of Dune, this is still a thoroughly worthy update to one of the finest softsynths around. The Swarm oscillator, new filter features and enhanced effects make front-panel programming more flexible than ever, and the Wavetable Editor gives Dune parity with the likes of Serum and Thorn when it comes to wave design. The Wavetable Editor is quite laggy and slow to respond to mouse input (on our test 2018 MacBook Pro, at any rate), however. Hopefully this will be improved in a patch.

If we had to choose a single instrument to cover all our synthesis needs without making compromises in any of them, Dune 3 would be towards the top of the shortlist. It's easy and enjoyable to use, up there with the very best sonically, and – if you're buying it for the first time, at least – great value for money. ■



Adjust the volume and phase of partials in your waveforms with the Wavetable Editor's Additive mode

THE ALTERNATIVES

XFER RECORDS
Serum
$189
Mindblowing wavetable synth with incredible effects, deep modulation and so much more

U-HE
Diva
$210
Semimodular synth with gorgeous modeled oscillators and zero-delay feedback filters

Waves

CLA MixHub

$199

waves.com

By Ronan Macdonald

Strengths

+ 'Buckets' system is great, genuinely emulating a full SSL console
+ Excellent SSL-style EQ, dynamics and analogue distortion
+ Insert point for a Waves plugin

Limitations

- No mute or solo
- Can't move between Channel Views

When Chris Lord-Alge puts his name to a plugin version of his own mixing console with a novel twist, great things are expected...

The latest collaboration between Waves and Grammy-winning engineer Chris Lord-Alge is an emulation of a full channel strip from the latter's actual SSL 4000 series mixer, with a particular feature that can only be described as 'killer'. Claiming to be "the first multitrack" plugin (Softube's Console 1 system is similar in concept but very different in execution), CLA MixHub lets you access up to 64 instances of the plugin via any one of them. Thus, a full SSL console is effectively recreated in a single window within the host DAW, conveniently grouped into banks of eight channels at a time, called 'buckets'.

Bucket list

Once inserted into as many channels of the host DAW's mixer as you need (see Kicking the Buckets), any running instance of MixHub can be switched between its Channel View (ie, all of its own channel strip components together) and the grouped Bucket View, where the selected bucket of up to eight channels is flipped between grouped operation of one of said components at a time. While that's certainly the headline, however, it wouldn't count for much if the channel strip itself wasn't up to snuff – which, fortunately, it very much is. The four main component modules largely mirror their SSL inspiration, comprising Input, Output, EQ and Dynamics. There's also an Insert slot into which you can plumb any one of your installed Waves plugins. Input, EQ and Dynamics can each be set to stereo, dual mono or mid-side configuration, and clicking the Expanded View button on a module blows it up to two separate channels (left/right or mid-side) for independent tweaking, and reveals the Dynamics module's Sidechain EQ, or opens the interface of the Insert plugin. The EQ, Dynamics and Insert modules can be freely rearranged by dragging them left and right.

All module cons

The Input section has Line and Mic input level controls, each dialing in its own style of harmonic distortion when pushed hard, and Mic adding up to 50dB of gain. This distortion (and more at the output) can be

"The multitrack functionality is revelatory, feeling like a natural replacement for the DAW mixer"

disabled by deactivating the Analog button, while the Noise function models the background noise of Lord-Alge's SSL. All of this analog flavoring is very subtle. Phase invert, low- and high-pass filters, and a -20dB pad round off the Input module.

The EQ is an exact match for the SSL original, with two fully parametric mid bands flanked by high and low bands that can be switched between shelving and bell modes. Up to 15dB of cut or boost is on tap for every band, and the LMF and HMF Q controls range from sharp, precise and surgical to broad and musical.

The Dynamics section is headed up by a compressor that switches between the original SSL design and sound, and an emulation of Lord-Alge's 'blue stripe' (or "Bluey", as he calls it) Universal Audio 1176, with its lightning fast attack and alternative sound. Both models offer ratios up to inf:1, and two Attack speeds. Below that, the Gate/Expander adds ducking to the SSL setup, complete with Hold control, and a full-on sidechain EQ for highly detailed shaping of external keying signals. There's also a Dry/Wet mix control for parallel processing.

Finally, the Output module houses a bank of buttons for selecting mono or various stereo output formats, stereo widening and narrowing, a 200Hz LF shelf, and comprehensive metering.

Kicking the Buckets

MixHub's multitrack workflow couldn't be easier to operate. You start by loading the plugin onto every channel of your mix – don't worry, it's impressively light on CPU usage. You then assign each instance to one of eight 'Buckets' – each of which holds up to eight channels – either via a menu or a color-coded assignment overview. So, you have one Bucket for your drum channels, another for your vocals, one for your guitars, etc. Then, flip any instance of MixHub from the regular Channel View to Bucket View to see your choice of the four channel strip modules (Input, EQ, Dynamics or Output) for the up-to-eight component channels of the active Bucket, ready for visually grouped editing, with eight tabs along the top used to switch between Buckets. Most of the Channel View controls are visible for each module, and those that aren't (the Output format buttons, for example) can be revealed via the Expanded View button, which shows the full module for one channel – but not, alas, its full strip.

The name of each plugin's host track appears at the top of its strip in Bucket View (although we did experience the odd bug in that department, necessitating manual renaming), and channels are rearranged within Buckets by dragging and dropping.

The Bucket View of a drum kit, showing the Dynamics module for six channels

Lord-Alge and master

The first thing to say about CLA MixHub in appraisal is that it sounds superb. Waves are no strangers to big desk emulation, and this one's a beauty, exuding analog warmth, depth and focus. Just like a real SSL, the EQ is easy to work with, while the extended Dynamics section gives plenty of enveloping options. All of which is great for shaping sounds.

The multitrack functionality is nothing short of revelatory, feeling like a natural replacement for the standard DAW mixer. Jumping between Buckets and modules is fast and intuitive, and there's a real sense of empowerment that comes from being able to apply such high-end processing to the individual elements of a whole drum kit, vocal group, etc. And although changing one instance of the plugin from Bucket to Channel View or vice versa also changes all other instances, you can have as many different Bucket Views open at a time as you like. With a big enough monitor, you could conceivably get eyes-on with the whole of the console!

There are a few niggles, though. Presumably due to the limitations of VST/AU/AAX plugin implementation at this point, there are no mute or solo buttons, and having to head back out to the host mixer to hear channels in isolation does break the spell somewhat. Also not Waves' fault. you can't step between instances in one plugin's Channel View. This isn't a major problem, thanks to the Bucket View's Extended View function, but when you want to see the full channel strip for a track, you do have to navigate to its plugin. And a wish rather than a complaint: it would be great if each Bucket served as a proper hierarchical bus, with its own top-level EQ and dynamics modules for processing its assigned channels collectively, rather than just being a visual grouping.

Unavoidable technological limitations aside, however, CLA MixHub's channel-aggregating approach works very well indeed, bringing your entire mix together in a single interface, and sounding fantastic doing it. ■

THE ALTERNATIVES

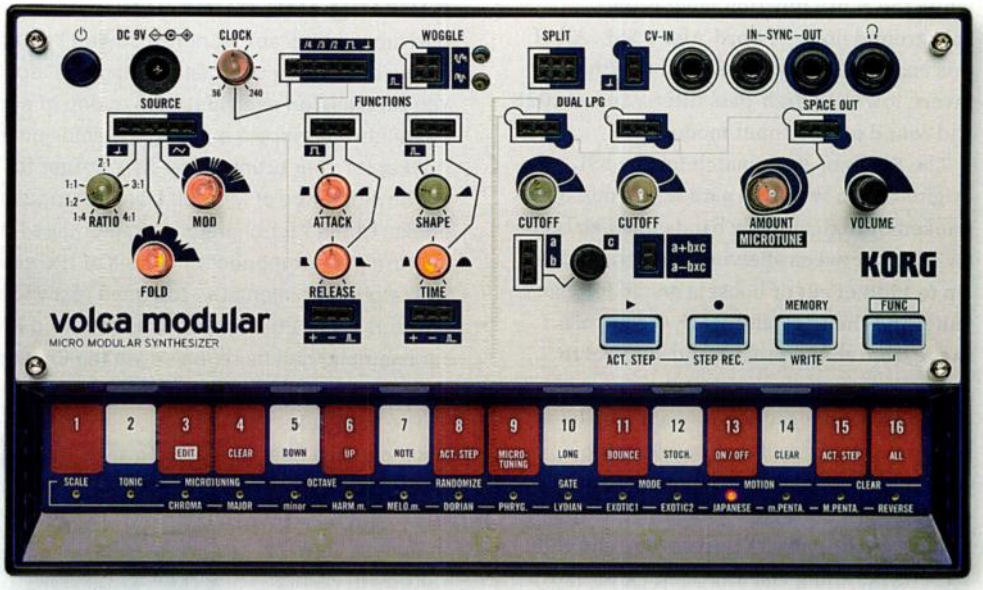
UNIVERSAL AUDIO
SSL 4000 E Collection
$299
UAD-2 owners can get their SSL fix with this awesome plugin

SOFTUBE
Console 1 MkII
$549
Softube's hardware/software hybrid presents a similar idea in a different way

Korg

Volca Modular

$199
korg.com



By Si Truss
Si Truss is the Editor of Electronic Musician and Future Music magazines

Strengths

+ West Coast synth elements
+ Fully-patchable
+ Portable

Limitations

- Can get very fiddly
- More complex and less intuitive than other Volcas

The pocket-sized synth range heads to the West Coast

Leaked images of the Volca Modular hit the internet several days before it was officially unveiled by Korg. Reaction was, perhaps understandably, sceptical; a Buchla-inspired modular packed in a Volca frame — surely that couldn't be anything more than the stuff of fanboy Photoshop fantasies?

The Volca Modular is real though, and fully lives up to that promise. It's a mostly analog, fully-patchable synthesizer built into a compact and portable frame. As with all other instruments in the range, it's powered by AA batteries (or an optional power adapter), packs a built-in speaker and can be controlled by its onboard touchstrip keyboard-come-sequencer.

Technically speaking, this is a semi-modular instrument, since there is a pre-routed signal path and it can generate sound from the keyboard without the need to patch individual modules together. The level of patchablity here is more extensive than most semi-modulars though, with multiple inputs and outputs for pretty much every element, as well as several utilities for splitting, combining and attenuating signals.

The real triumph of the Volca Modular, however, lies in Korg's decision to use a selection of synth elements influenced by 'West Coast' instrument design. West Coast synthesis is generally linked to the early synth designs of Don Buchla and Serge Tcherepnin, two California-based — hence the name — synth pioneers, who were creating modular and semi-modular instruments in the late-'60s and early-'70s. As opposed to more common 'East Coast' subtractive synths like the Moog modulars, West Coast synths tend to make use of complex oscillator modulation, as well as distinctive features such as function generators and low-pass gates. Sonically, West Coast synths tend to lean more towards experimental, metallic and atonal sounds, but can also be great for timbres that feel more natural and 'real instrument'-like than their East Coast counterparts.

In the case of the Volca Modular, the West Coast influence begins in the oscillator section, which makes use of two triangle wave VCOs in a carrier-modulator relationship to achieve FM synthesis sounds. Two rotaries control the relationship between these two VCOs: Mod adjusts the amount of modulation applied while Ratio controls the pitch of the modulator signal in relation to the carrier. By default, the carrier pitch tracks that keyboard/sequencer, although there are patch inputs for the carrier pitch, ratio and modulation amount. The oscillator section also features a wavefolder, which

> "This is a complicated instrument that specializes in experimental sounds"

applies an inversion to the shape of the oscillator's output signal, creating additional, gritty harmonics. The depth of this effect can be controlled by a front panel rotary or patch input.

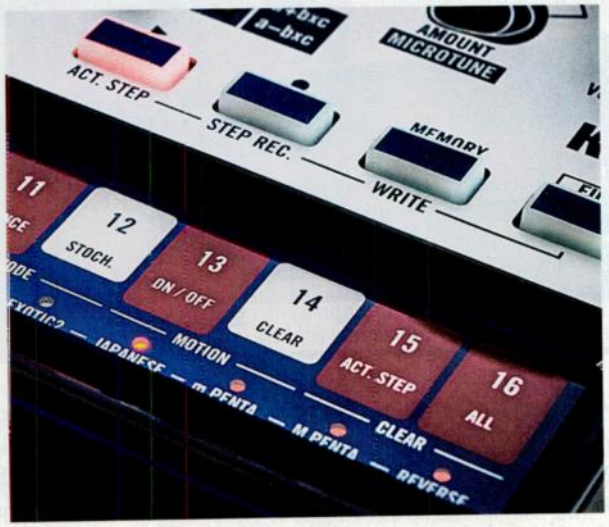
Following the pre-patched routing, the oscillator signal is then fed into one of two low-pass gates. These are another common characteristic of West Coast synth designs. They are, in effect, a combination between a non-resonant low-pass filter and a standard VCA, with a single control that affects both the volume and harmonic quality of the sound. Having two of these onboard opens up lots of possibilities, particularly since the oscillator has separate output patch points for both the carrier and modulator signals. This effectively means that the two triangle VCOs can be routed and filtered independently. It's worth noting too that, unlike some semi-modular synths, patching from an output on the Volca Modular doesn't break the internal routing, meaning that, for example, the modulator can be patched to an LPG whilst still modulating the main oscillator output, or the carrier can be routed to both LPGs simultaneously (without needing splitting first).

Modulation is provided by two Functions, which are essentially simple envelope generators. The first is an attack-hold-release envelope with controls for the Attack and Release time. The second is a two stage rise-fall generator with a Shape control that manipulates the ratio of attack to decay time, and Time control that determines the overall length. Both Functions have positive and inverted outputs, modulation inputs for their controls, as well as gate inputs and end trigger outputs. In both cases, the latter two elements can be patched into one another — end trigger out into gate in — so that the Functions can act as looping envelopes or, in the case of the rise-fall generator, a shapable LFO.

Sequencer upgraded

The Volca Modular's sequencer boasts a number of upgrades compared to those found on the rest of the range. Firstly, there's a scale mode here, with 14 selectable scales plus a micro-tuning function, which allows each note on the keyboard to be individually tuned +/- 100 cents. There are several randomization tools, which can be used to generate note data, randomize micro-tunings or randomly assign which steps are currently active. There are multiple sequencer directions too, with standard left-to-right playback accompanied by bounce and stochastic modes, the latter of which offering semi-randomized playback whereby the sequencer will jump back and forth through assigned sequencer steps. There's also a simple long/short gate selector, for adjusting the duration of all sequencer steps.

Most common Volca features make a return too. Korg's Motion Sequence automation system allows movement for all front panel rotaries to be recorded to the sequencer. While, given its hands-on analog nature, the Modular can't save patch presets, it does still allow for 16 patterns to be saved and recalled, including Motion Sequence data.

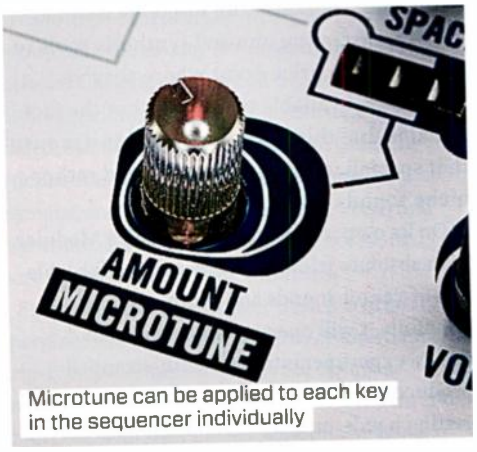


The Modular also has a sample and hold circuit, labelled Woggle. By default this uses a pink noise generator as its sample source, but it has both sample and trigger inputs, plus stepped and smoothed outputs.

At the end of the pre-routed signal path is a digital reverb effect, labelled Space Out. This is a smart edition; without it, the Volca Modular can sound dry and lifeless, but just a little of the effect can really bring the sound to life, adding depth as well as a touch of stereo width. The Space Out module has a single Amount control that adjusts the dry/wet and decay of the sound simultaneously. It's not the most natural sounding reverb but it is hugely characterful — lots of short, digital-sounding reflections pushing into short-delay/chorus-like territory at higher levels.

Separate from the pre-patched signal chain are a pair of utility modules. The first is a dual lane signal splitter, with two inputs each feeding into a pair of outputs, allowing two individual audio or control signals to be split in two. The second is a utility for combining and attenuating signals, featuring three inputs, labeled A, B and C, along with two outputs.

Oscillator modulation and wavefolding gives the Modular its West Coast character



Microtune can be applied to each key in the sequencer individually

Despite just two controls apiece, the two Functions offer plenty of flexibility

Where Pigments shines is its ability to apply four simultaneous audio-rate modulation sources to any of its wavetables, each with independent amounts that can be simultaneously modulated via the vast array of resources [Fig. 5]. Since each of these sections has a distinct flavor (and historic relevance), it's actually easiest to explore them using the Wavetable Engine's "Basic Waveforms" table, which consists of sine-triangle-saw-square. From there, you can advance to the more complex options in the factory library.

Fig. 6. The Modulator oscillator is tuned to semitones, but with a little math, you can figure out the harmonic ratios. It also includes 10 waveform and noise options

All four destinations are governed by the same audio-rate Modulator [Fig. 5] — an audio-range oscillator offering the standard waveforms (sine, triangle, etc) and five noise colors [Fig. 6]. Its coarse tuning knob can operate in relative, absolute, or non-tracking Hertz mode. For getting the hang of these techniques, it's best to select the "relative" option as it yields the most musically consistent effects.

Fig. 7. Frequency Modulation and Phase Modulation are both represented, each with subtly different tonal properties

Frequency Modulation

The first section is true Frequency Modulation, as it was originally discovered by John Chowning in 1967. While there are options for both linear and exponential FM types, linear is the more musical of the two and approximates the results of Chowning's pioneering work. Since the only other parameter in this section is FM amount, the modulator tuning and waveform play a crucial role in the timbral output.

Technique: While other waveforms will yield growly "Big Room" tonalities, begin by using a sine wave from both the Basic Waveforms table and the modulator. As for tuning, since the modulator's coarse tuning is in semitones (unlike mainstream FM instruments' use of harmonic tuning) it's useful to understand the correlation between specific notes and harmonics. That is, coarse tunings of 0, +12, +19, and +24 are the equivalent to the first four harmonics, respectively. It's also worth noting that proper FM delivers a more subdued effect than its more popular cousin, Phase Modulation, so if you're looking for vintage Yamaha timbres, the next section will be of particular interest.

Phase Modulation

Often a point of contention between synthesis historians, Phase Modulation is the actual technology implemented by the Yamaha DX and TX synths that dominated the 80s. Since the sonic output is nearly identical, this debate is largely academic. As a result, the FM description also applies to PM, with the difference being that PM is a tad brighter, overall.

Technique: Using the FM approach outlined above, assign envelope 2 or 3 to the PM amount knob as its modulator at around 50% and give the envelope an immediate attack, 0.400 second decay, zero sustain and quick release. This will create a classic 2-operator configuration.

From there, here are the coarse tuning values for some of the most famous harmonic ratios:

- 0 delivers a 1:1 ratio, good for basses and, with longer modulation attacks, some horns.
- +12 is a 2:1 ratio, which creates the famous "knocking" house bass tone.
- +19 is a 3:1 ratio, the core timbre of the DX "Jazz Guitar" preset.
- +22 is close to the 3.5:1 ratio for the iconic DX "Tubular Bells" preset. Use a longer decay and release on your modulator envelope, with a lower modulation amount for best results.
- +24 will give you a 4:1 ratio which has a similar character to the +12 tuning, but a tad brighter.

Fig. 8. Pioneered by Casio, Phase Distortion can transform a sine wave into a harmonically rich waveform. Pigments includes six waveshape targets for this process.

Phase Distortion

Originally implemented in the Casio CZ series back in the eighties, Phase Distortion was a competitor to the DX sound, favored for its more "analog" texture. While it imparts a filter-like "wow" on harmonically complicated waveforms, you can recreate the behavior of the original Casio CZ-101 by again starting with a sine wave, leaving the modulator amount at 0, then selecting from one of the six "Phase Disto" waveform targets from the PD menu.

As with the original Casio, this will smoothly morph your timbre from the pure tone of a sine to the brighter texture of the target wave, which was how the CZ achieved its pseudo-analog flavor.

Pro Tip: While the above technique may deliver "thin" results when used in the traditional manner, it sounds quite contemporary when paired with Unison settings higher than four voices.

Wavefolding

This transformation tool is having a bit of a resurgence thanks to its prevalence in West Coast design techniques. It's also found on Arturia's Minibrute series, disguised as "Metalizer", as it's most controllable when applied to a triangle or sine wave. The principle here is that it functions a bit like a distortion or saturator that specifically folds the top of the waveform back down on itself using one of three shapes: Sine, triangle, and an undocumented "squiggle", each brighter and grittier than the last.

Fig. 9. The Wavefolder section includes three shapes. Starting at 40-60% depth with a sine or triangle wave oscillator will yield results like Arturia's Minibrute Metalizer parameter

Technique: To familiarize yourself with the essentials — and understand what's happening with the Metalizer parameter in the Minibrute — start with a triangle wave in the basic wavetable, leave the modulator tuning at zero, set the fold shape to triangle, and increase its amount or modulate it with an envelope or LFO. From there, try the harmonic tunings outlined previously, in conjunction with the modulation depth knob at around 50%.

applies an inversion to the shape of the oscillator's output signal, creating additional, gritty harmonics. The depth of this effect can be controlled by a front panel rotary or patch input.

Following the pre-patched routing, the oscillator signal is then fed into one of two low-pass gates. These are another common characteristic of West Coast synth designs. They are, in effect, a combination between a non-resonant low-pass filter and a standard VCA, with a single control that affects both the volume and harmonic quality of the sound. Having two of these onboard opens up lots of possibilities, particularly since the oscillator has separate output patch points for both the carrier and modulator signals. This effectively means that the two triangle VCOs can be routed and filtered independently. It's worth noting too that, unlike some semi-modular synths, patching from an output on the Volca Modular doesn't break the internal routing, meaning that, for example, the modulator can be patched to an LPG whilst still modulating the main oscillator output, or the carrier can be routed to both LPGs simultaneously (without needing splitting first).

Modulation is provided by two Functions, which are essentially simple envelope generators. The first is an attack-hold-release envelope with controls for the Attack and Release time. The second is a two stage rise-fall generator with a Shape control that manipulates the ratio of attack to decay time, and Time control that determines the overall length. Both Functions have positive and inverted outputs, modulation inputs for their controls, as well as gate inputs and end trigger outputs. In both cases, the latter two elements can be patched into one another — end trigger out into gate in — so that the Functions can act as looping envelopes or, in the case of the rise-fall generator, a shapable LFO.

Sequencer upgraded

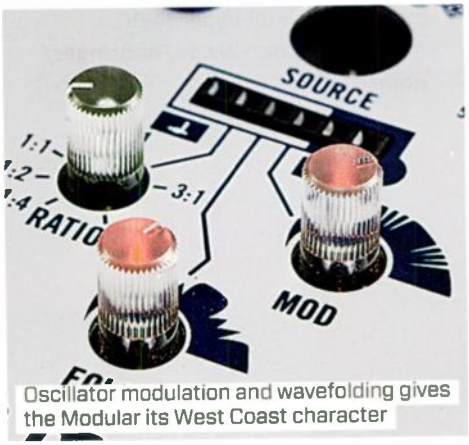
The Volca Modular's sequencer boasts a number of upgrades compared to those found on the rest of the range. Firstly, there's a scale mode here, with 14 selectable scales plus a micro-tuning function, which allows each note on the keyboard to be individually tuned +/- 100 cents. There are several randomization tools, which can be used to generate note data, randomize micro-tunings or randomly assign which steps are currently active. There are multiple sequencer directions too, with standard left-to-right playback accompanied by bounce and stochastic modes, the latter of which offering semi-randomized playback whereby the sequencer will jump back and forth through assigned sequencer steps. There's also a simple long/short gate selector, for adjusting the duration of all sequencer steps.

Most common Volca features make a return too. Korg's Motion Sequence automation system allows movement for all front panel rotaries to be recorded to the sequencer. While, given its hands-on analog nature, the Modular can't save patch presets, it does still allow for 16 patterns to be saved and recalled, including Motion Sequence data.

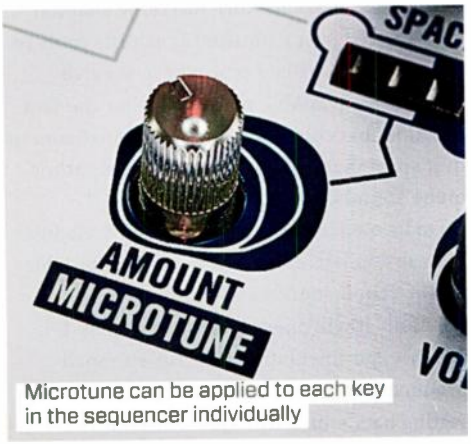
The Modular also has a sample and hold circuit, labelled Woggle. By default this uses a pink noise generator as its sample source, but it has both sample and trigger inputs, plus stepped and smoothed outputs.

At the end of the pre-routed signal path is a digital reverb effect, labelled Space Out. This is a smart edition; without it, the Volca Modular can sound dry and lifeless, but just a little of the effect can really bring the sound to life, adding depth as well as a touch of stereo width. The Space Out module has a single Amount control that adjusts the dry/wet and decay of the sound simultaneously. It's not the most natural sounding reverb but it is hugely characterful — lots of short, digital-sounding reflections pushing into short-delay/chorus-like territory at higher levels.

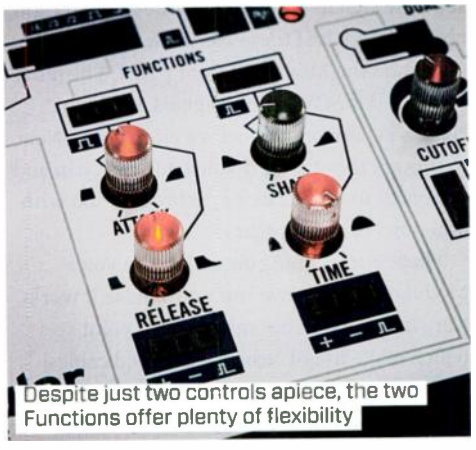
Separate from the pre-patched signal chain are a pair of utility modules. The first is a dual lane signal splitter, with two inputs each feeding into a pair of outputs, allowing two individual audio or control signals to be split in two. The second is a utility for combining and attenuating signals, featuring three inputs, labeled A, B and C, along with two outputs.



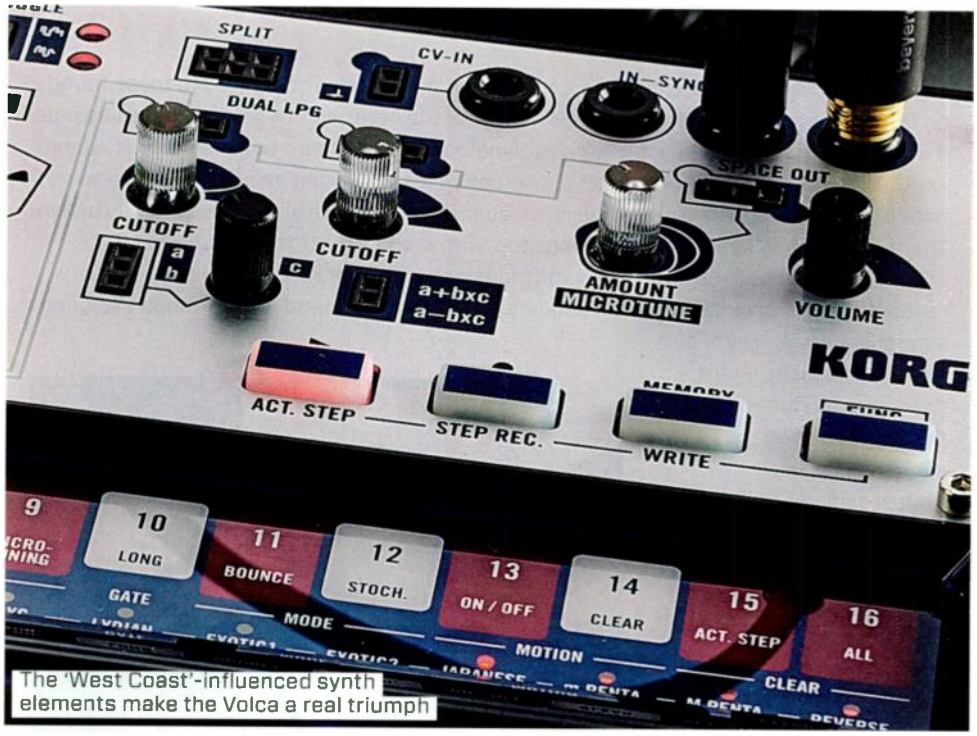
Oscillator modulation and wavefolding gives the Modular its West Coast character



Microtune can be applied to each key in the sequencer individually



Despite just two controls apiece, the two Functions offer plenty of flexibility



The 'West Coast'-influenced synth elements make the Volca a real triumph

Here, any signal fed into B can be attenuated by C (either using the patch input, or a front panel rotary). B is then combined with any signal fed into A — depending on which output is used, the attenuated signal of B is either added to or subtracted from A. When nothing is connected to B, that input defaults to a fixed 3.3V signal. Given all the options, it's a powerful little utility.

Control-wise, the Volca Modular has patchable outputs from the sequencer, as well as a separate CV input. The sequencer has a total of five outputs, for pitch, gate and three gate division outputs, which send a signal for every second, third and fourth gate respectively. These latter outputs are great for adding polyrhythmic movement.

The Modular's CV input replaces the Volca range's standard MIDI input. The input port itself is a stereo 3.5mm jack port, allowing separate signals to be sent via the left and right channels. Each of these has its own patchable output on the Modular's front panel. The left is formatted for CV control signals, while the right is set up for a 1V/octave pitch input. Next to this are the standard Volca pulse sync in and out ports, used for clock synchronization with other bits of analog gear.

There is a lot going on within the Volca Modular's diminutive frame. To make it work Korg has opted to use micro patch points, which make use of basic wire-tipped cables, rather than standard Eurorack mini-jack connections. It's an understandable design choice, but it does make the Volca Modular, at times, very fiddly to patch and manipulate. Once more than one or two wires gets involved, things become very busy. The wires themselves are quite flimsy too, and I wonder how durable they'll prove. Fortunately Korg supply a generous amount of them in the box.

It's interesting to compare the Volca Modular to the original Volca trio — the Beats, Bass and Keys — to see just how far the range has come. Those instruments were designed with immediacy and accessibility in mind and, as great as the Volca Modular is, undoubtedly some of that original ethos has been lost as a result of its complexity.

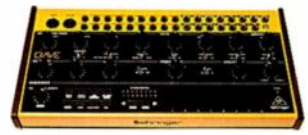
As someone with a fairly solid understanding of both the history and basic principles of synthesis, it's easy to get excited by the possibilities offered by the Modular, but I do wonder how well that will translate to the mass market. Korg has done a stellar job of transferring unusual synthesis tools to a product at a price point where they've never been available previously, but the fact remains that this is a complicated instrument that specializes in experimental and rather niche sounds.

On its own terms though, the Volca Modular is an absolute triumph. By bringing affordable left-of-center sounds and synthesis tools, hopefully it will open up a world of sound design experimentation for cash-strapped producers who previously could only dream of getting hands-on with West Coast hardware. ■

THE ALTERNATIVES

TEENAGE ENGINEERING
Pocket Operator 400
$499
TE's forthcoming DIY modular synth is more vanilla in its elements, but boasts Eurorack-compatibility via its mini-jack patch points.
teenageengineering.com

PITTSBURGH
Modular Microvolt 3900
$629
Slightly larger and considerably more expensive, Pittsburgh's desktop semi-modular has a similarly West Coast feel and a deceptively powerful sequencer.
pittsburghmodular.com



BEHRINGER
Crave
$199
We're yet to get our hands on Behringer's forthcoming semi-modular, but this East Coast-style synth offers a lot of features for under $200.
musictribe.com/brand/behringer/home

Everything you need for SampleLogic Drum Fury is on this single page of the GUI

Sample Logic

Drum Fury

$199

samplelogic.com

By Marty Cutler

Former *EM* staff editor Marty Cutler is the author of The New Electronic Guitarist, Published by Hal Leonard.

Strengths

+ An expressive selection of cinematic drums and ethnic percussion
+ Multiple articulations for each instrument
+ Energizer and Polisher provide aggregated, one-stop mastering effects
+ Simple user interface

Limitations

- Drum Mouse Trigger limited to three drums in each map, and with a fixed velocity
- MIDI control of pitch is limited to CCs. MIDI transposition of Keymaps not reflected in Kontakt keyboard

Editors for Drum Fury's effects open with a click on their gear icons. Each has a pull-down menu for presets

Sample Logic serve up a quality pack of percussion

Sample Logic is known for animated instruments capable of setting a cinematic scene with a single MIDI note, while providing the tools to customize sounds to the Nth degree. That approach yields awesome song starters, but they can often lead you down a rabbit hole. Self-contained, looping percussion beds may not fit every bill; sometimes it's just better to build the scene from a custom menu of choice, one-shot sounds. Drum Fury is the company's first venture into that territory.

Drum Fury's library (requires the full version of Kontakt) consists of roughly 11GB of cinematic drums, including toms; taiko; timpani; concert bass drums; bells; gongs and cymbals; along with a nice batch of ethnic percussion instruments, all with multiple articulations and strikes.

Facing the Fury

The user interface is simple; everything you can access is on a single page (See above); there are no mixers, step sequencers or arpeggiators. Dominating the display are three drum-head surfaces. Clicking on the head triggers a drum sound, providing an idea of the instrument you've loaded, but they trigger only three of the many articulations found in most of the Drum Fury keymaps, and only at a fixed velocity. You'll find your controller or Kontakt's virtual keyboard more useful in auditioning the drums.

Flanking the mouse-trigger drums are the Energizer and the Polisher, which group channel strip-type processors to sculpt the samples. Energizer combines compression, saturation, and distortion, with the Polisher handling imaging, EQ, transient shaping, and additional saturation. These are designed to affect the entire patch, rather than individual drum hits, and as such, you can tweak their amount, edit compression and EQ parameters or toggle them on and off (See left).

At the top right, a slider for pitch accepts Continuous Controller (CC) modulation, which is fine to raise the pitch of all drums, however, MIDI Velocity would have been a more direct and tactile approach to modulating individual drum pitches. On the opposite side of the page, a slider enables random response to Velocity, and at the bottom, sound-shaping tools include an AR envelope; delay; reverb; a low-and high-cut filter, and — if you want to create a multi (Drum Fury supplies no multis), you can transpose the MIDI-trigger notes in half-steps to make space to load additional patches and keep their trigger notes discrete from others. It would be a helpful visual aid if the transposition were reflected on the virtual keyboard's map.

Sound and Fury

Sonically, it's tough to find fault with Drums of Fury. the samples are uniformly exceptional, with a wide range of drums suitable for Cinematic production, and a very tasty, generous, and expressive batch of ethnic percussion to spice up the lot. Sample Logic has largely succeeded in its intent to provide an uncluttered user interface with a minimal compromise in sound design tools. The issues I've raised are minor, and hopefully, the company will address them in future libraries. Kudos to Sample Logic for an excellent focus on a fine set of percussion. ■

ARTURIA PIGMENTS

With so many wavetable synths flooding the market in the wake of Xfer Serum, it's easy to mistake Pigments for yet another clone of that insanely popular softsynth. However, nothing could be further from the truth. Pigments is very much its own synthesizer, cribbing elements from iconic analog synths, as well as a massive array of modulation resources that strongly evokes the capabilities of contemporary modular rigs.

Pigments' sequencing, arpeggiation, and effects all tread familiar territory, so in this month's masterclass, we'll take a deep look at the core synthesis engine, which does tricks no other softsynth in its class can. Consisting of two distinct tone generation engines, a pair of powerful filters with a few unique vintage elements, and modulation tools that include multiple random generators and a pair of elaborate function combinators, the synthesis amenities are far more than meets the eye and ear. With so many vintage-inspired elements, Pigments is also a great way to explore the history of synthesis itself, so let's get started.

Fig. 1. Based on the Minimoog architecture, the Analog engine includes three oscillators, sync, and blendable modulation from Osc 3 or Noise

Engines

Instead of traditional oscillators, Pigments offers two "engines", each of which can be either Wavetable or Analog. Since both are available at any time, your voice can be based on either two Wavetables, two Analogs, or a combination.

The Analog engine bears a strong resemblance to a Minimoog, with three oscillators and a noise generator serving as the basis for its sound. The Wavetable engine is reminiscent of Serum, but with the inclusion of four simultaneous audio-rate modulation tools, many of which are derived from the golden age of vintage digital. Both are so feature-rich that you can develop extremely detailed tones with a single engine. Combined, the results are gargantuan.

Analog Engine

Since the Minimoog oscillator architecture has been around for nearly 50 years, this section will be straightforward to all but the newest of newcomers. Even so, here's a summary of what it includes, along with some tips on Arturia-specific additions.

Fig. 2. Both Engines include a Tuning panel that offers an innovative key quantize panel and oscillator drift control for the Analog engine

As with the Mini, three oscillators with selectable waveforms (sine, triangle, saw, and square) can be tuned and mixed to create thick analog textures. The global tuning section for this engine includes a drift knob that really helps to warm things up and add a touch of vintage chaos to the pitch of all three oscillators simultaneously. Also in this section is a modern feature—tuning quantization—that's available for the global coarse tuning. With this, you can define a scale/mode by selecting notes on a one-octave icon. With quantization active, sweeping the coarse pitch with an envelope or LFO will create glissando or arpeggio effects.

Pro Tip: While setting this up for specific keys is impressive, especially when combined with delay effects, you can achieve more tasteful effects by selecting just the root and fifth for your key, creating simple arpeggio trills that won't clutter your tracks.

Like the Mini, you can toggle keyboard tracking off, here you have options for both oscillator 2 and 3, which is useful for unusual atonal effects. There's also a hard sync switch that ties osc 2 to osc 1 for harmonic sweeps.

Fig. 3. Expanding on the Minimoog, the Analog engine also includes Sync and independent keyboard tracking for Osc 2 and 3

Additionally, the width knob governs more than the duty cycle of the square wave, here it can also be applied to the triangle wave, which morphs it into a sawtooth shape at maximum values.

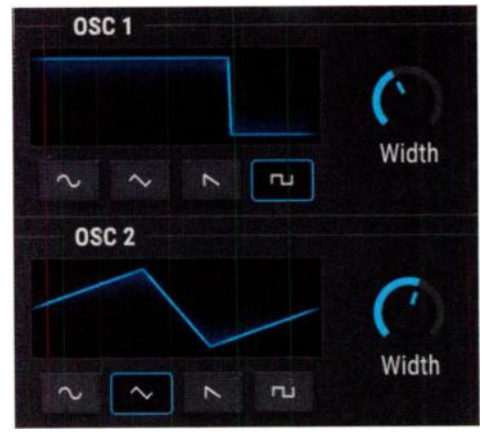


Fig. 4. The Width parameter applies to both pulse-width and triangle-saw transformations

Pro Tip: Applying envelope modulation to the Width of a triangle wave imparts a sweep that's a bit like a lowpass filter in motion as the waveform shifts from all integer harmonics (saw) to muted odd harmonics (triangle). In a single-oscillator context, this has a classic Kraftwerk feel that historians may appreciate.

While the Mini had a choice between white and pink noise sources, Pigments has a continuously variable color knob that transitions between red (diminished highs) to blue (diminished lows), operating a bit like a filter or tone control.

The last nod to the Mini here is the ability to apply frequency modulation from either oscillator 3 or the noise generator to oscillators 1 and 2 simultaneously, with an amount knob that can be modulated from any of Pigments' sources. Since this is a digital environment, the results are much more consistent than on the real thing.

Wavetable

Pigments' Wavetable engine is so reminiscent of Serum that you can actually import Serum wavetable files by pointing its browser at either a wavetable folder or set of files and instantly add them to your library. But that's missing the point, as Serum is still very much its own synth with unique strengths that keep it relevant, even now.

Fig. 5. Wavetable includes four vintage synthesis tools: FM, Phase Modulation, Phase Distortion, and Wavefolding.

Where Pigments shines is its ability to apply four simultaneous audio-rate modulation sources to any of its wavetables, each with independent amounts that can be simultaneously modulated via the vast array of resources [Fig. 5]. Since each of these sections has a distinct flavor (and historic relevance), it's actually easiest to explore them using the Wavetable Engine's "Basic Waveforms" table, which consists of sine-triangle-saw-square. From there, you can advance to the more complex options in the factory library.

Fig. 6. The Modulator oscillator is tuned to semitones, but with a little math, you can figure out the harmonic ratios. It also includes 10 waveform and noise options

All four destinations are governed by the same audio-rate Modulator [Fig. 5] — an audio-range oscillator offering the standard waveforms (sine, triangle, etc) and five noise colors [Fig. 6]. Its coarse tuning knob can operate in relative, absolute, or non-tracking Hertz mode. For getting the hang of these techniques, it's best to select the "relative" option as it yields the most musically consistent effects.

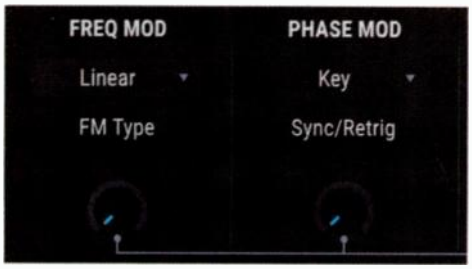


Fig. 7. Frequency Modulation and Phase Modulation are both represented, each with subtly different tonal properties

Frequency Modulation

The first section is true Frequency Modulation, as it was originally discovered by John Chowning in 1967. While there are options for both linear and exponential FM types, linear is the more musical of the two and approximates the results of Chowning's pioneering work. Since the only other parameter in this section is FM amount, the modulator tuning and waveform play a crucial role in the timbral output.

Technique: While other waveforms will yield growly "Big Room" tonalities, begin by using a sine wave from both the Basic Waveforms table and the modulator. As for tuning, since the modulator's coarse tuning is in semitones (unlike mainstream FM instruments' use of harmonic tuning) it's useful to understand the correlation between specific notes and harmonics. That is, coarse tunings of 0, +12, +19, and +24 are the equivalent to the first four harmonics, respectively. It's also worth noting that proper FM delivers a more subdued effect than its more popular cousin, Phase Modulation, so if you're looking for vintage Yamaha timbres, the next section will be of particular interest.

Phase Modulation

Often a point of contention between synthesis historians, Phase Modulation is the actual technology implemented by the Yamaha DX and TX synths that dominated the 80s. Since the sonic output is nearly identical, this debate is largely academic. As a result, the FM description also applies to PM, with the difference being that PM is a tad brighter, overall.

Technique: Using the FM approach outlined above, assign envelope 2 or 3 to the PM amount knob as its modulator at around 50% and give the envelope an immediate attack, 0.400 second decay, zero sustain and quick release. This will create a classic 2-operator configuration.

From there, here are the coarse tuning values for some of the most famous harmonic ratios:

- 0 delivers a 1:1 ratio, good for basses and, with longer modulation attacks, some horns.
- +12 is a 2:1 ratio, which creates the famous "knocking" house bass tone.
- +19 is a 3:1 ratio, the core timbre of the DX "Jazz Guitar" preset.
- +22 is close to the 3.5:1 ratio for the iconic DX "Tubular Bells" preset. Use a longer decay and release on your modulator envelope, with a lower modulation amount for best results.
- +24 will give you a 4:1 ratio which has a similar character to the +12 tuning, but a tad brighter.

Fig. 8. Pioneered by Casio, Phase Distortion can transform a sine wave into a harmonically rich waveform. Pigments includes six waveshape targets for this process.

Phase Distortion

Originally implemented in the Casio CZ series back in the eighties, Phase Distortion was a competitor to the DX sound, favored for its more "analog" texture. While it imparts a filter-like "wow" on harmonically complicated waveforms, you can recreate the behavior of the original Casio CZ-101 by again starting with a sine wave, leaving the modulator amount at 0, then selecting from one of the six "Phase Disto" waveform targets from the PD menu.

As with the original Casio, this will smoothly morph your timbre from the pure tone of a sine to the brighter texture of the target wave, which was how the CZ achieved its pseudo-analog flavor.

Pro Tip: While the above technique may deliver "thin" results when used in the traditional manner, it sounds quite contemporary when paired with Unison settings higher than four voices.

Wavefolding

This transformation tool is having a bit of a resurgence thanks to its prevalence in West Coast design techniques. It's also found on Arturia's Minibrute series, disguised as "Metalizer", as it's most controllable when applied to a triangle or sine wave. The principle here is that it functions a bit like a distortion or saturator that specifically folds the top of the waveform back down on itself using one of three shapes: Sine, triangle, and an undocumented "squiggle", each brighter and grittier than the last.

Fig. 9. The Wavefolder section includes three shapes. Starting at 40-60% depth with a sine or triangle wave oscillator will yield results like Arturia's Minibrute Metalizer parameter

Technique: To familiarize yourself with the essentials — and understand what's happening with the Metalizer parameter in the Minibrute — start with a triangle wave in the basic wavetable, leave the modulator tuning at zero, set the fold shape to triangle, and increase its amount or modulate it with an envelope or LFO. From there, try the harmonic tunings outlined previously, in conjunction with the modulation depth knob at around 50%.

Unison

While most readers will already be quite familiar with Unison effect, it's worth noting that Pigments also includes a Chord option for this feature. Unlike the Classic detuning effect, this lets you select specific chord types (major, minor, seventh, etc) which are great for vintage techno and house stabs.

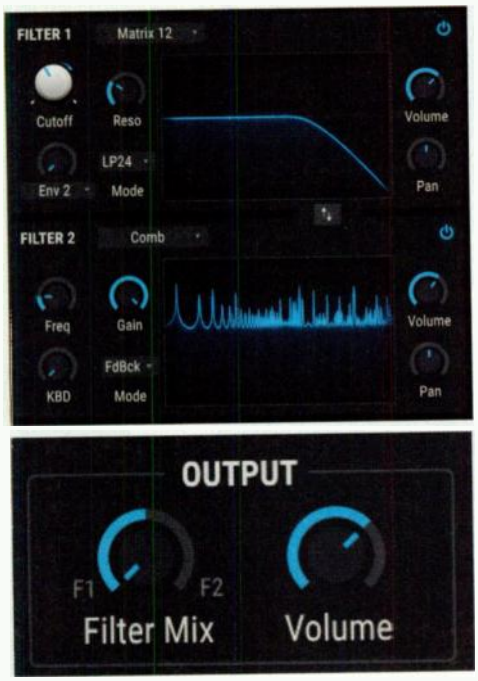


Fig. 10. Pigments' dual filters can be arranged in series, parallel, or a blend of the two routings

Filters

The dual filters can be configured in serial or parallel, with the added bonus of independent sends for each synthesis engine via the Filter Mix parameter. For example, in the serial configuration, you can route Engine 1 into Filter 1 followed by Filter 2, while simultaneously sending Engine 2 into Filter 2 alone. Parallel lets you send both engines into the filters, with independent mixes and volumes for each.

Filter types

Each of the dual filters can operate in any of the following formats, discretely: MultiMode, SEM, Mini, Matrix 12, Surgeon, Comb, Phase, Formant. Most of these types are recognizable to softsynth users, but Surgeon and Matrix 12 both deserve special attention.

Surgeon is an astonishing 64 dB/octave filter that is most dramatic in conjunction with bandpass, delivering tinny "telephone" effects when processing the Wavetable Engine or bright analog waves.

Matrix 12 offers uncanny recreations of the Oberheim Matrix 12 (and Xpander) synthesizers. While the primary options are fairly standard, the Notch+LP6 and Phase+LP6 [Fig. 11] types are remarkable, imparting a breathy "fizz" to bright waveforms—especially in tandem with an LFO.

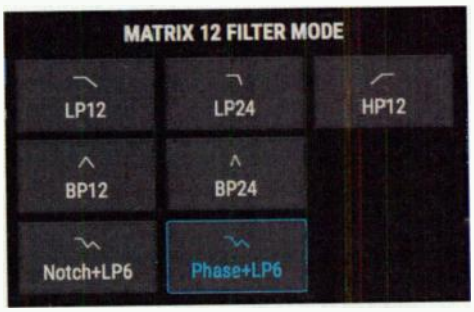


Fig. 11. The filters also include exotic models like the iconic Oberheim Matrix 12 phase and notch mode

Fig. 12. With a little pre-planning, you can route each engine to its own filter for layered effects, like those on Roland's D-50

Technique: In parallel mode, you can recreate the design approach for iconic Roland D-50 patches with a bit of advance planning. The secret here is to assign a Wavetable engine to Filter 1, with a chime/bell wavetable (Polygon Inharmonic is a great choice), giving it a percussive mallet-like envelope with very fast decay. Then turn Filter 1 off. Next, create a thick triple detuned sawtooth using the Analog engine and send it to Filter 2 in MultiMode with the LP24 slope and a low cutoff. Then give it a soft, pad envelope. At the end of the chain, apply generous amounts of chorus and reverb.

Modulation

Pigments' array of modulation tools is astonishingly comprehensive — and everything in its arsenal can be routed to nearly any parameter. In addition to essential tools like common MIDI controller assignments, three envelopes, and three LFOs, there are three Function generators with customizable shapes for step-sequencing and sidechain patterns. These can also be used as one-shot sequences for sophisticated enveloping tasks.

Fig. 13. In addition to Sample & Hold with integrated lag controls, the Random generators also include Turing and Binary options for recreating those glitchy modular textures that are all the rage these day.

There are also three Random function generators [Fig. 13], which are marvelous resources for adding organic "chaos" to textures. Here, you'll want to use very small amounts on timbral and tuning parameters. The first option is Turing, which generates random sequences that can also be locked into loops of up to 64 steps, for instant pattern creation. The Sample & Hold generator includes Rise and Fall lag generators for smoothing, which sounds fantastic if applied tastefully to Wavefolding—simulating the sound of decaying circuitry. It also lets you select the keyboard as its trigger source, like the ARP Odyssey, generating a different random value every time you hit a key. The Binary generator randomly flips between two extremes, with variable probability, which is useful for those hyper-erratic modular effects that are all the rage on Instagram.

Fig. 14. Different modulation sources can be combined, with unusual mathematical tranforms via the Combinate tools

Also noteworthy for modular fans are the dual Combinate tools, which allow you to select a source, a modulation input, and then apply one of eight mathematical transforms to the pair. These include everything from adding and subtracting, to multiply, divide, crossfade, and threshold gates. While these aren't terribly practical for practical musical applications, they're terrific for constantly evolving generative soundscapes, specifically with the Random value generators.

Conclusions

Once you get the hang of Pigments' sophisticated synthesis engine, adding effects like chorus, delay, and reverb is just the icing on an already delicious cake. The fusion of vintage and modern tools in this extraordinary synth is a one-of-a-kind design experience. ■

Pitchshifting Tools

We tweak the frequencies and explore how both large and small changes in pitch can make a big difference

Broadly, pitchshifting treatments fall into two categories within music production; the processes we undertake in a remedial way to correct errors (using plugins like Auto-Tune or Melodyne), and those we use in a creative context. As you'd expect for the more creative end of the spectrum, there's no limit to how pitchshifters could usefully make their presence felt. As we'll see from this month's walkthroughs, pitchshifting techniques can be used to turn single-line synth parts into ghostly pads. Equally, they can be used for subtle detuning which, in turn, can significantly increase the impact of a single sound as its presence is 'thickened' within a track.

Most pitchshifting plugins allow you to make both coarse and fine adjustments to pitch. Coarse changes tend to be pitch offsets in steps of semitones (so 12 steps for an octave, seven for a fifth). Alternatively, you could make a fine-tuning adjustment, measured in cents (there are 100 cents in a semitone). Detune effects are achieved within synthesizers by shifting one or more oscillators in a sound above or below the others, creating a wider pitch base; anything above or below 15 cents in either direction will produce a fairly extreme level of detune. If you're working with a pitchshifting plugin at the mix stage (rather than within a synthesizer), you'll be relying on effects to provide such processing and, as a result, the mix dial in your pitchshifter of choice will also prove to be an essential parameter — this sets the blend between the original pitch and the processed pitch. If a part is consistently half a semitone sharp, a -50 cent correction set to 100% wet will shift the entire part down into tune. Alternatively, let's suppose you wanted to create a ghost harmony which is a fourth below the pitch of the original part; a -5 Coarse offset with a '50% wet' balance will provide an even volume blend between the parts, as it would equally blend the original part and the pitchshifted harmony.

As with so many processes, great results can be made by stacking pitchshifters. Imagine a part with a consistent root note to which you want to overlay harmonies. You could set up a first auxiliary track, route your source sound to it and set up a pitchshifter to create the pitch offset you want. Then, you could set up a second auxiliary track and create a second harmony using a different pitch offset, and continue doing so, progressively adding new harmonies on as many auxiliaries as you like. If you suddenly hear that one part clashes with the harmony, you could use automation to shift the pitch offset to a more appropriate note at that moment.

QUICK TIPS

1 Use Pitchshifting tools to teach yourself more about harmony. By understanding which notes work alongside others, you can create pitch treatments which blend musically with your tracks.

2 Don't forget that your DAW is likely to contain offline processors for shifting pitch as well as plugins. Creating a 'permanently' pitch-adjusted audio file is a good idea if the original is consistently out of tune, getting it fixed for good.

3 Another creative way to work with shifting pitch is to load an audio file into a sampler and use pitchbend to warp and blend pitch in unusual and imaginative ways.

4 Pitchshifters also form the backbone of plenty of voice effects in film and TV. Cartman from South Park and Alvin (of Chipmunk fame) are but two characters whose voices would sound significantly different without pitchshifting.

Six top pitchshifters

Eventide H3000 Factory **$349**
Famous for their range of hardware Harmonizers, Eventide have designed this plugin to let you do wild and crazy things with pitchshifting, with modulation options aplenty.

Antares Auto-Tune **$299**
It's most famous these days for its T-Pain-alike hyper-tuned vocal effects, but Auto-Tune is highly capable of more 'natural' and creative pitchshifting capabilities too.

GRM Shift **$200**
With a pitchshifting algorithm which should make you sit up and take note, GRM Shift is capable of everything from subtle detune to more radical pitch offsets.

Celemony Melodyne **from $99**
Melodyne does polyphonic tuning, pitch correction, formant shifting and more. Melodyne's feature set makes it a go-to choice for creative professionals.

Universal Audio
Eventide H910 Harmonizer **$249**
This lets you warp pitch in a number of ways, pairing the sublime and the ridiculous in a delicious combination. It requires UAD hardware to run.

Your DAW's Pitchshifter
Almost every DAW comes with a pitchshifter. A few years ago, 'native' pitchshifting algorithms weren't up to much, but fortunately, they've come on leaps and bounds since then.

Adding pitch width to headline synth parts

If you haven't programed detune or pitch-widening techniques into your synth parts — you can do this at the processing stage instead

Pitch has a powerful impact on our relationship with music and the individual instruments which make up the tracks we listen to. We can tell, immediately, if a singer is out of tune, and yet when we run recorded performances through pitch-correcting software, it's often the case that the 'out of tune-ness' equates to only a few cents either sharp or flat. The reason notes sound out of tune is because they're notably (if only a little) above or below the other pitches within our tracks, providing an unwelcome clash. However, if sounds are stretched both sharp and flat at the same time (using detune to equally widen a pitch base in both directions), they can occupy a more dominant role in a mix, without sounding out-of-tune.

Our backing track contains plucks, kick and bass. The plucks are fine but they're a little ordinary and would benefit from a richer sonic presence. We can achieve this by reaching for pitch-spreading detune tools. Here's the track at its starting point.

The UAD Studio D Chorus plugin adds an Ensemble-style detune to spread the pitch base wider, and Eventide's H910 Harmonizer uses an Envelope to spread pitch wider as each note is detected, coloring the Attack of each note.

We also add a specialist reverb treatment to widen pitch spatially. This comes from Valhalla's Shimmer, to which we add a +12 semitone pitch offset. This creates a high, celestial cloud of reverb to add extra presence to the plucks.

Creating harmonies with pitchshifting

We explore a way to create a polyphonic, ethereal pad sound from a monophonic synth part

01 We start with a basic plucked sound from NI's Kontour, playing simple arpeggiated steps. We send this to a long reverb from UVI's Sparkverb which adds a warm bed behind the lead sound.

02 We set up a second auxiliary, and onto this we insert Soundtoys' Crystallizer. We set the mix to 100% Wet and the Pitch to 1200, to produce an octave-up pitchshift. We set both Splice and Delay to eighth-notes. We send this Auxiliary to the reverb Aux too.

03 This works nicely but it dominates too much. So we pan the sound -40 (to the left) and drop the Auxiliary return fader to -11dB to blend the octave Crystallizer effect more subtly under the original part.

04 We create a similar effect via a second Auxiliary. This time, we offset pitch +700 cents in Crystallizer, which equates to a fifth above the original pitch. This creates a richer cloud of sound, especially as there are now two pitchshifted treatments playing together.

05 We create a third treatment, with an offset of +300 cents, to produce a minor third. Even with the levels of the second and third pitch offsets blended, these don't sound quite right. Something is clashing in the pitchshifted treatments.

06 This is due to the Recycle amount in Crystallizer. With Recycle levels higher, more pitches are added. If the trigger note is a C, the first pitch-offset will be a G. With Recycle boosted, we hear D, A, E and B (the subsequent 5ths) too. These clash, so things sound better when Recycle levels are dropped.

Super-compressors

For when just attack, release, threshold and ratio won't do, these are the most compress-hensive dynamics devices out there right now

u-he Presswerk $129

u-he's compression brainchild can be taken many ways. To start, it's simple: Threshold, Ratio, and Attack that goes down to 0.1ms, Release and Adaptive Release, Dry/Wet, Auto Makeup, and a lovely saturation section to boot.

But let's head back into esoterica again. You can select between Feedback and Feedforward circuit models or an 'Interactive' mode that combines the best of both worlds, the detector's RMS Window can be tweaked, and the knee response changed using the Soft Knee and Non-linearity controls. The sidechain can be high- and low-pass filtered, and both internal and external sidechains can be blended between. There's a 'dual phase rotator' to make Presswerk behave more like an analog compressor. Very nice touches.

Bearing this complexity in mind, perhaps the best thing about Presswerk is the ability to load it in different views with more bitesize controls. These focused interfaces — Easy Compressor, Drum Compressor, Limiter, etc — let you concentrate on just what you need to carry out specific tasks. It's the perfect balance of simple tasks and advanced treatments within a single plugin.

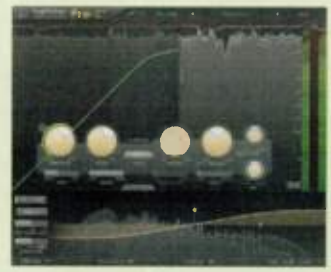
u-he.com

Blue Cat's Dynamics $149

A no-nonsense device that can handle both gating and compression in the same transfer curve, BC's Dynamics provides two circuits for these different duties. Both the 'Up' and 'Down' circuits contain Threshold and Ratio controls, a Knee, and Depth (range, by any other name). These two stages can be pushed into positive or negative ratios, with the two stages plotted on the same graph, and the attack, release and hold of both determined together in a separate pane. Attack and release 'Shape' parameters are a nice touch, although not obviously descriptive of what those shapes are. Elsewhere, there are some niceties like VCA/Opto processing style blending and Peak/RMS detection blending, Lookahead, and L/R M/S stereo linkage.

The Ashley Madison of compressors, BC Dynamics has just the right amount of new dynamics experiences to pique the interest of someone who's bored of the same four controls and wants to go further down the rabbit hole.

bluecataudio.com



FabFilter Pro-C 2 $179

A compressor souped-up with that legendary FabFilter sauce, Pro-C 2 gets you compression with eight flavors, including Vocal, Opto, Punch and Mastering, and Auto makeup Gain on output if desired. There's Lookahead, Hold and fairly comprehensive sidechain editing, with Pro-Q-style EQ. The Stereo Linking slider spans Unlinked (0%), Linked (100%), and pushing it past 100 lets you determine linking in mid/side combination.

Pro-C 2 doesn't quite take it as overboard as the other compressors in this round-up, remaining focused and restrained in its implementation of compression. The options it does give you, though, are all stunningly easy to use, immediately available and intuitive. Other than that, the best thing about this one is probably its waveform visualizer, running by alongside the knee so you can immediately see how your changes are affecting the compression.

fabfilter.com

DMGAudio
Compassion $149

Starting off in a similar way to your average compressor, Compassion offers Threshold, Ratio, Attack and Release dials. The first sign that the plugin goes above and beyond is the negative ratio settings, which bring the signal down further than the set threshold. Expand the interface, though, and you'll find the advanced controls in a fold-out panel. Features like auto release behavior, hysteresis, ceiling and expansion controls are here, giving access to a world of microtweaks to get your compression in any shape you can conceive. The result is an eminently customizable plugin, although the interface isn't as immediate as it could be, resorting to menus and separate views too often.

Compassion's Mods button is an interesting touch. Mods save and load similarly to presets, but they only change certain parameters within the plugin, letting you try out, say, an Opto or FET compressor type, settings prepared specifically for vocals or bass, or different noise floors, mid/side configurations and expansion types. With your other settings retained, loading a mod instead of a preset is a good way to manage the web of settings in the plugin as a whole.

dmgaudio.com

MeldaProduction
MTurboComp $199

On the surface, you get the usual Threshold, Ratio and detector EQ controls. Measuring their values as percentages is a strange choice, but this is easy to overlook once you see what's lurking beneath 'easy mode': a panoply of controls for what is probably the ultimate in compression customization in this round-up. With an astounding four envelope followers and another four processors to match, plus post-compression EQ and Saturation, this is a dynamics processor that hears the phrase 'less is more' and punches it into the ionosphere.

For example, not content with simply offering Attack, Release Minimum and Maximum, Peak Hold, RMS Length, Delay and assorted other controls for each of those four detectors, each also offers a further 'Settings' window, featuring custom-drawn Attack and Release Shapes (editors open in yet another window), Psychoacoustic Prefiltering, Spectral Smoothing... the list goes on! If you're a super-dooper compression nerd par excellence, this could be enough for you. And if it's all a bit too much, you've still got the Easy Mode to fall back on.

meldaproduction.com

GuitarPlayer

ON SALE NOW

Creating guitarists for more than 50 years

Available to order online at www.myfavoritemagazines.com

First Things First?

Everything You Know about Pedal Order is Wrong

By Michael Ross

When it comes to putting the pedal to the metal, why not put the metal to the pedal?

Clickbait headline aside, there have been certain "rules" about the order in which effect pedals are chained. Still, as we shall see, they beg to be broken. Often, the first question newbies to the world of stompboxes ask is, "What is the best order for a pedal signal chain?" — or, words to that, um, effect. There is a good place to start: Germanium fuzz first, as it needs to see an unfiltered guitar signal in order to respond properly to changes in the guitar volume knob. Next, any evelope filters, as they too respond best to the dynamics of a pure guitar signal (The fuzz would still come first but you probably wouldn't use them together). A wah pedal would follow, as some germainum effects oscillate when a wah is sent into the input. From there, the typical order would be compressor (so as not to add noise after overdrives), boost, overdrive and/or distortion, modulation (chorus, flange, tremolo), volume pedal, delay, and finally reverb.

However, anyone truly familiar with pedal use will then add the caveat that there are really no rules when it comes to pedal order. It comes down to what sounds good to you. Consider, for example, placing delay and reverb at the end of the chain in that order. I followed this "rule" for years and it has its charms. Placed there, the amount of perceived ambience remains constant, whether I am sending a clean or pedal-distorted signal through them. It is only recently that I realized that one key to Daniel Lanois' evocative ambience is that he sends his Korg digital delay into an overdriven amp. Moreover, he uses the preamp of the delay to help drive said amp. Steven Wilson (Porcupine Tree) has also gone on record as prefering his delays before the drive of his amplifier. As I like the sound of both signal chain orders, I set up my pedalboard with the choice of a slap or long delay going from my Eventide H9 into my Jetter Jetdrive, and the same two delay options after the dirt, courtesy of my Source Audio Nemesis Delay. For even further My Bloody Valentine-style murk, try putting the reverb before the distortion. Just be aware that when you back off your guitar volume to clean up the sound, the amount of delay or reverb you hear will decrease.

Wah pedals will create two different effects when placed before or after dirt pedals. Before, the changing frequencies will drive the overdrive or distortion differently as you rock the pedal, whereas post dirt, the wah acts more like a synth filter. More adventurous guitarists might even find that they like the oscillation created when running the wah into a germanium fuzz.

I ran my tremolo pedal into my delays and reverbs, until I started playing through two amps with great sounding tremolo built in: a Supro Comet and a 1966 Fender Bandmaster. Hearing the tremolo post ambience was a revelation. While the wavering amplitude of the pedal might get lost in a long delay or large hall setting on the reverb, having it post modified the entire signal path, ambience included, creating a much more prominent tremolo effect. By keeping my tremolo pedal on my board before the delay and reverb and also having the on/off switch for the amp tremolo at my feet, I retain the option of two different modulation effects available.

You should be getting the picture by now; there is no "best" pedal order. Placing a clean boost before an overdrive pedal will increase the drive without raising the volume much, whereas placing it after the drive will raise the volume without increasing the distortion. Feel free to start with the "standard" order, but don't be afraid to ignore me and run that fuzz into an envelope filter and see what happens. ■

Matmos

Drew Daniel explains their process and how they created the new album, *Plastic Anniversary*

Formed in San Francisco and now based in Baltimore, electronic duo Matmos — aka Drew Daniel and M.C. Schmidt — have become known for their high-concept albums that see them building sonic experiments around a central theme. Their latest, *Plastic Anniversary*, sees the pair creating rich soundscapes purely from recordings of plastic material.

When and how did you start making music?
I didn't come at this through training in a proper musical instrument in any way. When I was 16 years old I was heavily into William S. Burroughs, and specifically the "cut up" novels like "The Soft Machine" and "The Ticket That Exploded." Following his instructions about making field recordings and cutting them up, I started to collect cheap tape recorders and to do primitive experiments in tape collage using the pause button to create repetitions of sounds: mostly field recordings and old vinyl records, but also dialogue from films and my own voice. I would put cut-ups that I had made onto multiple tape decks and play them back simultaneously into another tape deck with a mic which would record those multiple sound sources and mix them together, so I was multi-tracking before I had a four-track. When I was younger I was really into hip hop and breakdancing and I think hearing tracks by the Art of Noise that had collage elements (like "Close To The Edit") was a kind of blueprint for me. So it was hip hop-meets-Burroughs? The results sounded very crude but it was so fun to do.

Tell us about your studio/set-up
We have always made our records at home, working in studios in whatever apartment we were living in in San Francisco in the early-'90s and now in the basement of our house in Baltimore. It's a large basement. Basements tend to flood in Baltimore so we wired all the electricity to the ceiling and there are no cables on the floor. After 26 years of making music we have a huge hoard of instruments, both acoustic (19th century psalteries, a hurdy gurdy, lots of horns and percussion and bells and odd objects) and electronic (our friend Paul Brown keeps his enormous modular synth in our studio which is full of cool modules, we have an ARP 2600, multiple SH-101s, some cool boutique synth instruments like the Double Knot and the KNAS Moisturizer). It all goes through a 32 channel Midas Venice F32 mixing desk which is connected to an Apple tower running Digital Performer, which is what we use to multi-track.

What's the latest addition to your studio?
We make our music by sampling objects: cacti, oatmeal, latex, chocolate pudding, you name it. So for us, the point is not to acquire new gear but to acquire new instruments for each album, essentially starting all over with brand new instruments every time we make an album. The new album's instruments include a police riot shield, a breast implant, a salad bowl, an ATM card, Styrofoam, Bakelite dominos, plastic billiard balls, and artificial human flesh generated by the SynDaver corporation. So I don't know if we can call that "music making gear" or not. The things we make music out of aren't musical instruments until we kidnap them and force them to serve musical ends. And the portable lesson there is that anything can be a musical instrument if you're willing to press the issue. My partner M.C. Schmidt has been using the Samplr app on his iPad a lot and it features heavily on the new Matmos album. It's simple, easy to use software but very flexible and powerful in terms of what you can do with it.

Where do you start with a new project?
I start with a concept that provides a restriction on how I might work. This gives an album project a strong and clear focus. The album we have just released, *Plastic Anniversary* is made entirely out of plastic as a sound source. The last album was made entirely out of the washing machine in our basement. We don't like that "blank page" feeling of freedom: it's too easy for it to sanction a kind of vertigo or freefall into self-indulgence. It's easier to focus when you make a strong commitment to one concept or one sound source or one idea and then follow through in a really fanatical way. But you have to be careful what you commit to: once you're inside the framework, it's like a dare that you've made and honor demands that you must carry it out. This led to us flying our washing machine around the world and running it onstage with a tub full of thirty gallons of water at each show. Live by the sword, die by the sword. ■

Drew Daniel's essential tips...

Work with what you've got.
Don't put off an idea because you don't have the latest and greatest gear: work with whatever means are ready to hand. There are great punk rock records which were recorded with a single mic pointed at a band. Some classics of industrial music were recorded onto a stereo cassette tape at home. Keep the focus on the expressive flow of the music itself. It may be very unpopular to say this, but "what" matters more than "how."

Save everything because you never know.
You may make something and think it's garbage, a doodle, pointless. But don't throw it away! A context may come, later on, in the middle of an entirely different song at a different tempo in a different key where you hit a wall and need to shift the mood, and in those situations, some components of older tracks and sketches and experiments can actually be harvested and integrated into the song you're working on at present, and the effect of bringing that other material into a song can be like opening a window and letting the form breathe. Songwriters do this with notebooks in which they jot phrases that might someday become a lyric, or part of a lyric; but electronic composers can do this too by holding onto all of their files and patterns and ideas, and giving them a critical listen every so often. This can make you less hard on yourself about your process. Today's trash could be tomorrow's treasure.

UNIVERSAL AUDIO
APOLLO X8P THUNDERBOLT

PRESONUS
ATOM

MOTU
8PRE-ES

KORG VOLCA MODULAR

SamAsh®

WE ARE TECHNOLOGY

ARTURIA
MICROFREAK

KORG MINILOGUE XD

SLATE
RAVEN
MTI
CORE
STATION

NORD STAGE 3

YAMAHA
MODX8

FOCAL
ALPHA 80

BEST SELECTION ONLINE

LOWEST PRICE GUARANTEE

SPECIAL FINANCING AVAILABLE

VISIT SAMASH.COM
OR CALL 1-800-472-6274
SPEAK TO AN EXPERT TODAY

It *is* ROKIT Science.

All models feature
LCD Room Correction Assistance
on back panel

ROKIT G4

Completely reimagined, designed and engineered, ROKIT Generation 4 will bring even more precision to your mixes, no matter the style of music. ROKIT G4 brings a new level of professional monitoring to everyone.

Learn more at **itisrokitscience.com**.

- Matching Kevlar® drivers ensure sonic integrity on every frequency
- DSP-driven LCD Room Tuning with 25 Graphic EQ settings for improved acoustics
- KRK App: Acoustic Real-time Room Analyzer helps guide your monitors inside your room
- Efficient Class D power amplifier for better audio integrity at reduced operating temperatures
- Scientifically-tuned front firing port offers exceptional low-end clarity, punch and extension

krksys.com KRK SYSTEMS