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GENRE DEFINING

Not all that long ago it seemed that there was a place in the mainstream for instrumental music — the Shadows, Mike Oldfield and Tangerine Dream spring to mind — but now it seems as though it’s overlooked, more often than not going undiscovered by the majority of potential listeners. It is certainly under-represented on the radio and difficult to stumble across online unless you already know what you are looking for. This concerns me on a practical level, as my own music is mainly instrumental — which brings up the thorny question of genre.

Most new, non-media-related instrumental music seems to get pigeon-holed into World, New Age, Ambient or Chillout slots, yet what do you do if you don’t feel your music sits comfortably in any one of these categories? Each of the above has its own stereotypes — nose flutes of the Andes, harmless but often uninspiring ‘crystal and lentil’ shop New Age, abstract ambient textures with little or nothing in the way of melody and Chillout with its use of traditional synths and contemporary drum loops. You get the odd name like William Orbit or Brian Eno who manage to poke their head above the parapet, but the majority remain invisible. I often joke that my compositions are ‘music to soak lentils by’, but I try to make them more than that while still retaining a sense of mystery combined with a relaxed mood. I’ll include elements of acoustic instruments, electric guitar and synthetic sound in a way that works for me without sparing a thought for genre — but I really have no idea what to call the end result other than ‘instrumental’.

Mainstream pop music is pretty much all about vocals, but look back at the history of music and you might see vocal-driven folk music on one hand and instrumental classical music on the other. Back in the pre-electricity days — which weren’t as far back as many of you might imagine — most ‘serious’ music was instrumental, yet nobody tried to force the pieces into an unsuitable category. It was just music and people either liked it or they didn’t. If it was big band music or a brass band, then the designation was exactly what it said on the tin, which is fair enough, but now there are so many possible styles using so many different combinations of instruments, that finding something to call it is a real problem. Maybe that’s why many of today’s successful composers of instrumental music write TV music or film scores rather than make records.

I don’t pretend to have a solution and maybe today’s record buyers have been so brainwashed by click track perfection and ‘sound like anybody but look pretty’ stage school vocals that they don’t want to explore anything else. However, I do feel that for those who have a genuine interest in exploring music rather than just wanting some inoffensive sonic wallpaper for their lives, good instrumental music has as much to offer as its vocal-driven counterparts.
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Ableton reveal Live 10

Ableton Live has long been a very different beast to other DAWs, not least because of its now much-imitated clip launcher (or Session) view that lets you remix your own tracks on the fly. With the release of Live 10, rather than trying to make Live more like the other DAWs, they have worked with their existing user community to refine and streamline it, and make it better at what it does best.

Simultaneously releasing major updates to their Ozone mastering suite and Neutron multiband dynamics and EQ plug-ins, iZotope have also introduced an innovative new way for these plug-ins to communicate with each other across an entire mix. Using the Tonal Balance Control plug-in, now part of both software packages, users can control all instances of Ozone and Neutron from one place, with the goal of making it easier to judge critical decisions and achieve a well-balanced mix.

Taken on their own, both Ozone 8 and Neutron 2 introduce a range of interesting new features. Among other things, Ozone gets a new Spectral Shaper module, designed to tame problematic frequencies, and the ability to A/B up to 10 reference tracks directly within the plug-in. Also new, the Master Assistant is Ozone 8’s answer to the Track Assistant feature introduced in Neutron v1. Using the Master Assistant, users can tell Ozone what they’re mastering the track for (streaming or CD) and how aggressively they want to process it (low, medium or high intensity), and the software will automatically dial in what it considers to be the optimal starting point for further tweaking. Master Assistant can also analyse a reference track and configure Ozone’s EQ and dynamics processors accordingly. Neutron 2, meanwhile, has an updated Track Assistant, a Masking Meter designed to identify frequency overlap between different channels, and the new Visual Mixer, which lets you view and adjust the relative levels and pan position of every element of the mix on an intuitive X/Y grid. To allow Neutron 2 to apply these features mix-wide, the Mix Tap plug-in lets you feed all tracks through Neutron without having to place a full instance on every channel.

Even without it, these new features would be big news, but the Tonal Balance Control plug-in looks to take things to the next level. Providing convenient access to all instances of Ozone and Neutron across your mix, it lets the user view and adjust the EQ settings for each track. The Target feature lets you call up the desired sonic profile, either from the plug-in’s own presets or by analysing a specific reference track, for comparison with the audio you’re working on. While the Tonal Balance Control is designed to work with multiple instances of either Ozone or Neutron, it can

Mixed signals

iZotope add inter-plug-in communication to Ozone 8 and Neutron 2
you would any other regular device. These look more ‘Live-like’ and will boast quicker load times and lower CPU usage. In addition, Max For Live can now access hardware MIDI ports, something that was not possible before.

Ableton are promoting a special offer in the run-up to the release of Live 10 in early 2018. Users can save 20 percent on Live 9 and get a free upgrade to 10 when it’s released. Once released, Live 10 downloads will cost £69 for Intro, £319 for Standard and £579 for Suite. Boxed versions cost between £20 and £40 more. Upgrade pricing will also be available for existing users. Keep an eye out for our full review coming soon!

www.ableton.com

also be used to control both at once, allowing you to quickly go back and tweak the mix while mastering, or look ahead to the mastering stage while mixing to preempt potential problems. Ozone 8 and Neutron 2 (£389 each for the Advanced version) are available separately, together as the O8N2 bundle (£549) and as part of iZotope’s updated Music Production Suite (£785), which also includes the RX6 Standard audio repair tool, the VocalSynth and Nectar 2 vocal processors, and the Trash 2 Expanded distortion plug-in. Ozone 8 and Neutron 2 are also available via the Splice ‘rent-to-own’ scheme, which gives users instant access to the full plug-in while letting you pay for it in small monthly instalments with no additional interest or fees.

Time+Space +44 (0)1837 55200 www.timespace.com www.izotope.com
Chandler Limited launch pair of new EMI/Abbey Road 500-series modules

The fruitful ongoing partnership between Chandler Limited and Abbey Road Studios has yielded a string of highly desirable outboard processors, making the classic sound of the EMI designs used to record the Beatles and Pink Floyd available once more — to those who can afford it, at least. Now Chandler and Abbey Road have released two new 500-series processors that make these sought-after sounds a little more attainable.

Joining the existing TG2-500 preamp in the EMI/Abbey Road Lunchbox range, the new modules are the TG Opto Compressor and the TG12345 MkIV EQ. Put all three together, and you effectively have a complete EMI TG12345 console channel in a 500-series rack. The dual-width TG Opto Compressor borrows from the full-size TG1 Limiter and Zener Limiter units, providing a single channel of versatile compression with continuous input level, output level/make-up gain, attack and release controls, a sharp/rounded knee switch and a true-bypass in/out switch. Meanwhile, the TG12345 MkIV EQ is a single-slot, mono equaliser that, true to the original console, provides just two bands. The presence and bass bands both feature continuous cut/boost controls, but while the bass band is a shelving filter switchable to either 90 or 150 Hz, the presence band offers the choice of seven bell-curve filters ranging from 500Hz to 6.5kHz, a high shelf at 10kHz, or it can be switched off entirely. The two new modules are set for release this winter, priced at £1199 for the TG Opto Compressor and £1099 for the TG12345 MkIV EQ.

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www.nova-distribution.co.uk
www.chandlerlimited.com

Sound of silence
Lewitt’s Subzero condenser mic promises incredibly low self-noise

Recent years have seen Austrian mic innovators Lewitt take their signature clean, crisp and modern sound in a number of different directions, from the affordable LCT 440 Pure to the LCT 640 TS, a sophisticated dual-output mic that lets you control the polar pattern from a plug-in even after recording. Their latest offering — the LCT 540 Subzero — simultaneously goes back to basics and takes things to extremes.

A large-diaphragm condenser with a fixed cardioid pickup pattern, a switchable pad (0, -6 or -12 dB) and low-cut filter (flat, 80Hz or 160Hz), its unassuming exterior conceals the fact that Lewitt’s engineers have gone all-out to build the quietest mic they can. The company quote the electrical self-noise as -1dB, while the acoustic self-noise is given as 4dB, which is more or less what’s generated by the random movement of air molecules alone. These are A-weighted figures, however, and Lewitt say that at 2kHz, the mic’s EIN is actually below -7dB, leading them to describe the mic’s performance as “better than your ears”! All this contributes to a dynamic range of 132dB(A), suggesting that this mic should excel where detail and sensitivity are required. The LCT 540 Subzero comes complete with a shockmount, a magnetic clip-on pop filter and a ‘military-grade’ carry case. The first batch of mics should be available soon, with the full launch planned for January 2018. Pricing had yet to be confirmed when we went to press, but Lewitt hinted that the mic should hit the street at “well below $1000”.

www.lewitt-audio.com

Fab two

Chandler Limited launch pair of new EMI/Abbey Road 500-series modules

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Vox fresh
EastWest release Hollywood Choirs virtual instrument

The latest addition to EastWest’s Hollywood range of virtual instruments geared towards cinematic scoring is Hollywood Choirs. Produced by Doug Rogers and Nick Phoenix, the team behind 2005’s Symphonic Choirs instrument, Hollywood Choirs promises epic choral arrangements from full choir and individual male and female sections, as well as new levels of flexibility and realism via the improved WordBuilder plug-in. Of all the instruments available in sampled form, the human voice is perhaps the hardest to get right. Capturing the sheer range of articulations, vowel and consonant sounds is task enough in itself, but reassembling them into convincing phrases remains the ultimate challenge. EastWest believe they have made a big step forward with this new instrument, however, promising unprecedented dynamics and realism, with the ability to type any word into the WordBuilder plug-in and have it sung by the choir. The instrument is available in two different editions — the Gold version ($499) and the Diamond version ($599), which features a user-controllable 13-mic array (including a Neumann binaural head) designed for creating surround-sound and 3D virtual reality mixes. Hollywood Choirs is also available as part of the subscription-based Composer Cloud service, which provides access to all of EastWest’s virtual instruments for a regular fee starting at $24.99 per month.

Surface detail
PreSonus launch biggest FaderPort control surface yet

The PreSonus FaderPort range, which began with the diminutive, single-fader FaderPort and grew with the fully featured FaderPort 8, has a new top dog. The FaderPort 16 Mix Production Controller is a USB 2 control surface boasting 16 touch-sensitive motorised faders, dedicated transport controls and a host of useful function buttons for controlling any DAW that supports the Mackie Control and HUI protocols.

The sleek control surface’s 100mm faders are arranged into 16 channel strips, each with select, mute and solo buttons and a small ‘scribble strip’ digital display. Alongside the array of buttons giving instant access to a range of frequently used functions, there are four user-programmable buttons as well as a scroll wheel and cursor keys. While the FaderPort 16 will work happily with other DAWs, integration with PreSonus’ own Studio One is particularly tight, as you would expect. Studio One users will be able to do things like bypass all plug-ins on a track with a single button press, navigate around the session from the control surface, and make use of the Control Link feature. When activated, this means that you can hover the mouse pointer over any parameter on screen and adjust it using the FaderPort’s pan/parameter rotary encoder. The FaderPort 16 will be available from the end of November, priced £959.

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Imitation game
Antelope launch Discrete audio interfaces and modelling mics

Having originally established their reputation with ultra high-quality digital converters and clocking hardware, Antelope moved into audio interfaces with the Orion and Goliath ranges, portable recording with the Zen Studio and Zen Tour, and now seem intent on giving the UAD platform a run for its money with their FPGA FX — a growing library of real-time, hardware-modelling effects running on dedicated DSP inside their interfaces. Now Antelope are about to make another leap forward with what they’re calling the Discrete Microphone System.

The range consists of two audio interfaces — the Discrete 8 and the Discrete 4 — and two condenser microphones — the large-diaphragm, multi-pattern Edge and the small-diaphragm Verge. The interfaces boast high-quality mic preamps, Thunderbolt and USB connectivity and Antelope’s desirable A-D/D-A conversion and clocking technology. The full range of FPGA FX is also present, covering compression, EQ, guitar-amp modelling and reverb, but the real innovation is Antelope’s new Accusonic microphone and preamp modelling, which promises to recreate the sound of a range of classic mics and mic preps. Thanks to the system’s very low latency, performers can track with modelling in place, while the engineer can later change the model or settings during mixing. Antelope say that the preamps are designed to get the best out of any mic, though using the Edge and Verge mics, which have been specifically designed for the purpose, will presumably deliver the best results when applying modelling. The Discrete 8 features eight fully discrete mic/line preamps, eight analogue outputs, a stereo monitor output, two headphone outputs and two guitar re-amp outputs, alongside S/PDIF in and out, two ADAT optical ins and outs and word clock in and out.

Clap your Hans
Spitfire release Hans Zimmer Percussion sample library

With a back catalogue of blockbuster film scores, from Gladiator to The Dark Knight trilogy, not to mention his recent live concert tour, Hans Zimmer is one of the biggest and most visible names in movie music. He pioneered the use of synths and samplers alongside conventional orchestration, but it’s Zimmer’s penchant for using ethnic percussion instruments to drive his dramatic, bombastic cues that has had the biggest influence on contemporary scoring. That’s the focus of this latest collaboration between Zimmer and British sampling powerhouse Spitfire Audio.

Billed as the ultimate cinematic drum sample library, Hans Zimmer Percussion features samples recorded at Air Studios, produced and mixed by the man himself. It includes many of Zimmer’s go-to percussion instruments played by his first-call percussionists, ranging from taikos, tombeks and tam tams to buckets, snares and anvils. Featuring both ensemble performances and key solo instruments, the library promises to deliver not only the epic, thunderous drum cues you would expect but also whisper-quiet layers suitable for more delicate and atmospheric cues. Containing over 48GB of uncompressed audio, the library provides close, room and ambient mix perspectives with up to nine round robins and six dynamic layers per hit. Running in Native Instruments’ free Kontakt Player, Hans Zimmer Percussion is available now priced at £349. A second library — Hans Zimmer Percussion Professional — is also in the works and should be released soon. Featuring additional mixes from Junkie XL, Geoff Foster, Alan Meyerson and Steve Lipson, plus CPU-saving stereo mixes of the ensemble content and 10 additional individual mic positions, it’s available to pre-order now, priced £429.

www.spitfireaudio.com
FOCAL I SHAPE

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The three monitors, Shape 40, Shape 50, Shape 65, are all made in France and integrate five innovations to maximise acoustic transparency. Designed to meet the needs of nearfield monitoring, Shape monitors combine an ingenious design and numerous settings optimised for the acoustics of small listening rooms.

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Rapid reference
Sonarworks Reference 4 adds zero-latency monitor correction

The latest version of Sonarworks’ speaker and headphone calibration and correction software has been released. Reference 4 adds a range of new features including ‘zero latency’ processing, meaning that the plug-in can be used to remove coloration and provide a more accurate sound while tracking as well as mixing.

As with previous versions, Sonarworks Reference 4 comes in two different editions. The Headphone Edition (£99) applies convolution-based EQ to correct the frequency response of your specific model of headphones, delivering either a completely flat response, a custom frequency curve or one of several presets designed to imitate the sound of various popular speakers or headphones. The library of supported headphones has now grown to 101 different models, while existing profiles are said to have been updated to offer greater accuracy. In addition to these facilities, the Speaker Edition (£249) does the same thing for speakers in a room, with a companion measurement app that can be used to calibrate the plug-in to your specific listening environment. Version 4 is said to offer improved usability, enabling reliable speaker measurement in under 10 minutes. Both versions also include the Systemwide stand-alone application that works with Windows and Mac OS as a virtual audio driver, applying correction to all system audio. Alongside the software, the Speaker Edition is also available bundled with the XREF20 measurement mic (£299), while the Reference 4 Premium Bundle (£699) includes the Speaker Edition software, the measurement mic and a set of pre-calibrated Sennheiser HD650 headphones.

www.sonarworks.com

Sound On Sound is saddened to hear that Seth Firkins died in his sleep on September 23rd this year. Seth, who described his approach to mixing in May’s Inside Track feature, was a highly talented engineer, with credits on records by Jay-Z, Rihanna and Drake. However, he was best known as in-house engineer and a key member of the team behind Atlanta rapper Future, also working with a host of other Atlanta artists such as Gucci Mane, Young Thug, Zoey Dollaz and Mike Will. At the time of going to press, the causes of his death remain unknown. In his obituary, the Firkins family asked for any donations to be made to the Academy of Music Production Education and Development (AMPEd) youth program in the engineer’s home town of Louisville, KY.

www.ampedlouisville.org

DPA Microphones have announced a significant update to the technology found in their renowned miniature mics. The new Core capsule features a redesigned amplifier that promises to not only increase the dynamic range of these mics, improving performance at high SPLs, but also lower distortion and improve clarity across the whole dynamic range, from whisper quiet to screaming loud. The first mics to receive the new Core capsule will be the d:screet 4060 and 4061 lavalier mics and the d:Fine 4066 and 4088 headset mics, providing up to 14dB additional dynamic range at one percent THD. Omnidiirectional Core models will also boast water and moisture resistance thanks to their nano-coated, hermetically sealed electronics. These new models will be shipping very soon, while the rest of the d:screet and d:Fine ranges will be available with Core technology in 2018.

www.dpamicrophones.com

Wearable subwoofer suppliers Subpac have launched the new and improved Subpac M2X. Billed as a ‘pro’ model, the purpose of the M2X is to allow the wearer to physically feel the low end without having to monitor at ear-battering volume levels. It’s designed to give live performers and DJs a more direct connection to the beat, to help producers understand how their music will translate to a club setting, and generally to provide a more immersive and inspiring experience. The M2X covers a frequency range from 200Hz down to a bowel-shaking 5Hz and is said to be near silent in use, raising the possibility of monitoring using headphones and an M2X yet getting a full-body experience. Weighing 2.25kg (5lbs), it’s powered by a built-in rechargeable battery, giving a quoted seven hours of use between charging. The M2X connects via a mini-jack cable with a pass-through output for headphones, while there’s also a Bluetooth streaming input for non-critical listening. The Subpac M2X is available now, priced £349.

www.subpac.com

DPA Microphones

Sonarworks Reference 4 adds zero-latency monitor correction

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www.sonarworks.com
Today’s modern mixing consoles fully integrate with your DAW combining the convenience of “in the box” workflows with the sonic advantages and tactile hands-on responsiveness only offered by analogue consoles. Feature-rich, compact designs are now more affordable than ever, making it possible to truly have the best of both worlds. Contact us today for friendly, expert advice.
Native Instruments
Maschine 3 & Komplete Kontrol 2
Controller Instruments
Are the Maschine 3 and Komplete Kontrol 2 enough to keep NI at the top of the integrated controller/software game?

SIMON SHERBOURNE

New versions of NI’s flagship controllers bring the onboard display features of the Maschine Studio to both the Maschine and the Komplete Kontrol S-series (KKS) keyboards, offer multiple usability enhancements, and bring them up to date with the latest Maschine software workflows.

The new controllers share a number of features and the keyboard can now work closely with Maschine software, so we’ve reviewed them together. However, the keyboard shouldn’t be confused as a Maschine controller: it’s primarily a master keyboard and plug-in instrument workstation. I’ll look at them separately, but note similarities and differences as we go.

First Impressions
The new Maschine is a sleek, solid wedge, with simple, straight edges that free up surface area to accommodate the larger screens and extra buttons. The unfussy lines, surface finish and backlit button legending create a definite Push 2-like impression, which is to say classy and modern.

To summarise what’s changed for skim-readers, the new hardware squeezes in bigger, better pads, more dedicated...
function and mode buttons, big colour displays, touch-sensitive encoders, a unique multi-directional push encoder, a touch strip, and a built-in audio interface.

A mains adaptor is supplied in the box, but both the Maschine 3 and KKS keyboard can run from USB power. As with Push, a mains power connection makes the screens brighter, but (unlike Push) they’re perfectly usable without it, and in fact I hardly ever plugged in.

**Finger Food**

The all-important pads have been significantly enlarged, leaving just a sliver of a space between them. All labelling has been moved on to the pads themselves to facilitate this. The span between the centres of the pads has been kept the same, so it should feel completely natural moving from the earlier models. Impressively, the response seems to be completely even across the whole area of each pad, so you can take full advantage of the new size. This makes the pads forgiving of sloppy accuracy, but more importantly you can play rolls, triplets and fast 16ths with multiple fingers on a single pad.

These are the first pads I’ve used that compete with those on the latest generation of Akai MPCs and controllers. They don’t have quite the same velvety texture, but they match the sensitivity and even response, and don’t have Akai’s tendency towards over-sensitivity and double triggering. They are simply brilliant.

Above the pad grid is a new strip of buttons for switching the grid’s mode between Pad (drum kit) mode, Keyboard (notes mode for instruments), Chords and Step. This is a big improvement on the previous models where mode changes were often a secondary Shift function.

**A Nice Touch**

An entirely new performance input is the Smart Strip. While the KKS has returned to traditional wheels, it makes total sense to get performance control on to the pad controller via touch. And this time the designers have really nailed it. On the KKS I was never confident that I’d left the mod ‘wheel’ at zero. On the Maschine 3, there’s a seamless gradient from the surface into the touch area so you know you’re picking it up or leaving it at the very edges of its ‘travel’.

The strip has several modes. Pitch and Mod perform these traditional roles, which weren’t available on previous models. Perform mode will be familiar to Jam users, giving you touch control of a key parameter from Maschine’s Performance FX. Shift-Perform lets you choose and assign a Performance Effect to the current Group master. By default, Performance FX are in Touch Enable mode, so you can drop in and add momentary interest then let go and return to the clean signal.

Finally you have Notes mode, which is, once again, a fantastic refinement of a feature from Jam’s touch strips: note strumming. In Jam’s Notes mode you use the grid to assign notes which can be ‘strummed’ or arpeggiated by running your finger up the strips. On the Maschine 3, the notes can be selected in real time by holding pads, resulting in a truly playable and expressive new way of performing.

**Screen Scene**

The most prominent new feature shared by the two new controllers is the dual display section. The screens

### NI Maschine 3 £469

**P R O S**

- The screens.
- The pads.
- The new rotaries.
- Performance input with touch.
- More dedicated buttons.
- Sound previews.
- USB or mains powered.

**C O N S**

- Ideas mode not well supported.
- We’re still waiting for performance capture and audio tracks.

**S U M M A R Y**

The MkIII is the new face of Maschine, packed with brilliant new hardware features and consolidating all the advances in the platform ready for the next era.
have the same size and resolution (and functionality) as those found on the Maschine Studio, but with increased brightness and colour saturation, despite the reduction in power consumption that allows you to work unplugged.

I’ve only had a couple of brief plays with the Maschine Studio previously, and wondered if the years of obsessing about what I might be missing would ultimately lead to disappointment when I finally got those screens: it didn’t. All the screen functionality of my MkI is present and works the same way, but with more space and enhanced colour graphics. And there’s much more besides, including the high-res colour mixing and browsing modes.

But the key difference, and the one that makes this Maschine a single-focus, hardware workstation, is the Arranger view. This brings timeline and pattern editing to the surface in a way that’s quick and effective enough to stop you reaching for the mouse. You can toggle between viewing Sections (the high-level arrangement timeline) and Patterns (your clips and their individual notes). The left screen always shows a complete overview, while the right gets you into the details via zoom and scroll knobs.

In Section mode the left and right cursors select one section, or you use the pads to jump directly to any position. The knobs under the left screen then let you choose a Scene for that Section, move the current scene, or adjust its length. Switching to Pattern mode displays the pattern from the active Group in the current Section or Scene. It’s then surprisingly easy — once you’ve learned how — to select notes and adjust their position, length, pitch and velocity.

In the past, the left and right cursors would select one section, but the new features of the MkIII allow you to work like you did in the past, or you can toggle between viewing Sections and Patterns. This brings timeline and pattern editing to the surface in a way that’s quick and effective enough to stop you reaching for the mouse. You can toggle between viewing Sections (the high-level arrangement timeline) and Patterns (your clips and their individual notes). The left screen always shows a complete overview, while the right gets you into the details via zoom and scroll knobs.

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**Design Of The Times**

It would seem that one of the goals of the new hardware is to expose and streamline features and workflows that have developed recently in Maschine. A notable example of this is the dedicated Lock button on the MkIII (originally seen on Jam). This grabs a snapshot of all current settings: mutes, sound and effects parameters, etc. A subsequent press recalls that state, allowing you to go off on crazy flights of fancy and change everything during a breakdown, then bring everything back in an instant. It’s also possible to store multiple Snapshots on the dedicated Locks pad view, accessed with the Shift button, and set morph times for gliding between the states.

Another innovation that came with Jam was the Ideas view, which separates Maschine into two workspaces: Ideas, where you create Scenes from your pool of Patterns and Groups; and the Arranger, where you place Scenes into Sections. On the MkIII you can work like you did in the past, but Shift-Scene toggles you between...
the two workspaces in the software. Unfortunately there’s little on the hardware to tell you this has happened, but in this mode Pattern selection from the pads makes assignments to the current Scene. As before, to assign Patterns from each Group to a Scene requires you to methodically switch between Groups, and this is where Jam really speeds things up. The KKS keyboard team have, however, added a great mode for controlling Scene/Ideas mode (see Rise Of The Maschines) and I hope this will come to the MkIII soon too.

Variations also gets a dedicated button, and this time it’s an improvement on the Jam implementation, because the various functions can be accessed from the in-line displays instead of via the software overlay display/scroll wheel system employed by Jam.

Joy Division

Both units feature the new 4-Directional Push Encoder design, that replaces the older pairing of encoder and directional buttons. It’s a custom design, which adds four-way directional switching by moving the encoder. It’s a kind of ‘joystick’, but let’s never call it that. The encoder has all the functionality and modes as the one on the earlier models, and takes some of the duties of the Maschine Studio’s wheel.

You can use the main encoder for sound browsing, much as you did on the other Maschines and original KKS, and it’s much easier not having to combine it with separate cursor buttons. However, as browsing is now so good from the screen encoders there’s not much call to do this.

The encoder is also used in the mixer, where it can navigate between tracks, jump between the Groups and Sounds mixers, and set levels and pans. It also has the same dedicated mode buttons as the MkII. There’s definitely room for it to take on more work though. In particular, I expected it to function in the Arranger, and I’d like to be able to park it on current parameters by touching any screen encoder.

Audio

Maschine 3 is the first Maschine to include a built-in audio interface. There’s stereo line output plus a headphone output, twin line inputs and a dynamic mic input. All connections are quarter-inch jacks on the rear of the unit along with their associated level controls, which isn’t particularly convenient for the headphones, but is understandable given the low profile at the front.

The line and headphone outputs are separate pairs of channels, which is ideal for a live setup if you want to configure a separate Cue feed. Maschine stand-alone has its own Cue system which lets you toggle mixer channels between the Master and Cue busses. Sample and patch preview, and the metronome, are also automatically routed to the Cue bus. Unfortunately, there’s no way to assign channels to Cue from the Mixer view — you have to do it from each Group or Sound’s dedicated channel view. This really restricts the usefulness of this feature in a live situation.

It would have been useful to be able mirror the two pairs of output channels, and apparently this feature is in the works. Arguably, though, in most non-live situations you’re only likely to use the line outs or headphones, not both. In fact, for most studio-based work, your computer will probably have another interface anyway, so Maschine’s audio is really there for live use and using headphones when out and about.

As the arms race escalates among performance samplers, it’s no surprise that NI would want audio inputs on board Maschine. The line inputs are certainly useful, and spurred me on to do some direct sampling from my iPad music apps and a hardware synth. I sometimes overlook sampling in Maschine, but I actually think it has the most straightforward workflow for capturing and editing audio, then slicing to pads or zoning to instruments, of any

NI Komplete Kontrol 2
£469

PROS
- The displays.
- Sound previews.
- Same great keyboard.
- Dedicated Maschine kit mode.
- DAW mixer control.
- Unique Maschine Scenes mode control.
- USB bus powered.

CONS
- No ‘smart’ mapping to DAW built-in instruments/devices.
- DAWs with full support currently limited.

SUMMARY
Displays with touch-aware browsing and instant auditioning make the S-series even more enjoyable and powerful.
comparable system including Push or MPC. I’m not quite so convinced about the usefulness of the mic input. However, an XLR input would have necessitated a thicker unit overall, so a dynamic mic is probably the right compromise. What would be more useful to me, though, would be an instrument input for connecting a bass or guitar.

Form & Function

Let’s turn our attention to the KKS2 now. While the Maschine hardware has clearly moved up in the world in terms of look and feel, it would be hard to improve on the original Komplete Kontrol’s physical beauty. The KKS2’s revisions prioritise usability and feature upgrades, sometimes at the expense of the original’s sleek lines, but the compromise is worth it. The screens necessitate a slightly stubbier chassis, and replace OLED parameter bars that were easy to read below the rotaries. But what the screens bring to the table far outweighs this. Likewise, the chunky Mod and Pitch wheels might not look as neat as the previous touch strips, but the feel and playability are superior.

A notable design improvement is to the Light Guides. These are the multi-colour lights used to great effect by many NKS format patches to show zones, sample switches, even step sequences, and are used in Scale and Chords modes. On the MkII, the lights have been significantly narrowed, and somehow this makes it much easier to see which light belongs to which key.

The keys themselves are just lovely to play. They again use a semi-weighted, Fatar keybed with aftertouch. It could be exactly the same one, and my original may have loosened up over time, but the new one feels a touch weightier and even more stable.

Plug-in Workstation

The KKS keyboards provide advanced browsing and control of all your plug-in instruments via a host app also called Komplete Kontrol, which you can use stand-alone or as a VST, AU or AAX plug-in. There is a generic MIDI controller mode, where you can create and recall templates. (This feature has moved from the separate editor app to Komplete Kontrol, and has temporarily lost the ability to make key zones.) But you get the juicy features when you’re working with your soft synths in Komplete Kontrol, or for that matter, Maschine.

When we reviewed the first Komplete Kontrol, Komplete really did mean Komplete. The NKS preset format created by NI, and used by the KKS, was only supported by their own Komplete instrument and effects bundle, and you couldn’t load other plug-ins. Even as primarily a dedicated controller for Komplete the first release was a compelling product, but it left a lot of gaps that Akai could fill with their Advance keyboard and VIP host software.

Since then many other plug-in developers have embraced the NKS format, meaning they get tailored control mapping and inclusion in the KKS and Maschine browsers. Non NKS plug-ins can also now be added in Komplete Kontrol and benefit from auto-mapped control assignments, but you can only add them via the software interface. User customisation of plug-in mappings is also possible these days, although I don’t know of a way to change these for a whole plug-in (as you can with VIP), only per preset.

Komplete Kontrol reaches beyond its plug-in workstation capabilities, and the KKS2 is clearly aimed at professional music makers.

Studio Line

A common question from potential Maschine 3 buyers has been about the fate of the rest of the Maschine range, especially the Studio. The MkIII ticks all my boxes, having everything the Studio has that I wanted, without any of the stuff that put me off. However, it’s natural to wonder if there’s a Studio MkII around the corner. I don’t have any insight into this, but my hunch is that the Mikro and Studio will simply be discontinued and NI will be able to focus on a unified workflow around the MkIII, Jam and KKS in the future.

But... if I was speculating about a next-generation premium Maschine model, I’d be picturing touchscreens. Parts of the current interface such as the browser, mode selectors in channel view, device chain view, etc, already seem optimised for touch, to the point where a couple of times I absent-mindedly tried to tap or swipe the screens. I’m also imagining more audio I/O with phantom power and an instrument input, plus CV outputs to challenge the MPCX.

One thing that stood out was that the MkIII’s profile and height don’t match Maschine Jam; although as a tag team they are by far the best way to use Maschine, they don’t sit nicely together on the desk. As the Jam is only a year old, it seems a bit soon for a MkII, but if NI were considering a refresh, it would be a chance to upgrade the grid with pressure sensitivity...
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- Wi-Fi control

motu.com/828es
Rise Of The Maschines

With the MKII, the KKS has essentially become a fully fledged Maschine controller. Browser, Plug-in and Mixer modes all work exactly the same as on the MkIII. To the left of the screen is a column of buttons dedicated to Maschine. Scene mode recreates Maschine’s Ideas view on the screens. Scenes are selected from the top buttons, and you assign patterns from the Group columns using the rotaries (or master encoder).

In Pattern mode you can view and create Patterns, though you can’t edit them as you could on the MkIII, apart from Quantising or changing the Length, but it’s enough to keep your focus in one place and see what you’re doing. And finally there’s my top feature request from the first keyboard: a dedicated mode for triggering drum kits on the keyboard instead of pitching individual sounds. Brilliant stuff!

One of my favourite things about Maschine is that you can use it with any one of the controllers, but each one specialises in different things, so when you bring them together it gets better and better. Though all devices sync to the same Group, you can have different views on each. Right now, the MkIII, KKS 2 and Jam working together make an incredibly fluent hardware studio interface that’s just waiting for the software to catch up and provide what we need for more stand-alone operation.

librarian and hardware-mapping duties with the excellent Smart Play performance enhancements, which aren’t new but deserve a mention. There are arpeggiator, scales and chords modes, all of which interact with each other. As well as an Easy Scale mode that remaps scales, you can use the Light Guides to show you which notes are in key — a great education feature.

DAW Kontrol

The KKS MkI has been the ’Smart’ DAW Master Keyboard against which I judged others. Several MIDI keyboards offer ways to directly navigate within your DAW and map to functions via direct scripting or Mackie Control, etc. The KKS relies on a hybrid of some direct support of DAW features and the use of Komplete Kontrol as a plug-in host to get the best experience. I tested with Ableton Live and Logic Pro X; Cubase support is to be added soon.

The original KKS could select between your DAW’s tracks, and control the transport. Its killer feature though — and the one that the rival VIP-based keyboards from Akai, et al couldn’t match — was to detect when you selected a track containing an instance of the Komplete Kontrol plug-in and snap the keyboard to it. Others DAWs can work with the KKS via the Komplete Kontrol plug-in and generic MIDI mode, but won’t get the benefit of these advanced integrations.

In the KKS2 the original features are joined by a few core DAW functions and mixer control. The former includes metronome toggle and tap-tempo buttons (which actually replaceREW and FFW), Undo, Quantise and Automation Enable. Like Push, only functions that are currently available are lit via the backlit legending on the buttons.

In Mixer mode your DAW mixer channels appear on the displays in a monochrome version of what you see in Maschine. The knobs adjust levels and have a secondary mode for panning. The buttons above the screens select tracks, and also serve as Mute and Solo buttons when combined with the appropriate modifier button.

KKS 2 has more DAW Control and big improvements to sound browsing in the Komplete Kontrol plug-in.

Alternatives

Competition is only getting hotter for Maschine, with new MPCs, Elektron devices, the excellent newcomer Beatmaker 3 for iPad, and of course Live, all of which are great in different circumstances. For me it boils down to really specific details between Push 2 and Maschine. Live has real-time performance capture from the Session view to the timeline; about the best equivalent in Maschine is recording its audio outputs. However, Maschine gives you full timeline editing and arrangement on the hardware, which is impossible on Push. Live has linear audio recording, but Maschine can run inside other DAWs and lets me work in both Live and Pro Tools. I’m not helping much am I?

Komplete Kontrol has the VIP-powered keyboards from Akai, M-Audio and Alesis wanting to eat its lunch. VIP is very powerful, with a full internal mixing and snapshot environment, and has pre-made mapping for just about every plug-in out there. However, the hardware control is definitely not as polished as NI’s, and of course you don’t get the Maschine integration. If you’d prefer to work without an instrument host plug-in, Nektar take a different approach with their controllers, by squeezing as much smart functionality as they can out of your DAW’s native MIDI control options.

Alternatively, you can use the directional encoder to control the mixer with one hand. Left and right selects through the tracks, and turning adjusts the level. In Live’s Session view, the encoder also behaves as a cursor that navigates through the clip grid. You can use this to launch clips or scenes, or start and stop recording on an empty clip, which I found genuinely useful.

Where the S-series still falls short is control of your DAW’s built-in instruments and devices. Because Logic/GarageBand and Live include instruments that are not available as plug-ins in any other host.
they can’t be loaded in the KK shell. Many controllers, Nektar for example, can map to Logic’s Smart Controls, or Live’s device Macros, meaning you can control the built-in devices without any manual mapping. In fact, Maschine 3 and Jam both have scripts and templates that do an excellent job of this in Live, so it would be nice to see the same on the keyboard.

In Summary
The KKS2 is a joy to play, with its lovely keybed and smooth touch encoders. The sound-browsing experience is massively improved, with the screens freeing you from the software interface, and ‘Prehear’ auditioning making the whole concept of a unified patch library for multiple plug-ins really work. The Komplete Kontrol plug-in has stayed simple, providing a simple workflow and hardware experience and letting your DAW do its job, although this rules out the kind of composite patches or live setups possible with Kore or Akai’s VIP. In any case, much of this could be done inside Maschine, which the KKS can now control.

And so to Maschine: I’ve spent an unhealthy amount of time in the last two years hopping between the various beat workstations: Maschine, Push 2 with Live, Launchpad Pro with Live, MPC, even Beatmaker on the iPad. None of them is perfect, but Maschine is where I inevitably return. It’s the place where I can most quickly capture ideas and, more importantly, improvise an arrangement with variation and interest.

Yes, we’re still waiting for some key software features that have been flagged in every review since version 1.0 — in particular performance capture and audio tracks — but as a hardware upgrade the Maschine 3 is everything I wanted and more.

Maschine 3 £469. Komplete Kontrol 2 £499 £649, £539. Prices include VAT.

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Sontronics have had great success building mics for specific applications — but their new flagship is designed to take almost anything you can throw at it.

Liquid Metal

The microphone is predominantly made here in the UK — while some of the metalwork is manufactured by Sontronics’ partners in China, the electronics are all domestic, and the product is assembled here, at the company’s facilities in Dorset. Referring back to premium electronics for a moment, Coley tells me, “There are three capacitors that are absolutely critical to the audio path: a polystyrene capacitor, and two silver mica capacitors. All have one percent tolerance. They’re as good as you can get.” While he remains understandably tight-lipped about exactly who supplies them, he will say that they’re also of UK manufacture.

The machined brass body features the familiar bold, screened logo, making it clear which side of the mic is which. The microphone weighs in at a reassuring but not outlandish 2317g and measures 250 x 90 x 80mm. It comes in a velvet-lined wooden box, which sits in a foam-lined aluminium road case, the various sections of which snugly enclose the power supply unit, shockmount, eight-pin audio cable and the power cable. This selection of accessories, robust and functional if not especially beautiful, will be familiar to any Aria devotee, with the exception of the larger shockmount and the addition of the continuously variable rotary polar-pattern selector on the front panel of the PSU. Also on that panel are a -10dB pad switch, and a 75Hz high-pass filter switch. Further technical specifications are as follows:

- Frequency response is 20Hz-20kHz;
- Sensitivity is 18mV/Pa, or -33dB ±1.5dB (0dB = 1V/Pa at 1kHz);
- Impedance is ≤200Ω;
- Equivalent noise level is <12dB (A-weighted); and
- Maximum SPL (for 0.5 percent THD at 1kHz) is 125dB (more on headroom later).

As mentioned above, this is not simply a premium reinvention of the Aria design, with the addition of a few bells and whistles. Sontronics often voice their products in an application-specific way — the Delta for guitar amps, or Aria for vocals, for example — but the Mercury is aimed at more general use out on the studio floor. And a studio mic this certainly is — it’s not really ‘one for the road’ as such. The continuously variable pattern, and lack of any deliberate tailoring in the frequency response, offer the mic to a broad range of applications and techniques. It has a great deal of headroom, on paper, yes, but perhaps even more so in practical use. It’s designed to be a ‘go to’ mic, the one you put up by default out in the studio — maybe the one you don’t really take down. It’s designed to be the overhead, the room mic, the piano mic, the acoustic guitar, the amp... or perhaps the main vocal.

Sontronics Mercury

£1349

Pros

- Superb sonic performance regardless of cost.
- Affordable.
- Presence and transparency.

Cons

- Precise position of pattern control dial could be more obvious.

Summary

In the Mercury, Sontronics have achieved new heights with yet another star that doesn’t cost the earth.
It’s supposed to take what you throw at it, to keep your options open, not to box you into a corner sonically. Its development was characterised by three years of extensive listening tests, and it’s designed to sound superb at all times. So... Does it?

**Out Of This World?**

If I had to pick two words that best describe the sound of the Mercury microphone, they would be ‘present’ and ‘transparent’. There’s a clarity here, a definition and coherence all the way from the bottom to the top of the frequency response. It sounds modern, open, solid and defined.

The design choices, both electrical and mechanical, contribute to this — the relative separation of the headbasket from the body yields an increase in presence, and the tight tolerances in those capacitors yield advantages in terms of the phase between the dual diaphragms. Where two back-to-back diaphragms are combined to achieve variable patterns in this way, they would, in an ideal world, be identical. In practice they never can be, but the quality and stability of the circuit to which each is connected affects how close they can get to that ideal. There’s a distinct lack of that veiled smearing that affects some lesser products, and no part of the frequency spectrum that feels indistinct or cloudy, so it’s easy to imagine why Coley felt that there was some benefit to this additional production cost, even without the ability to make the comparisons that he himself made on the test bench.

I made a number of drum recordings at The Old Chapel Studios in Nutbourne, which showcase these qualities. Drums can tell us a lot about a microphone. They exhibit a wide range of frequencies, and in a relatively ambient space like this, they reveal off-axis coloration. Any loss of definition can have an obvious negative impact, and a lack of headroom quickly flags itself up as a ‘thwocky’, constrained character on louder transients. Both drummer James Ivey and I found none of those disadvantages — the clarity, presence and open nature of the sound was impressive. Transients were crisp and clear, but the low-end depth and definition indicate that this isn’t simply a question of being bright. Yes, the microphone is bright, but it’s a quality borne out of coherence and transparency, rather than artificial boost or deliberate resonance. Moving between omni, cardioid and figure-8 patterns, and between spaced and coincident positions, the flexibility of the mic reveals itself quickly. I found myself able to dial the room sound up and down in the transition between omni and cardioid, and exclude certain elements of the kit using the deep nulls of the figure-of-eight pattern. The room blended beautifully with the direct sound of the kit, allowing for a lively, natural portrayal.

At this point, it’s worth coming back to the subject of headroom. This is a mic with more apparent headroom than the specifications would suggest. Coley confirms this: the numbers are, he says, conservative, with several more decibels in hand. And so, as a drum overhead, this is a microphone that remains crisp and clear on the transients even without the use of the -10dB pad. This is a considerable advantage for a mic of this type; so often I find myself ultimately swapping a condenser overhead out because it just can’t cope with the louder sections of a track. I’m forced to go for a slightly different character as a result, but

**Plotting Course**

The frequency response plots exhibit a fairly typical sort of shape for a modern, large-diaphragm microphone of this type. Broadly speaking they show a relatively flat response, with a gentle high-frequency lift. In my experience, you can’t really draw any conclusions about the sound from these sorts of plots when taken in isolation. Sometimes you can see evidence on a plot for a characteristic you can hear when you test a microphone, but in general I’d caution against attaching too much importance to them. I’ve included them here for the sake of completeness, and if I was asked to summarise how the plots relate to the results on offer, I’d say they present no surprises.
THE VERY BEST IN PRO AUDIO
this microphone represents a legitimate option in this respect. On acoustic and electric guitars, that same definition and coherence yielded some beautiful recordings. Again the continuously variable pattern behaved almost like a tone control, moving the listener closer to the source as the omni became cardioid, closing out the room and increasing the proximity effect.

A grand piano is an instrument with a complicated, harmonic high end and upper mid-range. If a microphone ispeaky, a close position can often result in an uneven, confused presentation. Typically, you then need to pull back outside the instrument, away from the soundboard in order to minimise this ugliness. But in the context of a pop or rock recording, that might not be appropriate — often you might prefer to be ‘up close and personal’, perhaps with the lid down to the ‘short stick’, with heavy blankets or covers to increase isolation from other instruments in the room. Placing a pair of

revealing an unnatural amount of ‘esses’ and fricative content.

This is characteristic of almost all of what one might term ‘posh, modern’ microphones. It’s one of the reasons why old U67s, for example, are so desirable on vocals — they control the tap end, by their nature, as does the Aria. And one of the things about old U67s, again as with the Aria, is that they could sometimes benefit from a greater degree of presence and clarity in other applications. As ever, it just depends on your source. I was able to achieve some beautiful vocal recordings during the review period — a degree of detail on a dark, delicate, almost mumbled vocal that other options, Aria included, were unable to compete with — but Sontronics have designed this mic to spend its time pointing primarily at instruments out in the studio, and it’s worth keeping this in mind if you’re buying blind.

So that’s the limitation. It’s not really a criticism, more the illustration of an inconvenient universal truth in audio recording — the microphone is irrelevant without the source. In other words, if you need to record a variety of different sources, the chances are you’re going to need a variety of different microphones with which to do so. But I do have a criticism. The continuously variable polar-pattern control has markings screened on the casing, showing the pattern at three positions: hard left (at seven o’clock) for omni, straight up (12 o’clock) for cardioid and hard right (five o’clock) for figure-8. Considering the significance of stereo applications with a product like this, and that much is made by Sontronics of the near-identical nature of two examples of the Mercury, it seems to me that more in the way of a visual means of matching patterns between two mics could be important here. Sure, it’s fine if you’re going omni, cardioid or figure-8 — you can’t go far wrong — but in the intermediate positions it’d be nice to have a bit more of a chance at precisely aligning the two controls. I don’t really want to guess where 10:45 is in a dimly lit live room. Going a step further, a little dab of fluorescent paint in the groove indicating which direction the control is pointing would be a nice touch too. This design is not the worst offender at any

Alternatives

There are a number of possible alternatives in terms of switchable- or variable-pattern large-diaphragm condensers. In this price range, products from the likes of Blue, Peluso and Telefunken all suggest themselves. I would however suggest that even if your budget extends further, putting more expensive options from Brauner, Neumann, or Microtech Gefell within reach, it would still be worth auditioning Mercury before making a final decision.

Hearing Is Believing...

I’ve provided a series of audio examples to illustrate the capabilities of this mic, including files from the drum recordings with James Ivey at The Old Chapel, an electric guitar recorded here in my own studio, as well as a nylon guitar also recorded here in my control room, and taken from the Oktoba single ‘Chance’. You can hear them at http://soundonsound.com/mercury-1317.

budget in this respect. It’s something that a number of manufacturers could possibly consider, especially since with a continuously variable pattern control they’re not cloning the classic German designs of the 1940s and ‘50s.

Mercury Rising

This is another extremely strong product from Sontronics. It’s a fresh design with a nod to classic form, rather than allowing itself to suffer the restrictive inhibitions of a clone. It also represents extraordinary value: as with previous Sontronics products, it doesn’t burn its budget on excessive cosmetic gloss, but preserves it instead for the components that affect how it will sound. You could buy a pair for the same price as a single mic of similar sonic quality. Of course the devaluation of Sterling plays a part in the way professional audio products now stack up on the Google ‘Shopping’ tab, but that’s not the only contributing factor. We’re coming to expect Sontronics to punch above the price range, and in this case, again, they do.

If, just for a moment, we buck the trend for romanticising vintage equipment and consider investment from the standpoint of a professional with a job to do, a pair of Mercurys, an Aria, a handful of Solo dynamics and a couple of Orpheus condensers or Sigma ribbons and you’d have the basis of a mic collection that would stand tall in any typical studio session — and all for less financial outlay than a single eye-watering foray onto eBay. I’m not the only one to feel that way — Gary Barlow was the company’s first customer for a Mercury, and the latest Ed Sheeran record relies on an Aria as well as an assortment of other products from the Sontronics stable. This is, again, a product that puts its money where its mouth is. I might just do the same.

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www.sontronics.com

December 2017 / www.soundonsound.com
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Trident 78 Series

Analogue Mixing Console

Derived from the larger flagship Trident 88 consoles, this new range offers many of the same features for a lot less money.

HUGH ROBJONNS

The Trident brand dates back to the early 1970s and a very influential London recording studio that set up a console manufacturing business. Although the studio and console businesses went their own separate ways, Trident Audio Developments still make products that draw upon that heritage.

Currently, Trident offers two hardware channel strips, a 500-series EQ module and the 10-slot Deca-Dent rack and power supply, as well as their flagship product, the Trident 88 large-format analoge console. The subject of this review is a new, more affordable version of that console — the Trident 78. Like its upmarket sibling, it was designed by Taz Bhogal, whose experience dates back to the development of the original Trident Series 80C from 1986. It retains the same core design philosophy and signal path ideas as the 88; the keener price is possible thanks to some logical feature-set simplifications and a more cost-effective semi-modular construction.

Overview

The Trident 78 is available with eight (as on the review model), 16 (as in the main picture), 24 or 32 input channels, and all have the option of moving-coil VU or more affordable LED bar-graph metering in the meter-bridge. Following the traditional in-line concept, each mono channel strip accommodates two signal paths (channel and monitor), and it features six aux sends (four mono, one stereo), a four-band EQ derived from the Trident 80B console, a balanced pre-fade insert and a configurable balanced direct output.

Being an in-line console, the monitor path can borrow the entire equaliser section and the stereo aux sends, if desired, and it routes into the stereo mix bus via a rotary fader and pan-pot. The channel path can also be routed into the stereo mix bus and/or any of eight mono sub-groups (in pairs), via a large fader and pan-pot. For mixdowns, the channel line and monitor input connections can be swapped over at the press of a button, allowing the monitor signal to be routed through the full channel path, but the channel and monitor faders can’t be flipped over (as they can on most in-line desks).

The console always features eight mono sub-groups but these can be used as monitor returns if preferred, effectively working the console in the traditional split format. Each sub-group is equipped with balanced pre-fader inserts and can access two mono and the stereo aux sends. They also feature balanced outputs, post-fade metering, and a long fader, but not EQ. By default, the (post-fade) sub-group output is routed to the stereo mix bus via a separate rotary ‘monitor’ fader and pan-pot, but...
each group module also has its own monitor input connection, which can be selected to replace the post-fade sub-group signal, if required. Each sub-group also accommodates a stereo effects return which feeds directly into the stereo mix bus, and the first six sub-groups also carry the corresponding auxiliary master controls.

A master module houses the master stereo mix fader and simple control-room monitoring and talkback sections. There’s also a pair of external two-track returns, and both main and alt monitoring outputs for two sets of stereo speakers, with talkback to the groups and auxes.

Construction

Whereas the Trident 88 series consoles are built from individual channel strips, with each fader housed in a separate module, the new Trident 78 models are of “semi-modular” construction: each physical module carries four input or four sub-group channels including their faders. The simplified metalwork brings down the construction costs, obviously, but, importantly, it doesn’t adversely affect the operational ergonomics or make maintenance problematic or inconvenient. In fact, the largely surface-mount, but in critical places through-hole electronics for each channel are built on independent, vertically-mounted, circuit boards suspended below each module’s panel in the traditional way. Not only does this simplify servicing, but it ensures excellent crosstalk performance between adjacent channels. It also makes everything a little more reachable: the modules are about 14cm (almost six inches) shorter than those of the Trident 88, and the top of the meter-bridge is about 3cm (an inch) lower.

Another major cost saving comes from the choice of external connections. Whereas the Trident 88 features individual XLR or TRS connectors for every input and output, the Trident 78 is connected almost entirely via AES59 (balanced eight-channel Tascam format) D-subs. XLRs are provided only for the stereo master and main monitoring outputs (both are duplicated on the D-subs). This much less expensive arrangement makes it quicker and neater to install the console, and brings some welcome benefits such as balanced inserts across the console. (Currently, the Trident 88 has only unbalanced inserts.)

The inputs are all transformerless balanced types, while the main stereo, sub-group, aux master, and primary monitor outputs are active and symmetrically balanced. Impedance-balanced outputs are used for the channel direct outs, inserts sends, and the alt monitor outputs. Like its sibling, the Trident 78 can optionally be fitted with Lundahl transformers on the stereo mix outputs. (At the time of writing, the manual and schematics referred to the optional transformers being on the main monitor output. I’m told that this and a number of other errors in the schematics and documentation should have been corrected by the time you read this.)

I gather that the next production run of the master/monitor boards will be modified so that the output monitoring is taken post the transformer outputs (where fitted), so that the effect of overdriving them can be auditioned... which seems like such an obvious requirement that I’m surprised it wasn’t already arranged that way. Other production changes are being made to the input modules (see below), and taken overall I’m left with the impression that this console is, to some extent, a work-in-progress.

Power comes from a 2U rackmount switch-mode supply, which accepts 100-240 Volts AC. Terminals are provided to access and link the chassis and audio grounds, and a seven-pin Hirose connector carries the ±18V and +48V DC power rails to the console via a chunky cable. The PSU is cooled by a fan, which is a quiet one, though some might prefer to locate it in a separate machine room. The manufacturers clearly have confidence in their product: a two-year warranty is included (three years if the console is registered within 30 days of purchase).

Input Signal Paths

Having outlined the console structure already, I’ll now highlight some of the more interesting features. For anyone unfamiliar with the in-line concept, the monitor signal path in each channel strip was originally intended to handle the output from the corresponding multitrack tape-recorder.
The 78-series consoles are available in configurations with different numbers of input channels, and a couple of cost options. Prices exclude VAT.

<table>
<thead>
<tr>
<th>Input Channels</th>
<th>Meter Bridge</th>
<th>Price (£)</th>
</tr>
</thead>
<tbody>
<tr>
<td>8</td>
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<td>11,506</td>
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<tr>
<td></td>
<td>VU</td>
<td>12,223</td>
</tr>
<tr>
<td>16</td>
<td>LED</td>
<td>13,807</td>
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<td></td>
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<tr>
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<td>LED</td>
<td>19,117</td>
</tr>
<tr>
<td></td>
<td>VU</td>
<td>20,035</td>
</tr>
</tbody>
</table>

Lundahl output transformer option: £192.

The channel strips include bypass buttons for both the EQ section and the insert return.

channel, so a rough mix could be built up across the console on the channels that weren’t being used for recording. Today, those monitor inputs are more likely to come from a DAW/audio interface. The channel path is used to feed the recording signal to the corresponding track of the recorder, via the channel’s direct output. For mixing, the monitor inputs can be flipped across to the channel paths and routed through to the stereo mix bus via the big faders, and with full access to the aux sends. When tracking, the traditional ‘analogue’ practice was to record the processed signal after EQ, outboard dynamics and fader-riding (helping to overcome the limited dynamic range of the recorder). Today, though, it’s common to record a clean input signal to the DAW, and process this later, either inside the DAW or via the console and/or outboard.

The Trident 78 provides some configuration options for the channel direct output, though I feel they could still be improved upon. On the first consoles, jumper links on each input PCB provided post-fade and insert-return options, but both being post-EQ and post-insert (the channel insert point is fixed post-EQ here, whereas it is switchable pre/post EQ in the Trident 88), neither guaranteed a completely clean recorded signal (at least, not without engaging the EQ bypass, which makes it impossible to track clean while building a console mix). I’m told that all subsequent production runs of the Trident 78 (and 88) will offer a third option, which is a pre-fade feed. However, as the insert return connects directly to the fader via the insert switch, in all practical situations I can imagine, these two options are essentially identical.

It appears that the insert-return option is borrowed directly from the Trident 88, where it makes sense, since that console’s unbalanced insert socket has normalised send and return connections (so its insert return is effectively a pre-fade/post-EQ direct output, which always provides a signal of some kind and complements the post-fade option in an obvious way). The D-sub in the Trident 78 precludes automatic normalising across the console’s insert send/return connections, though. So if the insert switch is engaged (via the dedicated switch in the EQ section) but there’s no device patched in to the return, the insert-return option will provide no signal. That could be confusing in the heat of a session, so you’ll either want to get used to using the insert switches, or to connect the desk up to a normalled patchbay (I’m told that some manufacturers also make D-sub cables for this purpose).

I’d much rather see the third direct-out option being immediately post-preamp; this would always allow a clean signal to flow to the DAW/recorder, while starting to build the analogue mix.

Moving on to the mic preamp, this is identical to the Trident 88’s, although it lacks the input transformer option. It’s a discrete-transistor Class-A design boasting low noise, low distortion, a wide bandwidth (>100kHz), and an enormous headroom margin. The gain range spans +5 to +60 dB in mic mode (-20 to +35 dB in line mode) and a maximum of +17dBu can be accepted at the mic input, or +42dBu at the line input. The EIN figure is better than 128 5dBu (20Hz-20kHz, 1500Hz source, 60dB gain) — that’s within a decibel or two of theoretical perfection! Illuminated buttons on each channel engage phantom power, invert signal polarity, and select the line input.

The 88 model’s separate line gain control has been omitted; instead the output from the dedicated balanced line-receiver amp is padded down and routed through the mic preamp. This saves the expense of a second variable gain stage and rotary control, as well as panel real-estate, but it doesn’t compromise the signal quality — the 78 and 88’s line input’s tech specs are virtually indistinguishable.

Another button selects what’s displayed on the channel’s meter: the signal at the direct output (to monitor what’s sent to the recorder/DAW), or the monitor return signal (pre-fade). Another button labelled ‘I/P Rev’ swaps the line and monitor input connections, so that the monitor signal is available to the channel path and its long fader and aux sends for mixing, and the line input flows through the monitor path if extra mix inputs are needed.

The four-band EQ is slightly simplified compared with the Trident 88. All four bands enjoy ≥15dB boost/cut ranges but, instead of fully tunable high and low shelves, there are two selectable turnover frequencies (8/12 kHz and 60/120 Hz, respectively). The two mid bands are sweepable between 1-15 kHz and 100-1500 Hz, but the Q is fixed. There’s also a switchable 50Hz, 18dB/octave high-pass filter, and a button switches the entire EQ in or out. The whole EQ section can also be switched into the monitor path, if desired.

I’ve discussed the inserts already, but the two consoles’ auxes differ slightly too: the Trident 88 has eight auxes (four mono and two stereo), whereas the new console has six, arranged as four mono and one stereo. That’s a sensible place to make savings, as there are more than enough auxes for most applications, and all are switchable.
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pre/post-fade. The stereo aux (5/6) is also movable into the monitor path, if required, as it’s intended to serve as an artist cue mix.

The monitor path feeds the stereo mix bus via a rotary fader and pan pot, and has a dedicated mute button but no PFL/AFL facility (unlike the Trident 88). The channel mute kills all post-fade sends: the stereo mix, sub-groups, direct out (if so configured), and post-fade auxes. A pair of LEDS above the fader indicates signal levels at -20dBu (green) and +10dBu (red), with an internal jumper to meter either the pre- or post-fade signal. Unlike on many multitrack desks, there’s no destructive solo mode, but a button on each channel selects whether solo is PFL or (post-pan) stereo AFL (there’s no global PFL/AFL mode switching).

**Sub-groups**

The top of each sub-group module carries a long bar-graph meter which can be switched to show the sub-group output, group monitor return (pre-fade) or the corresponding aux master send. These LED meters are in the sub-group modules because the space above in the meter-bridge is occupied by a pair of large VU meters for the main stereo output/monitor signals and PSU status LEDs.

Aux send master controls sit below the bar-graphs with a rotary output level knob, AFL solo, and a button to send the post-fade aux output to the corresponding sub-group’s meter. Moving down the strip, the next section controls a stereo effects return (which routes directly to the stereo mix bus), with level, balance, and mute controls. On the Trident 88, the eight stereo effects return inputs are relatively well equipped, each feeding directly into the stereo mix via rotary level and pan/balance controls. They also have a basic tilt EQ, mute and solo buttons and, importantly, the ability to send the effects signal to the stereo aux 7/8 artist cue mix (for a ‘comfort reverb’, say) you can’t use the effects returns. Instead, you’d have to patch the effects signals into some spare input channels and access the aux busses from there, or (possibly a more practical solution in many cases) use the group module’s monitor inputs for the stereo aux returns. In this way, the re-patched return signals still route directly to the stereo mix bus via the monitor level and pan controls, but they also gain an AFL button and pre/post access to mono auxes 3 and 4 and the stereo aux 5/6. The group monitor inputs are mono, so eight groups could handle only four stereo returns, but that shouldn’t be a problem for comfort reverb, or for the few occasions where you want to route one effect output on to another effect send (something else that can’t be done directly from the dedicated returns).

As I mentioned earlier, the sub-group signal has its own balanced pre-fade insert, although the switch to activate the return is strangely hidden away between the aux master and effects-return sections. Again, there’s no EQ here, but the sub-group can access aux sends 3-6 (two mono auxes and the stereo one) for effects and artist headphone mixes, with the sends derived pre or post the monitor level control. The group output is controlled by a long, red fader and this feeds the stereo mix bus via a rotary level control labelled ‘Monitor’, and a pan-pot. There’s also a mute button and AFL monitoring, and a button to replace the sub-group signal, at the point it reaches the Monitor level control, with that from a separate external monitor line input. The post-fade sub-group signal always remains available at the corresponding group output.

**Master Module**

A blue stereo fader is provided to control the main stereo mix output, with a button at the top of the strip to activate the balanced insert return. The master module hosts a button at the top to switch in the optional Lundahl LL1517 output transformers, where installed.
Only UAD hardware and Apollo interfaces run genuine analog emulations from Neve®, Ampex®, Fairchild®, Manley®, Lexicon®, and dozens more. Because only Universal Audio faithfully and exhaustively models the warmth, grit, and color of the original vintage hardware. Test drive an Apollo interface or UAD Accelerator today. Analog lives here.

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The channel and group strips (the groups are pictured here) are semi-modular. Each module hosts four channels, but beneath the panel there’s a separate card for each channel/group, which should make repair and maintenance fairly efficient, while reducing manufacturing costs significantly.

The monitor control section defaults to reproducing the stereo mix-bus output, but buttons select either of the balanced stereo two-track returns (the first of which can be substituted by a front-panel 3.5mm mini-jack input for a portable media player). There are no facilities to adjust the relative levels of these external inputs.

If any of the console’s solo buttons are pressed, the monitoring automatically switches to reproduce the soloed signal(s) with a ‘solo active’ warning light, and the solo level can be adjusted with a rotary control. Facilities are also provided for switching the monitoring to mono, to adjust the monitoring volume, and to dim (-30dB) or mute the signal. The monitoring output can also be despatched to a set of alternative monitor outputs, but there’s no provision for balancing the relative levels of the main and alt speaker outputs.

The console’s main output metering above the sub-group modules is derived from the monitor signal path, so it shows the levels of whatever is being monitored, including the two-track returns (if selected) and the soloed signals. I noticed that this meter feed is taken after the mono button, which is unusual but potentially useful. The two high-powered headphone output sockets (a 3.5mm mini-jack on the monitor panel, and a quarter-inch socket under the armrest) are wired in parallel, with a shared volume control which is independent of the speaker monitor volume.

Talkback facilities include both an internal electret mic and provision to connect and select an external mic (phantom power is provided). A rotary control sets the talkback level, and buttons despatch the signal to all eight sub-groups and/or all six auxes. Activating talkback to the sub-groups automatically dims the monitors to avoid feedback and aid speech clarity, but not when talking to the auxes. The Trident 88’s line-up oscillator has been omitted.

Overall Impressions

Many of us share a deep-rooted attraction to large mixing consoles that are liberally covered in knobs and buttons, and the Trident 78 doesn’t disappoint — this is a proper old-school analogue multitrack console built to an imposing scale. The construction is solid, elegant, and nicely finished, with a comfortable armrest and smoothly weighted controls that inspire confidence. The overall styling pleasantly reflects Trident’s heydays, with those characteristic straight-sided shiny red, green, and black anodised metallic knobs, back-lit VU meters, and lots of illuminated buttons. The control layout is logical and familiar, and the signal paths are straightforward enough that the learning curve is minimal — which can’t be said of many large-format in-line consoles!

While the feature set is slimmed down compared with that of the Trident 88 series, the Trident 78s are nonetheless a very capable, high-quality, and versatile consoles and the feature-to-cost ratio has been well judged. Even the smallest versions of the Trident 78 can accept a lot of inputs for mix-down — a 16 input, eight mono bus model.
can accommodate 56 mix inputs, and a 32-channel one 88.

Inevitably, there are a few things I’d have preferred to be done differently, although I’m sure not everyone will share my views. For example, it’s a shame that the high-pass filter travels with the full EQ section when switched to the monitor path; most engineers would like to be able to apply high-pass filtering as they’re recording, but to molest the monitor return with the main EQ as the guide mix is built up. There are work-arounds, of course, but I’d prefer the high-pass filter to be part of the input preamp, rather than the main EQ.

Something Paul White commented upon in his Trident 88 review (www.soundonsound.com/reviews/trident-88) also applies here: visually, the monitoring volume control is indistinguishable from the solo, headphone and talkback volume controls; most big console manufacturers employ generously over-sized knobs for the monitor volume, as it’s a control you often need to find in a hurry. (Familiarity and daily use would mitigate this issue to a large extent.)

Of more concern is the evidence of the design’s ongoing development, such as the redeployment of the optional output transformers from the monitoring to the main outputs, and the re-working of the channel direct-output options (which I feel, as discussed above, still need work). Also, being unable easily to route an effects return into the artist cue aux may well be a cost-saving measure too far for many.

There are work-arounds, but I found these foibles rather disappointing.

On a more positive note, when used conservatively the preamps sound neutral, clean and quiet. But they sound musical and appealing, rather than clinical, and deliberately pushing things harder introduces an attractive thickening before you hit the (very high) overload point. So this is a console that can, if you wish, be ‘played’, to develop a sound character. The EQ is also very easy to use, and allows nice tonal shaping without ever becoming muddy or screechy — it’s the attractive behaviour that has long been associated with Trident-console EQs.

While there are obvious advantages to ‘mixing in the box’, mixing on a big analogue console still has its benefits too, and the sheer physicality of it makes it a more enjoyable experience for me. In many ways I think the inherent limitations of ‘manual mixing’ help me focus on the big sound picture, rather than getting lost in minutiae, so I find I arrive at finished mixes quicker. I think they sound better too (although that could be confirmation bias!). There was a time when integrating a large mixer with a DAW was expensive and difficult, but the profusion of audio interfaces with a lot of analogue I/O makes that much easier and than ever before.

The Trident 78 consoles have a lot going for them. Despite some practical limitations and a few foibles, they’ll undoubtedly appeal to those who are attracted to old-school recording consoles and manual mixing, but lack the budget for the equivalent 88-series model.
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Classic Tracks
‘Paid In Full’
(Seven Minutes Of Madness)

Coldcut’s version of Eric B & Rakim’s ‘Paid In Full’ launched their career and reinvented the concept of the remix at the same time.
TOM DOYLE

This year Coldcut — Matt Black and Jonathan More — celebrated the 30th anniversary of their partnership. Over the span of those three decades, they were initially responsible for ground-breaking remixes and their own productions (featuring singers including Lisa Stansfield and Yazz), before they diversified into the development of desktop video, computer games and apps, and launched their highly successful independent label Ninja Tune.

More and Black, originally a pair of part-time DJs, first met in 1986 when the former, having halved his hours as an art teacher, was working three days a week behind the counter at Reckless Records in London’s Soho. Black, whose day job was as a computer programmer, was a regular customer and the pair began chatting and bonding over certain much-coveted records on the funk, soul and jazz scene known as rare groove at the time. But it was the Lessons series of sample collage 12-inch hip hop records made by American duo Double Dee & Steinski, featuring creative cut-ups of the likes of Little Richard, the Supremes and James Brown that first sparked their imaginations.

“We were some of the only people who’d forked out 45 quid a copy to get those records,” remembers Black. “That was because they were really fascinating artifacts representing a new way you could put music together, by cutting it up and making collages out of it. Hip hop was itself putting music together, by cutting it up and reassembling it. We thought, well, we’d love to do something like this.”

So began the journey to More and Black becoming Coldcut and making their name — and having their first hit — with their landmark 1987 ‘Seven Minutes Of Madness’ remix of New York hip hop pairing Eric B & Rakim’s ‘Paid In Full’. It was to be the first remix to enjoy chart success in its own right, reaching number 15 in the UK.

“I think we knew that we were onto a winner with it,” says Matt Black, looking back on their creative and commercial breakthrough.

“We’d cracked something there,” adds Jonathan More. “But we didn’t expect it to blow up in the way that it did though.”

‘Beats + Pieces’

Before joining forces as Coldcut, Jonathan More had been DJ’ing on the London warehouse party scene and hosting the Meltdown Show on pirate radio station Kiss FM. Matt Black meanwhile had already begun experimenting with making Double Dee & Steinski-styled sample collages using the ‘pause button’ method on cassette with a Yamaha four-track.

“I’d been living with a bunch of guys and we were just into getting high and making tapes,” Black remembers. “We’d try to outdo each other doing these pause button edits, to make them more and more complicated. Of course, there was no ‘undo’. In fact you could never be sure that you’d got it right until you went back and listened, and to go back and listen meant you had to stop what you were doing, which meant that you could then get an error when you went back to where you were. So it was a pretty risky process. That sort of thing did provide a challenge to get right.

“I was looking for a partner to work with, so meeting Jonathan who was really on the same track as me, it was like, OK, let’s work together and put this out as Coldcut.”

The first product of their collective labour was ‘Say Kids What Time Is It?’, a 12-inch white label single released in January 1987, cutting together James Brown’s ‘Funky Drummer’, Ennio Morricone, Sly Stone and even ‘I Wanna Be Like You (The Monkey Song)’ from The Jungle Book. The track received much attention on the UK underground scene as the first British breaks record. “That was recorded on cassette,” says Black.

“Just pause-button-edited together and then put in the Yamaha four-track for another couple of tracks of scratching over the top.”

For the more elaborate follow-up, ‘Beats + Pieces’ (featuring everyone from Led Zeppelin to Kurtis Blow), Coldcut used a professional recording studio for the first time in their career. “With that we were quite lucky,” says More. “Kiss FM had been given some studio time in exchange for some advertising. The boss Gordon Mac didn’t really know what to do with it, and he said, ‘Well, do you guys want to go and use it?’ So we had a day in a studio in Camden.”

“It had a two-inch 24-track machine,” says Black. “I remember working out on graph paper how to try and allocate the tracks.”

“We’d done a fair few kind of rehearsals and versions of it at Matt’s place,” says More. “I digitised all the cassette sketches a while ago and there’s quite a load of ideas that we didn’t use.”

Working with engineer Raine Shire on ‘Beats + Pieces’ opened Coldcut up to the wider possibilities of the recording studio. “Raine introduced us to the concept of the tape loop,” says Black. “I’d been trying to edit four minutes of the backing track together on cassette, from only a one-bar loop of Led Zeppelin’s ‘When The Levee Breaks’. Raine showed us how to do this tape loop which literally involved getting the break off vinyl, recording it on to tape, copying that multiple times and then editing that together using a reel-to-reel editing technique, which of course is a lot more precise. That became a physical loop of tape which ran off the half-inch machine and around a broom handle which we sort of strapped to a desk.”

Along with the various samples and scratches added, the pair used turntable stabs from rock records by Ted Nugent, Yes and Grand Funk Railroad, adding delay to them for further enhancement. “There’s no delay in Double Dee & Steinski or Grandmaster Flash’s records,” Black points out. “But there were certain records that did use it, like Steady B’s ‘Bring The Beat Back’, which is a dub hip hop record with these echoed stabs. I remember that was pretty influential.

‘Run-DMC had made a record called ‘Rock Box’ [in 1984], a sort of hip hop-rock fusion. I didn’t actually like the
track that much but it was like, Yeah, mixing rock and hip hop, that's cool. So I spent a weekend going through all my mate's heavy metal and rock records, which I didn't like, but just trying to find those good scratchable sounds. We took our own delay down there to put the echo on the stabs. This cheap Maxim digital delay which was really shit quality, but we loved it.”

‘Paid In Full’
At the same time, Jonathan More was a big fan of go-go, the Washington DC syncopated funk sound that was popular in the mid-'80s and championed in the UK by Island Records offshoot Fourth & Broadway's A&R man Julian Palmer. “I actually flew to Washington DC, to buy go-go records, like an idiot,” laughs More. “I was seriously into it. And I think I probably pestered Fourth & Broadway to get free copies of various go-go records that they were putting out at the time.”

Palmer was also aware of the cut-up records that Coldcut were releasing. As a result, the A&R man approached the duo — without telling his bosses at the label — to remix Eric B & Rakim’s ‘Paid In Full’ from their 1987 debut album of the same name. “He’d got his ear to the ground,” Black says of Palmer. “He knew we’d been putting out these underground releases and he thought, ‘There's something there. I don’t think we were his first choice. I think we were the third choice and the other people he didn’t get to do it. But we jumped at the chance.’”

Since this was a furtive, exploratory commission, however, Palmer wasn’t able to supply Coldcut with the master tape of the track. “He gave us, like, 10 copies of the album,” More notes with a chuckle. “Literally he gave us vinyl copies of the album and that was it,” says Black. “We just took elements. Like ‘Beats + Pieces’, the ‘Paid In Full’ remix didn’t even start in a studio. It was a small corridor in my flat. I had a couple of decks and the four track and we would try things out.”

One of the key elements that was to make Coldcut’s ‘Paid In Full’ remix stand out was their use of a vocal sample of Israeli singer Ofra Haza’s ‘Im Nin’alu’ from her 1984 album Yemenite Songs. “I’d been turned onto it by Charlie Gillett who was a Capital Radio DJ,” says More. “He was sort of the John Peel of world music. Fantastic DJ, lovely man, sadly passed now. I guested on his show on Capital and he played that record and I was like, ‘I’m gonna have that. DJ’ing out, ‘slurping’ was one of the many crazy expressions we had to describe to people what we were doing.”

“Slurping was sort of you’re mixing it in but you’re not too worried about whether it’s in time or not because maybe it hasn’t got a beat,” Black explains. “So we slurped it in and it was like, ‘Yeah this could work, but it doesn’t sound quite in tune. OK if we pitch it down...’ And it was minus eight on the turntable, which is as far down as you can go. But at minus eight it was perfectly in tune, and in time effectively as well, by good coincidence. That turned out to be the hook.”

Elsewhere the elements that the pair took from the original ‘Paid In Full’ included the scratching of the line "This stuff is really fresh" from Fab 5 Freddy’s ‘Change The Beat’, the distinctive rolling bass line from Dennis Edwards and Siedah Garrett’s 1984 single ‘Don’t Look Any Further’ and the beat from the Soul Searchers’ ‘Ashley’s Roachclip’ from the 1974 album, Salt Of The Earth.

For their remix, Coldcut wanted to employ the same method as ‘Beats + Pieces’ and so requested two days in the Island Records studio to work again with engineer Raine Shine. “We actually had a sampler by then,” says Black. “We’d got a Casio RZ-1 drum machine, with 0.8 seconds sampling time, which was just enough to get four drum hits. But we wanted to do something with better quality, so the next step was to use a Bel delay as a sampler. Again it was Raine who showed us that you could take a loop into it and then edit the start and end point.”

“What was weird was it seemed to enhance the bass of the kick drum. The Bel unexpectedly added that to it. So then we had a solid looping drum break, which we laid down for several minutes. That became the backbone of what we could lay stuff on top of. We were able to cut in bits of the vocal on top of that.”

“This was our first use of an SSL desk as well, so you could do automation and write into it when you wanted which...”
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tracks to cut in and out. That was a way
to make the mix more intricate and to
automate the structure of it more.”

For the more complex scratching that
Coldcut wanted to hear on ‘Paid In Full’
they brought in London hip hop crew
Bass Inc’s DJ Cell. “He was a wicked
talent,” says Black. “By far one of the best
scratchers that I’d come across. So that
was a good extra element to it.”

Among the other tracks Coldcut
sampled for ‘Paid In Full’ were James
Brown’s ‘Hot Pants’, the Peech Boys’
‘Don’t Make Me Wait’, the Salsoul
Orchestra’s ‘Ooh I Love It’ and Long Island
rap combo Original Concept’s ‘Pump That
Bass’. At the same time, Coldcut pulled
out another Eric B & Rakim track from the
Paid In Full album, ‘I Know You Got Soul’,
for the key “pump up the volume” sample.

Non-hip hop spoken-word samples,
such as Humphrey Bogart from the film
The Big Sleep (“Wait a minute, you better
talk to my mother”) and the standout
“This is a journey into sound” introduction
(by actor Geoffrey Sumner) were spun into
the track using an ad-hoc technique the
pair had first used on Kiss FM.

“We would mix spoken word and
jingles,” says More. “They didn’t have
a radio-style jingle cart machine as they
were called in those days. Couldn’t afford
it. So we used to get computer cassette
tapes, which were like 15 minutes long,
or something, the shortest cassette tapes.
You were able to actually unscrew them,
unlike a lot of other cassette tapes. So
using the same pause button technique,
we’d record a phrase onto those
cassettes, unscrew them, cut the leader
tape off, fit the tape back in again, and
get it so that it was spot on, so as soon as
we pressed the cassette button, it
would play the phrase.

“I’d found a record called
A Journey Into Stereophonic
Sound and actually the music
on it’s really boring. But there’s
this guy introducing it. I took it
home and put it on and that was
the first thing that came up on it
and it was like, Yes! It’s like gold
mining basically, y’know. You
sift through it. Most of it’s crap,
but then every so often you find
a nugget. We used that as an
introduction to the record. It’s
a good signature sample.”

One of the more surprising
and unexpected samples on ‘Paid
In Full’ is actually from the BBC
Records album, Bang On A Drum,
featuring songs from the kids’
TV shows Play School and Play
Away. “All the hipsters were into

The Roland MC-202 that gave so
much trouble during the making
of Doctorin The House.”
these old James Brown and Dennis Coffey records and so on," Black points out, still amused. "But you see a record called Bang On A Drum and you’re gonna buy that because you never know, there might just be something."

“We put that in pretty much at the end of the mix and purposely didn’t allow any of it to kind of leak out,” says More. “It’s faded so that nobody else could take it [laughs]."

**Plastering Over The Cracks**

Even with an SSL desk at their disposal, the creation of ‘Paid In Full’ was still very much a process of trial and error. “There’s a sort of dodgy edit [at 3.13],” says Black. “We spent ages on it, but you can still hear it. We didn’t know how to get out of it, so we just sort of brutally cut in and put some delay on it and hoped that no one would notice. Sometimes when there were the joins and things, you sort of had to paint over them. One technique was to put some echo on, or another technique was to have some kind of exciting sound just before, so you get distracted by that and you don’t notice the backing.”

Although a record comprised entirely of samples, ‘Paid In Full’ was actually quite a tricky track to mix. “The complicated thing,” says More, “is that all of those records we’d sampled contained their own reverb. So the issue is that if you want to start adding reverb, which is a traditional kind of mixing element, you can very quickly make something that’s sample-based sound incredibly muddy and shit. “Raine was a great engineer with a great ear. She was very, very careful about how to balance that record. Really it was a question of EQ’ing it so that it sounded really big and tough with judicious use of reverb and delay. I think there’s probably a tiny bit of reverb on the scratching to sort of seat it in. But actually for a traditionally trained engineer, it was probably quite a difficult thing to mix, ‘cause it’s outside of their comfort zone.”

The success of the ‘Paid In Full (Seven Minutes Of Madness)’ remix was a huge profile-lifting boost for Coldcut upon its release, which was a fitting reward for the fact that they were only given £750 for it at the time. “It sold several million copies, I believe,” says Black. “Later we did a version called ’Not Paid Enough’...”

“But we got full credit,” adds More, “which actually I think was unusual. We didn’t realise that. It seemed normal, but I think probably we were only the second or third remix artist to get a proper equal billing.”

“[The record label] could do that ‘cause it didn’t cost them any money and we were hip, so actually they gained from that,” says Black. “But it did get us on the front cover of the NME which was a pretty decent result, and so it was a big moment in our career.”

Apparently though, Eric B & Rakim, who suddenly found themselves with a surprise UK hit, had mixed reactions to the track: Rakim loved it but Eric B dismissed it as “girly disco music”. “Some people say that Eric said that about a different mix that was done of it at the time,” stresses More. “But they came over to do Top Of The Pops for it.”

“Imagine, they don’t know anything
about this record,” says Black. “It’s not their single from the album, they don’t know that anything’s been done. Suddenly they get a phone call from the record company: ‘Guys, your record ‘Paid In Full’ is a hit in the UK, you’ve got to fly over and be on Top Of The Pops.’ They come over here and they get shoved onto Top Of The Pops, which was lots of girlies dancing around handbags.”

“It’s par for the course, as a hip hop artist though,” laughs More. “Having a good old whinge.”

‘Doctorin’ The House’

1988 was the year that Coldcut broke through as hit-making producers in their own right, with ‘Doctorin’ The House’, featuring London-born singer Yazz, which reached number six, before the pair produced her number one dance cover version of Otis Clay’s ‘The Only Way Is Up’. “We started doing our own programming and then used samples on top of that,” says Black, “so that took it to another level. For ‘Doctorin’ The House’, the drums were programmed on the RZ-1. The sounds weren’t that great though, and in the studio our engineer did manage to sort of beef them up a bit. The bass line on that is a Roland MC-202, and I hadn’t got the manual for it. It took me days of fucking around to get it to loop, ‘cause you had to get the notes to add up to 128 and I didn’t realise this.

“That record was very much us trying to emulate these Chicago records where it was like, ‘That’s just a drum machine and a sequencer and a sample on top. We could do that.’ For years our sampling workhorse was the Casio FZ-1 and then as the sequencer we used an Atari with C-Lab Creator. That was where my background in computing became useful because I wasn’t intimidated. We were able to start using that as our main engine for putting stuff together. It was the sequencer, the sampler, combined with a turntable as our sources. That was how we went forward.”

Coldcut, however, were interested in more than simply music. In 1988, they formed Hex with Miles Visman and Rob Pepperell, a ‘multi-media pop group’ using Amiga computers to create the Top Banana video game and the Global Chaos CDTV CD-ROM featuring techno tracks and ‘rave graphics’. “I think we realised that there was a fruitful relationship between making music in a cut-up hip hop sampling style and then applying that to visuals,” says Black. “We realised we could do computer games, we could do desktop video. All these things could feed into a sort of combined art form, and so that was kind of the bridge into the next phase of our career.”

Having been signed to Big Life Records and then Arista, Coldcut soon tired of trying to churn out the hits. This led in 1990 to the launch of their independent label Ninja Tune, with them producing DJ Food’s series of Jazz Breaks albums (which helped to shape the trip hop and down-tempo genres) and signing the likes of the Cinematic Orchestra, Amon Tobin and Bonobo.

More recently, they’ve moved into apps, creating Ninja Jamm, which allows users to remix elements or ‘Tunepacks’ by Coldcut and other Ninja Tune artists. The program’s features include the Coldcutter, which chops any sample into ‘cuts’ and offers various ways of manipulating them. “Ninja Jamm is a culmination of more than 20 years work actually,” says Black. “We can trace it back to Top Banana, which had this ability to randomly chop stuff up and make these drum fills. That became the Coldcutter later on. I’ll say...”
one thing — we haven’t made any money on the software, but it’s our R&D side, and we use it to perform, and we’re constantly updating it. I think the next version will be more of a fully featured music-making tool.”

**Future Indefinite**

In terms of their own music, this year the pair put out their latest release, *Outside The Echo Chamber*, a collaboration with Adrian Sherwood under the name Coldcut x On-U Sound. These days More and Black are very much aficionados of Ableton Live. “It’s the speed of being able to do ideas really more than anything,” says More. “That forms the basis of a creative part of any track. But subsequent to that, those tracks can come out of Ableton and go into lots of other different programs, have work done on them, come back into Ableton, go out again. So I do use Logic as well because some of the people we were working with use it. Sometimes I’ll mix stuff in Logic, ‘cause it’s got things that Ableton doesn’t necessarily have.”

“We used Cubase as well for a few years,” adds Black. “But with Ableton, you’ve got the session layout and then the arrange layout, so you’re able to jam in the session and then record that into the arrangement, then edit that later. In a way that’s what’s great about modern electronic music is that you can freak out, but then rather than all that being lost you can then take what you’ve recorded multi-track and refine it in an editing process.”

Coldcut recently developed MidiVolve for Ableton, a Max For Live arpeggiator, riff generator and sequencer. “Over the years we’ve designed various tools and toys, visual and audio, and they’ve not really been proper products,” says Black. “I’ve been a bit frustrated sometimes that I think actually these ideas are really good but they don’t fit into any existing box. So working with Ableton is a chance to work with one of the top companies.

“The MidiVolve product is an idea which is built still on the Coldcutter: taking something, chopping it into pieces and putting it back together. Again, it’s cut and paste on a kind of micro level. It’s been quite successful. It’s the best-selling Max For Live patch this year actually. It’s become our first software product to actually make a profit.”

Thirty years on, Coldcut continue to push forward and innovate. But does it really feel like they’ve been working together for three decades?

“I dunno, it’s one of those strange things that it seems only momentary,” says More. “It seems like it’s just started.”

“Time flies when you’re having fun,” Black laughs. “We look back and we can say ‘Well, we could’ve done more, we could’ve sold millions.’ But if we had made millions, we would’ve mansarded out, which is our phrase for when you make a lot of money but you probably end up stuck in your mansion and don’t make anything interesting again.

“We’ve been very lucky to just be able to make enough money to keep doing what we do and keep pushing on with it. Building up a music tribe has been the key to the success really. Y’know, with advances in longevity, technology, we could be good for another 30 years, or another 300 years. We’ll have to see. But we’re still really into it.”
DigiLink connectors. This flexibility has helped to keep those devices in Lynx’s catalogue for over a decade, ensuring that users weren’t stranded when, for example, FireWire was superseded by Thunderbolt.

Now, however, the Aurora 8 and 16 have been discontinued in favour of a product which takes this flexibility to the next level. In terms of interfacing, the Aurora (n) is based around the same LSlot technology as its predecessors;
so, depending on your needs, it can be a native audio interface connecting to a Mac or Windows PC over Thunderbolt or USB, or a stand-alone A-D/D-A converter that hooks into a recording rig over DigiLink, AES or Dante (the now-discontinued FireWire and MADI cards are physically compatible, but not supported in firmware). But unlike its antecedents, the Aurora (n) is also fully modular in terms of audio I/O, making it much more adaptable to the specific needs of individual studios. What’s more, it also feature built-in recording to micro SD cards.

In effect, the Aurora (n) core unit is a 1U rackmounting chassis with built-in power supply, display, controls and headphone outputs. A single chassis can host a total of six modules. One of these must be an LSlot card, and another one is always occupied by Lynx’s word-clock module, which provides a single input and three outputs on BNC connectors. That leaves four slots for audio I/O modules.

The only such module available at launch was the AIO8E, which provides eight line-level analogue inputs and outputs on a pair of DB25 connectors, but others are forthcoming. By the time you read this, the LM-DIG digital module should be available, offering 16 channels of AES3 digital I/O plus expansion ports for another 16 channels in either AES3 or ADAT format (and thus, among other things, enabling the Aurora (n) to be used with Lynx’s PCIe interface). Also imminent is a module that will provide four ‘combi’ XLR/jack sockets offering high-quality mic preamps and instrument inputs as well as line-level performance identical to that of the AIO8E, and an intriguing summing mixer module which piggybacks on the line-level I/O.

Keen mathematicians will no doubt already have noticed one area in which the Aurora (n) improves over earlier Aurora: provided you stick to line-level I/O, a single unit can host up to 32 analogue inputs and outputs, as against 16 or eight in previous models.

### **PROS**

- Excellent audio quality.
- Flexible, future-proof modular design with up to 32 analogue inputs and outputs per unit.
- Includes high-quality dual headphone amps and three word-clock outputs.
- The ability to record to SD cards provides a useful level of redundancy in case things go wrong.

### **CONS**

- Software control utility is currently something of a stopgap solution, which leaves many parameters adjustable only from the front panel.
- USB operation limited to 16 channels.
- No Thunderbolt cable supplied.
- Some aspects of its operation are not documented at present.

### **SUMMARY**

A comprehensive update to Lynx’s modular interface doubles the potential I/O count, improves audio specs still further and adds a very useful redundant recording feature.
Latency & The Aurora (n)

We’ve come to expect superb low-latency performance from Thunderbolt audio interfaces, so it seems a trifle churlish of me to report that the Lynx Aurora (n) is merely excellent. On Mac OS, the lowest buffer size available for all the Thunderbolt interfaces I have tested so far is 32 samples. At a 44.1kHz sample rate, that typically translates into a round-trip latency of around 4ms. With the Aurora (n), both Logic and Reaper reported that the total round-trip latency was 5.1ms, of which 3.3ms was on the input side. In both cases, a re-recorded click actually appeared a few tens of samples earlier on the timeline, suggesting that the actual figure is a little lower. Assuming you can run the Aurora (n) at this buffer size, though, that should be plenty low enough for most real-time monitoring needs, and is better than the vast majority of USB interfaces. As the LT-USB LSlot card wasn’t supplied, I wasn’t able to test the Aurora (n)’s performance over USB.

versions. That is, unless you choose to connect it over USB, Lynx have chosen to implement USB 2 rather than USB 3 connectivity, and the resulting bandwidth restrictions mean that only 16 channels of I/O are supported in this mode. Existing Aurora users might also note that the Aurora (n) retains a limitation of the original, in that only one LSlot card can be installed at a time. To convert an Aurora (n) from, say, Pro Tools HDX to Thunderbolt operation would require swapping cards out, though I’m told there are plans for a Thunderbolt 3 card which may be able to coexist with the HDX or Dante options.

The review unit was fitted with the Thunderbolt LSlot card and two AI0BIE modules, for a total of 16 analogue ins and outs. The Thunderbolt card has two connectors, allowing other Thunderbolt devices to be daisy-chained, but I was disappointed to find that a Thunderbolt cable is not included.

Back To Base

The Aurora (n) is intended to act not only as a native audio interface, but also as a stand-alone converter and Pro Tools HDX expander. Lynx have thus provided fairly comprehensive front-panel control over many of its parameters, as it can’t be assumed that a computer will be available to control them. In its present form, though, any Aurora (n) parameter that can be adjusted from the front panel can’t also be manipulated from the computer, and vice versa, which might prove inconvenient if you want to keep the Aurora (n) in a separate machine room. Most of the things that are controllable from the front panel are housekeeping functions you wouldn’t need to change very often, but not all. In particular, recording to and playback from the in-built SD card reader is possible only by pressing the buttons on the unit itself. I’m told this situation will not be permanent, and that forthcoming versions of the firmware and software will enable remote control from the computer.

Central to the Aurora (n)’s front panel is a high-resolution colour screen. In normal use, this serves as a very effective metering display, showing either small vertical ladders for a comprehensive selection of inputs and outputs, or a more detailed horizontal stereo view focusing on two of each. To the left of the display are Record and Play buttons for the built-in SD card recording, plus up and down arrows which are used mainly for navigating between takes. Configuration of the Aurora (n) itself is handled using four more buttons and a rotary encoder to the right of the screen. Hitting the Function button or pressing the rotary in brings up a menu that offers various configuration options. These include the ability to switch analogue inputs and outputs between +4dBu and -10dBV standards in banks of four, to send test tones to any of the analogue outputs, and a Routing submenu which determines whether the analogue outputs are fed from the Thunderbolt playback, SD card playback, direct from the analogue ins or any combination of the three.

At the extreme right of the front panel are two headphone sockets. Both are fed from the same stereo bus, which can pick up any pair of channels from either the analogue inputs, the Thunderbolt playback channels or the SD card playback. Each has its own level control, which operates in the analogue domain. Lynx describe the headphone amps as being “audiophile-grade”, and they do indeed sound noticeably better than your average audio interface headphone socket. Not only that, but they also go very loud indeed, which can’t be said of most headphone amps built into interfaces.

We’ve come to expect superb low-latency performance from Thunderbolt audio interfaces, so it seems a trifle churlish of me to report that the Lynx Aurora (n) is merely excellent. On Mac OS, the lowest buffer size available for all the Thunderbolt interfaces I have tested so far is 32 samples. At a 44.1kHz sample rate, that typically translates into a round-trip latency of around 4ms. With the Aurora (n), both Logic and Reaper reported that the total round-trip latency was 5.1ms, of which 3.3ms was on the input side. In both cases, a re-recorded click actually appeared a few tens of samples earlier on the timeline, suggesting that the actual figure is a little lower. Assuming you can run the Aurora (n) at this buffer size, though, that should be plenty low enough for most real-time monitoring needs, and is better than the vast majority of USB interfaces. As the LT-USB LSlot card wasn’t supplied, I wasn’t able to test the Aurora (n)’s performance over USB.

versions. That is, unless you choose to connect it over USB, Lynx have chosen to implement USB 2 rather than USB 3 connectivity, and the resulting bandwidth restrictions mean that only 16 channels of I/O are supported in this mode. Existing Aurora users might also note that the Aurora (n) retains a limitation of the original, in that only one LSlot card can be installed at a time. To convert an Aurora (n) from, say, Pro Tools HDX to Thunderbolt operation would require swapping cards out, though I’m told there are plans for a Thunderbolt 3 card which may be able to coexist with the HDX or Dante options.

The review unit was fitted with the Thunderbolt LSlot card and two AI0BIE modules, for a total of 16 analogue ins and outs. The Thunderbolt card has two connectors, allowing other Thunderbolt devices to be daisy-chained, but I was disappointed to find that a Thunderbolt cable is not included.

Back To Base

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Most high-end audio interfaces include some sort of provision for monitor control, even if that often amounts to little more than a volume control and a mute button. This has obvious benefits, but usually falls well short of meeting the full ‘master section’ requirements of any serious studio. Perhaps with that in mind, Lynx have included no monitor-control features at all. Apart from the headphone sockets, there are also no full-sized audio outputs, so it’s pretty clear that the Aurora (n) is designed to be used as part of a larger ecosystem rather than on its own. This won’t generally be a problem for people integrating it into a studio environment, but those who do a lot of location recording might prefer something more serious than what the Aurora (n) offers. While Lynx have included no monitor-control, it’s still possible to control the outputs, so it’s pretty clear that the Aurora (n) documentation, but on asking Lynx for clarification, it turns out that DA W software doesn’t address the physical outputs directly. Instead, it feeds these Play channels, which, in turn, can be routed to physical outputs.

The Mixing Lynx

Those who do use the Aurora (n) as a computer interface can employ the Lynx Mixer utility to route inputs to outputs so that they can be directly monitored without passing through the computer’s mixer. The current version also looks fairly primitive from a graphical point of view, and, as I’ve already mentioned, lacks the ability to remotely control many of the Aurora (n)’s settings. Thankfully, all this should change pretty soon. A completely new mixer and control utility is in the works, and is expected to be made available early next year.

The Mighty Micro

One of the most interesting new features in the Aurora (n) is the ability to record to a micro SD card, independently of an attached computer. This opens up at least two roles that weren’t possible with earlier Auroras, or indeed with most other audio interfaces. One is that you could leave the machine recording all day to provide a redundant backup in case of hard drive failures or other disasters afflicting your studio computer, and a chance to recover those magic moments that simply weren’t captured because the DAW wasn’t in Record at the time. Another would be to use the Aurora (n) as a computer-free option for live concert recording and virtual soundchecks; at present it’s not possible to synchronise SD card recording across more than one Aurora, but Lynx told me that they are investigating the possibilities on this front.

What the SD card reader doesn’t do is turn the Aurora (n) into a fully fledged studio recorder. There’s no individual track arming, punch-ins, editing or any of that malarkey. In fact, there are no individual tracks, as such: each recording you make to the SD card consists of a single WAV file containing either two, four, eight, 16 or 32 adjacent channels, starting at your choice of analogue inputs 1, 9, 17 or 25, or Thunderbolt playback channels 1, 9, 17, and 25. So, for example, you could set it to record analogue inputs 9-24 as a single 16-channel WAV file, but you can’t pick and choose individual channels to include or exclude.

These WAV files are referred to as Takes, and they can be organised into Sessions. Both can be named, and Sessions can be hidden, but what you can’t do is delete files on the SD card. Were data to be distributed in random fashion across the card, as would happen if you repeatedly deleted and then recorded files, the card reader wouldn’t be able to keep up with the necessary streaming data rate. The suggested approach is to dump all files from the SD card to a computer as needed and then reformat it before making further recordings. It’s also, apparently, possible to create projects on a computer and transfer them to the SD card for playback, though I was unable to locate the promised video tutorial that explains how to do this. Said video should also explain how to manipulate the massively multi-channel WAV files that are created by the Aurora (n) to make them palatable to DA W software — if not then you could always download the file splitter utility that Lynx’s rivals RME make available!

In practice, since recording currently needs to be set up from the front panel, I doubt you’d want to get involved in complex session naming and so forth on a regular basis, but its simplicity itself to wipe the SD card and hit Record at the start of a day’s work, and the peace of mind this brings will be very valuable in a studio environment.
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Its outstanding features tailored to the requirements of game audio, TV and film post-production make Nuendo the leading solution for audio-to-picture work. Alongside unique tools like Game Audio Connect 2 for connecting directly to Audiokinetic’s Wwise, Nuendo 8 provides a powerful Direct Offline Processing system to process audio clips quickly and easily. Also new to the table are the new Sound Randomizer for automatic creation of sound variations plus comprehensive possibilities for surround and immersive sound mixing, including Dolby Atmos. Together with its integrated dialog recording system (ADR) and high-performance automation capabilities, Nuendo is at the helm of providing dedicated technology to audio post-production.
Lynx Aurora (n) layout

"If you walked into a studio and found this at its heart, you’d feel confident that the people running it know what they’re doing!"

Points Of Comparison

Probably the most direct competition for the Aurora (n) comes from Apogee’s Symphony I/O MkII, another modular system that can act either as a native interface or as a converter for Avid HDX systems. Prices vary with configuration, but a Symphony system with Thunderbolt connectivity and 16 line-level analogue inputs and outputs is directly comparable to its Aurora (n) equivalent, cost-wise. Both are extremely capable and desirable systems offering superb sound quality, and unless you need a feature that only one of them supports, choosing between them would be a tough call.

Of the two, I’d say the Symphony offers the slicker user experience, thanks to its larger front panel with colour touchscreen, and it also provides basic monitor control. Expansion options include a very fine mic preamp module that is available now rather than at some point in the future, and further down the line, Apogee are also promising an option card for connecting to Waves/Digico DigiGrid systems. In the Aurora (n)’s favour are its USB and Windows support, the option to connect to a Dante network, and, of course, the ability to record to and play back from SD cards. This last in particular will be a pretty big plus point in many people’s books.

A different approach to modular interface design is reflected in Focusrite’s Red range. Each of these units offers a fixed quotient of analogue and digital I/O, but the appropriate connectors for Thunderbolt, HDX and Dante operation are all built in; so although you can’t add extra cards to the base unit to increase the I/O count, you can easily plug in an additional Dante expander. The new Red 16Line, for example, offers 16 line-level inputs and outputs, along with all above mentioned interfacing options and basic monitor control, and is expected to be somewhat less costly than the equivalent Aurora (n); but, like the Symphony, it lacks the Aurora’s in-built recording features. The Aurora (n) also scores over all of its competitors I can think of in providing three word-clock outputs, giving it the potential to act as a high-quality master clock in all sorts of setups.

Less obvious competitors, meanwhile, might include Antelope Audio’s more affordable Orion 32+, and JoeCo’s BlueBox Workstation Recorder, which is available in several configurations including one with 24 line-level ins and outs. This has rather more fully developed recording features, but considered as an interface, is a USB-only device, with no Thunderbolt or HDX options.

The Aurora (n)’s SD card slot and headphone outputs are helpfully positioned on the front panel.

In Use

The case that houses the Aurora (n) is notable for having lots of air vents in the top panel. Lynx say that it is designed to be passively cooled, and recommend leaving empty the rack space immediately above and below it. Having inadvertently disregarded that advice when I first used the Aurora (n), I’d say it really is essential to do so, because otherwise it becomes alarmingly hot. Once the unit was appropriately racked, though, I had no problems with heat, and in general, my time with the Aurora (n) was pretty much plain sailing. As I’ve described above, Lynx’s software mixing utility hasn’t yet had an overhaul to match that of the hardware; an improved version is clearly a necessity in the mid-term, but in normal use, the current version is serviceable if not exactly slick.

In fact, it’s perhaps testament to the strength of Lynx’s modular approach that the one problem I did run into in normal use was confined to the user interface and didn’t seem to affect the Aurora’s recording functions. On this occasion the screen and front-panel controls froze, but both Pro Tools and the Lynx Mixer continued as if nothing had happened, and audio streaming was unaffected. Other than that, audio came and went exactly as it was supposed to throughout the review period.

Lynx have forged a stellar reputation for the sound quality of their interfaces, and the Aurora (n)’s converters are said to improve on those of the already-very-good Aurora 8 and 16. I’ve used the Aurora 16 in the past, and if I didn’t hear enough of a difference to justify upgrading on reasons of audio quality alone, that’s more a reflection on how good the original Aurora series is than it is a criticism of the new design. Both specs-wise and subjectively, the Aurora (n) very much holds its own against similarly priced rival interfaces such as the Apogee Symphony I/O MkII and Focusrite Red range. For me, the audio quality of all these devices is so good as to be a non-issue, and I take my hat off to anyone whose ears are golden enough to establish a clear preference between them. You’d certainly need to audition them side-by-side in a very carefully controlled test in order to do so.

The word ‘professional’ is much abused, but if you take it to mean that the design of a product is led above all by engineering considerations, then the Lynx Aurora (n) is very much a professional product. There are no gimmicks here, or spurious bells and whistles, but you can be confident that it’ll serve you well for many years to come, and that it will adapt to changing needs over time. And if you walked into a studio and found this at its heart, you’d feel confident that the people running it know what they’re doing!
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Audionamix
ADX Trax Pro 3 SP

Speech & Melody Separation Software For Mac OS

Mixing audio can be difficult, but unmixing is impossible — unless, that is, you have access to Audionamix’s innovative ADX technology.

Founded in 2008 and now based in Paris and Los Angeles, Audionamix have built a global reputation for their audio separation and isolation software. The company’s proprietary ADX technology and its expertise in isolating the speech, vocal and other monophonic melodic elements from a full mix has not only allowed engineers, producers and artists to craft new arrangements, but also has given television and film companies the ability to enhance older mono and stereo soundtracks both by adding new music and sound effects to old dialogue and by creating 5.1 surround mixes.

In early 2014, Audionamix released the Mac-only ADX Trax, the world’s first non-destructive, automated melodic audio source separation software. By retaining the integrity of the original source, ADX Trax allowed repeated refinements of the initial separation to achieve the best possible results, and in late 2014 Trax Pro appeared, bringing with it non-destructive spectral editing. June 2016 saw the release of Trax 3 and Trax Pro 3, both of which offered faster separation processing speeds, a consonant annotation tool and a pan-specific editing feature that enabled direct editing of audio content in user-defined areas of the stereo field. In 2017 a new, speech-optimised algorithm that automatically detects and separates speech from background elements — musical or otherwise — was added to both Trax 3 and Trax Pro 3 to create what Audionamix say is the first fully featured, automated speech separation software: ADX Trax 3 SP and ADX Trax Pro 3 SP.

Making Trax
The Trax Pro 3 SP user interface, like that of Trax Pro 3, has three screens named Separate, Process and Spectral. The Separate screen handles the separation process.
of the target speech or melodic source from the background elements of a stereo or mono audio file. Extraction target options include Speech and Melodic Vocals, Speech Only, Melodic Vocals Only or General, which extracts all melodic content in the file. Initially, the Trax Pro 3 SP software analyses the audio file and encodes specific information from it. It then uploads the result to Audionamix’s cloud-based servers, where an automatic separation is performed. The separated files are then downloaded and decoded to produce a Vocal track (the extracted target) and a Music track (everything else).

The Separate screen displays that which Audionamix refer to as a Pitchogram — a frequency-restricted (60Hz-1.1kHz) spectral view optimised for displaying both speech and melodic vocal — which shows the material that has been automatically identified as the target content. A highlight blue line called the Pitch Guide runs across the screen, marking the fundamental frequencies of the melodic content that it has targeted for extraction. Although only the fundamental frequencies are marked, the software actually extracts all the harmonics of these frequencies that it is able to identify. In addition, if automatic marking of consonants was specified, the pitchogram displays vertical green lines marking those areas where the ADX software has detected what it believes to be consonants. Also at this point, the Process screen carries a waveform display of the automatic extraction that can be switched between Vocal and Music, whilst the Spectral screen displays a spectrogram that can be similarly switched between the two extractions.

Both the automatic Vocal and Music extractions can be auditioned on the Separate screen, and they can be soloed, muted and adjusted in level. For some uses, this first draft may be good enough, in which case you can simply proceed to the next step, which takes place in the Process screen. However, the ADX algorithms are not quite as discriminating as the human ear, and areas of the Pitch Guide may indicate erroneous identification of the target’s frequencies.

To improve the separations, Trax Pro 3 offers several ways of editing the automatically generated Pitch Guide. Perhaps the most useful of these is the Eraser tool, which allows you to remove areas of the Pitch Guide or consonant markers where there is no relevant content. The Pitch Guide can be drawn freehand using the Pencil tool, while the Pitch Magnet function can be used to identify and redraw the most likely line in a section of the Pitch Guide. The Consonants tool allows you to insert or move consonant markers where required. The Pitchogram can be zoomed in and out in the horizontal and vertical dimensions and a useful Guide Tone function not only allows you to hear the pitch of the line being drawn by the pencil tool, but also to check a Pitch in the track against a virtual, vertically oriented piano keyboard. In addition, if you’re trying to extract a melodic track, and you have or can create a MIDI file of the melody, you can load that as the Pitch Guide, provided that its length matches that of the file being processed.

Once the Pitch Guide has been edited, the result is uploaded to the Audionamix cloud servers and a ‘Refined’ automatic separation, based on the revised Pitch Guide, is returned. You can carry on refining this refined separation for as long as you need. A Marquee tool allows you to upload a portion of the Pitchogram, rather than the entire file, to the cloud for processing, speeding up upload, processing and download times.

**AUDIONAMIX ADX TRAX PRO 3 SP $999**

**PROS**

• Automatically separates vocals and melodic content from tracks in a way that can be easily refined.
• Non-destructive separation, extraction and editing.
• Powerful and effective editing tools.
• Capable of producing excellent results.

**CONS**

• Not cheap — but if you need what it does, there may be no alternative!

**SUMMARY**

ADX-Trax Pro 3 SP is an innovative, powerful and extremely effective program that can automatically target, identify, separate and extract the speech, vocal and monophonic melodic components of a full stereo or mono mix. Its extensive, non-destructive editing capabilities make it possible to refine an automatic separation to achieve extremely impressive results.

Once generated, the Pitch Guide can be manually edited.

**Due Process**

Once you are satisfied with the revised separation, the action moves to the Process screen, where a new separation, based on the refined separation’s Pitch Guide, can be created. This process can be run in HQ (high quality), with or without boosted consonants, and can also include any short or long reverberation associated with the target. The process can also be run only on a user-defined section of the stereo field, which is useful if the target being extracted is panned off-centre. The resulting separation can then be further refined by the Post-Processing Drum Enhancement/Removal and Spatial Isolation tools. There are no parameter adjustments on the Drum tool, which reduces percussive elements in a separation when run on its Vocal tab and enhances them on its Music tab. The Spatial Isolation tool is just that: by adjusting Frequency, Ambience, Tonal, Noise and Pan...
range controls, you can isolate the spatial elements of the wanted sound, auditioning and A/B'ing your efforts in real time.

Running the Process and either of the Post-processing functions creates a new separation file, and you’ll probably find yourself building up a number of these. Since you can only display four files at any one time on the Process screen, you may find it advantageous to start comping the best bits into one master comp track, which is a simple procedure in Trax Pro 3 SP.

With some source files, you may have produced completely usable Vocal and Music separations by this stage. However, for more complex separations where, for example, voices are ‘hiding’ in music beds, or too much of one track has been left in the other, you’ll want to move to the Spectral editing screen. Spectral editing is a skilled and somewhat time-consuming operation, but Trax Pro 3 SP’s implementation makes it probably as easy as it can be to produce good results quite quickly.

The key lies in the non-destructive nature of all ADX Trax editing, which means that whatever is removed from, say, the Vocal track is transferred to the Music track and vice versa. This, coupled with an undo history that can be up to 100 changes long, and the ability to select, audition and edit across either frequency or time or through a smart (and scalable) harmonic identification and selection tool, gives you the freedom to experiment freely in moving unwanted content from one track to the other. An Invert function helps in identifying frequencies and areas in a track that might usefully be moved to the other track, and three additional spectral editing tools give you the ability to work on unwanted noises.

The first of these, the Tonal/Noise Filter, targets harmonic or noisy components for removal and could be used, for example, to remove overhanging harmonics from a guitar chord from the sound of a singer’s breath by targeting the area and changing the balance of the filter from tonal towards noise. Doing this would protect the breath sound at the expense of the tonality in the harmonics, and its reverse would protect the harmonics as opposed to the breath. The Smart Attenuation tool uses spectral information to the right and left of a selection to attenuate the selection, so that a finger squeak from an acoustic guitar could potentially be removed without affecting the sound of the track. The third tool, Denoise, can learn the spectral profile of the unwanted sound and then remove that profile from a selected area. For example, if you had bleed on a vocal track from a loud guitar or the singer’s headphones, the tool could be used to sample the bleed from an area of the track without vocals and then remove that bleed across the whole track.

Once you’re satisfied with what you’ve got, the last step on the Spectral screen is to render your track, at which point it appears in the Trax Pro 3 SP file list and can be accessed in the other two screens. The last remaining step is to Bounce your Vocal and Music tracks (and a mix and/or a STEMS-format file if that’s appropriate) out to disk via the Process screen’s ShuttleExport function.

**Separation Anxiety**

Although Audionamix are currently emphasising the speech separation qualities of Trax Pro 3 SP, this aspect of its operation has simply been added to the program’s existing ability to separate melodic vocal and instrumental parts. With the ADX Trax Pro SP 3’s spectral editing is non-destructive: whatever you identify as not belonging to the Vocal part is automatically moved to the Music part, and vice versa.
proviso that the lead vocal or featured instrument is not buried too deeply in the accompaniment, the initial automatic separation of their melodic lines is extremely impressive, as is the software’s ability to keep the reverb with the melodic line when post-processing the refined separation after manually optimising the Pitch Guide.

Detailed spectral editing is a time-consuming process, but Trax Pro 3 SP’s non-destructive operation certainly saves time, if only because I didn’t have to worry about moving a little too much of a part of the frequency spectrum between the Vocal and Music tracks: I found it much quicker to return the excess rather than having to undo the original edit and try to refine the selection while ‘flying blind’.

The speech separation was a revelation. I began with a typical restoration project: a copy of the original recording of Martin Luther King’s ‘I have a dream’ speech, marred by a 60Hz hum. Trax Pro 3’s automatic separation produced a very usable Vocal track that required only minimal Pitch Guide editing to generate a very good refined separation. Post-processing using the Denoise tool and a modicum of spectral editing gave a result that sounded subjectively much better than simply removing the 60Hz hum component.

Never being one for an easy life, I next tried my luck on a track in which a deep male spoken voice was accompanied by a synthesized drone with prominent bass and low-mid elements. Although the automatic and refined separations managed (largely successfully) to extract the higher-frequency components without difficulty, in places where the pitches of the drone and the spoken voice were tightly intermeshed, it took a considerable amount of editing to get to where I felt relatively happy with the result. However, I hate to think how much longer it would have taken me without the ability to refine separations both continuously and non-destructively.

Extraction Fan?

Working with Trax Pro 3 SP enabled me to create, with relative ease, high-quality separations and extractions that I felt were superior to those that I’ve managed to achieve with other software packages in the past. However, my efforts pale in comparison to the results that you’ll find in the Trax Pro 3 SP tutorials and the other videos on the Audionamix web site. I’d urge you to watch these to hear the quality of the results that practised and experienced editors can produce from both spoken and melodic sources. If ‘unmixing’ of either musical or dialogue-based tracks is something you ever need to do, then you should undoubtedly audition ADX Trax Pro 3 SP; and although the perpetual licence is not cheap, the subscription options provide cost-effective ways both of evaluating the program, and of accessing it as required on a project-by-project basis.

Alternatives

I don’t know of any programs with the same automatic and non-destructive capabilities as ADX Trax Pro 3 SP but Magix’s SpectraLayers Pro 4 does enable you to manually separate speech, vocals and melodic lines from the other elements in a mix, and vice versa. For music-related requirements, the more affordable, non-SP version of Trax Pro 3 is the obvious alternative, with Celemony’s Melodyne offering some similar features.

www.soundonsound.com / December 2017 61
Brass and horns ignite the fire at the heart of the orchestra.

PART 6: A BLAZE OF TRUMPETS
DAVE STEWART

Having taken a detailed look at strings and woodwind over the last four articles, it's time to turn our attention to the tremendous sonority that is orchestral brass. For casual listeners, the jet-engine roar of brass in full flight epitomises symphonic grandeur and power — given their capacity for tumultuous, high-decibel delivery, it's no wonder that trumpet fanfares are associated with cataclysmic events like bringing down the walls of Jericho, or sounding the final trump on Judgement Day (does the MU have special rates for those gigs?). Apocalyptic duties apart, quietly played brass instruments can also sound beautifully rich, warm and expansive, making this the most dynamically versatile section of the orchestra.

The four main orchestral brass instruments are trumpet, trombone, tuba and horn. While pop and jazz so-called 'horn sections' usually contain a mixture of brass and wind instruments, in orchestral circles a horn invariably means a French horn, that complicated assembly of coiled tubing with a distinctive flared bell attached. For more details on how this instrument works, I recommend you watch Katy Woolley's excellent demonstration at www.philharmonia.co.uk/explore/instruments/horn, where you can also see the rest of the orchestra's brass team in action.

A common orchestral brass line-up would be four horns, three trumpets, three trombones and one tuba, but depending on the nature of the piece and ambitions of the composer, section sizes vary wildly. Stravinsky scored The Rite of Spring for eight horns, while Arnold Schoenberg's Gurre-Lieder demanded 10. (The latter work requires a total of 150 instrumentalists and 200 singers; not the sort of thing you should attempt to stage in your local village hall.) When budgets permit, film composers also go over the top — Joel McNeely's bombastic score for the 1998 film Soldier called for 18 horns, 12 trumpets, 12 trombones and six tubas, arranged into three separate sections so their performances could be dramatically fanned out across the stereo speakers.

Noting its enduring popularity in action movie soundtracks and trailers, sample library manufacturers have jumped on the ‘cinematic brass’ bandwagon over the last few years. One company even described a horns preset as “a ‘good pirate program’”, which baffled me until I realised it was a reference to Pirates Of The Caribbean's brass-rich score — a bit of a disappointment, as I’d enjoyed the thought of a group of scurvy seadogs practising horn pieces below decks, cursing each other in salty language for minor irregularities of intonation. Regardless of their suitability for accompanying big-screen piratical hijinks, all the current pro-quality orchestral brass sample libraries are listed in the ‘Orchestral Brass Sample Libraries’ box.

Trumpets & Trombones

As with the woodwinds, the principal brass instruments are augmented by a variety of models which extend each section’s pitch and timbral range. Standard trumpets (usually built in Bb, but C trumpets are increasingly common) are occasionally boosted by the bright, piping tones of a piccolo trumpet, prominently featured in Bach’s Brandenburg Concerto No. 2 and, 250 years later, the Beatles’ ‘Penny Lane’, both of which show off the small instrument’s precision in the top register. At the other end of the scale, the rare bass trumpet plays an octave lower than the Bb instrument. The flugelhorn, a popular jazz instrument notably played by the late Kenny Wheeler, operates in the trumpet register but produces a darker, more mellow tone; some orchestral works also feature the cornet, the lead instrument in British brass bands, said to sound halfway between a trumpet and a flugelhorn.

Trombones have been an orchestral fixture since the early 19th century, though the instrument’s history stretches back into antiquity. Unlike the rest of the brass, orchestral trombones don’t have valves, but instead use an extendable slide to alter pitch — this allows precise control of tuning, and can be a great source of ribald glissandi (as amusingly heard at the front of the David Rose Orchestra’s 1962 single ‘The Stripper’). In the orchestra and elsewhere, it’s common to find tenor trombones.
accompanied by a bass trombone, whose stentorian, rasping tone was highlighted by British composer John Barry in his Bond film scores. The 40-piece brass section assembled for Danny Elfman’s 2001 Planet Of The Apes music featured 10 bass trombones — to further max out the low end, six of the players doubled on contrabass trombone, a larger instrument designed to provide a solid bass register to a four-part trombone section. The higher-pitched alto trombone is little used in orchestral performances nowadays, but is still available to the resourceful sample user.

Another instrument of note is the cimbasso, a tall, floor-mounted contraption with a striking ‘bent’ design. Technically part of the trombone family, cimbassos are fitted with valves rather than a slider, and their aggressive sforzando bass notes have become a favourite sonority for some film composers — however, they also work well for warm-sounding quiet supporting parts. Tuba player Doug Tornquist discusses the instrument in Cinematics’ online Composers’ Workshop at www.youtube.com/watch?v=Wgfqe4XHA78.

While orchestral trumpets, trombones and tubas are known as the ‘heavy brass’, horns fall into a category of their own. In the classical repertoire, horn players spend much of their time playing alongside and blending with the woodwinds, which cultivates a certain air of musical refinement; on the other hand, some composers like to exploit the instrument’s brassy side (as heard in the braying, high-pitched riff at the front of John Barry’s ‘Goldfinger’).

Ultimately, though the players may regard themselves as separate from the brass section, horns are generally viewed as the bridge between the woodwinds and the brass, which corresponds to the position on the page horn parts normally occupy in an orchestral score.

Brass Extras

Another instrument that defies exact pigeonholing is the saxophone, traditionally (and somewhat counter-intuitively) classed as woodwind, due to its sound being produced by blowing into a reed. In jazz big bands, saxes play a different role from the brass, often operating as a self-contained, smooth-sounding melodic section in contrast to the agitated brassy stabs of the trumpets and trombones. Saxophone sections began to creep into symphonic works in the late 19th century, and solo saxes remain a popular choice for some contemporary composers.

Baroque instruments such as natural trumpets and horns have begun to crop up in sample libraries, which may be of interest to media composers engaged to produce period scores. Built without valves, these early instruments have a pure, open sound which provides a nice contrast with modern orchestral brass. In the present day, brass bands and military-style marching bands have their own array of instruments that fall outside the scope of this article: suffice it to say that euphoniums, helicons and sousaphones are all types of tuba, the last resembling a boa constrictor wrapped around the player’s body, with a large flared bell positioned above the victim’s head.

Mutes

Mutes radically alter the sound of brass instruments. The most obviously transformative is the Harmon mute, which creates a very thin, metallic tone when inserted into the bell of a trumpet, cornet or trombone. By removing its detachable stem, players can emulate the muted trumpet sound heard on many of Miles Davis’ classic recordings. If comedy’s your thing, both the Harmon and the hand-held plunger mute (which looks like something you’d use to unblock a sink) can create excellent ‘wah-wah’ effects.

Although Igor Stravinsky made extensive use of both Harmon and plunger mutes in his admirably deranged Ebony Concerto, in the orchestral world you’re more likely to find the straight mute — this metal implement removes body from the sound while leaving high frequencies relatively untouched, thus creating an edgy, cutting tone, which is particularly piercing at loud volumes. Made in a variety of sizes, the straight mute can be used with all the instruments listed in this article, from piccolo trumpet down to the large contrabass tubas and trombones.

When it comes to muted passages, horn players can choose between fitting a straight mute, or simply inserting their right hand in the bell (known as ‘stopping’). The latter method allows for varying degrees of ‘mutedness’ and also permits the pitch to be subtly varied via small hand movements. Strangely, when the hand is pushed inside to the limit, the instrument’s sound suddenly becomes brighter, creating that characteristic loud, biting and slightly buzzy French horns timbre often heard in blockbuster movie scores.
As one of the most used effects in the audio world, reverb sounds come in all forms and flavors. A great reverb sounds natural and sits in the mix perfectly. At the same time, it should not confuse you with over-technical controls, but must be easy to set up and a joy to work with. Enter FabFilter Pro-R.

www.fabfilter.com
Diagrams 1-5 show the broad playing ranges of the aforementioned instruments, as found in today’s sample libraries (the range limits may differ from product to product). Composers should be aware that the depicted top and bottom notes might be problematic for many players in real life — if you’re planning to hire live players, it’s advisable to check that any extremes of range that occur in the score are manageable by the musician in question. That said, if your track is destined to stay in the sampled domain, don’t be afraid to occasionally stretch an instrument’s upper and lower ranges by a few semitones if it suits your musical needs. Purists might complain, but if it sounds good, why not do it?

Bugle Call To The Stars

Simple, historic brass instruments like the Roman cornu and military bugle have no valves, finger holes or slides; pitch variations are achieved entirely by breath and lip control, which limits the instruments to notes within the harmonic series. This gives the familiar, rousing ‘bugle scale’ of C4 (middle C), G4, C5, E5 and G5, sounded in various permutations in all bugle calls. It follows that canny composers looking to stir up war-like, atavistic emotions will write trumpet fanfares incorporating these intervals. A classic example is John Williams’ Superman theme, based on C4, G4 and C5; Fanfare For The Common Man, written by Aaron Copland to help the American war effort in 1942 and later paraded in front of rock audiences by Keith Emerson and ELP, uses similar, triumphant-sounding major-chord figures.

The bugle call to end them all must surely be Richard Strauss’ Also Sprach Zarathustra, better known as the theme music of the 1968 sci-fi epic 2001: A Space Odyssey. It opens with four trumpets in C playing the first three notes of the bugle scale in ascending order, heralding a mighty ‘ta-da’ from the whole orchestra. You can see the brass parts notated in Diagram 6; note that the two loud accented chords...
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trombones and bass trombone.

Later in the piece, the theme reoccurs in the new key of E major — this time the tune is pitched eight semitones lower, placing it within the comfortable playing range of the horns. Consequently, the composer wrote it as a unison (not octaves) part for three trumpets and four horns, which produces a noble, heroic sound; had the tune been in the original key, horns would need to be written an octave lower, thus creating the classic and majestic orchestral brass sonority of trumpets and horns playing together in octaves.

Earlier in this series I spoke about how rhythmic pulse can be generated in orchestral music by simple rhythm patterns known as ‘ostinatos’. This time-honoured device works with strings, woodwinds and brass, and is arguably easier on the ear than clobbering taiko drums! Diagram 8 shows some typical ostinato eighth-note rhythm patterns which sound great played by sampled brass — when executing them, use a staccato patch and play with a light ‘bouncing’ action, leaving a short gap between each note. Such patterns work well at around 140bpm but, if necessary, you can temporarily slow the tempo of your track when recording the parts. I’d advise against hard-quantising the MIDI notes (too mechanical-sounding), but a ‘soft quantise’ of 80 percent will tighten up the rhythm while retaining human feel.

Though I’ve notated the rhythms as single notes for simplicity’s sake, composers very often write such patterns as repeated one out), it remained for Mankind to master the ultimate skill: the ability to fly a bicycle up into the sky. Here again, we humans were given a valuable leg-up by a helpful extraterrestrial species, in this case the phone-loving alien from Steven Spielberg’s E.T. The Extra-Terrestrial.

Written to accompany the exhilarating bicycle chase near the end of the film, John Williams’ ‘Adventures On Earth’ features a triumphal brass theme, which I’ve transcribed in Diagram 7. The theme (written in C major and incorporating a bugle-scale phrase) is played by three unison trumpets, doubled in the lower octave by two trombones and bass trombone.

Later in the piece, the theme reoccurs in the new key of E major — this time the tune is pitched eight semitones lower, placing it within the comfortable playing range of the horns. Consequently, the composer wrote it as a unison (not octaves) part for three trumpets and four horns, which produces a noble, heroic sound; had the tune been in the original key, horns would need to be written an octave lower, thus creating the classic and majestic orchestral brass sonority of trumpets and horns playing together in octaves.

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10, a trio of trombones (two tenors and a bass) plays a stately theme in the key of B minor for two bars, then three trumpets take up the tune while a lone tenor trombone continues underneath, doubling the high trumpet part down an octave. Pizzicato cellos and double basses (not shown in the diagram) pace out a steady descending B, A, G#, E repeated quarter-note figure underneath, adding harmonic movement to the upper parts’ static B minor tonality.

Our last musical extract is an outstanding example of brass harmony: Diagram 11 shows an extract from ‘Song Of Titus’ by Sir Richard Rodney Bennett, a brilliant musical all-rounder who composed symphonies and operas, performed as a jazz pianist and wrote sophisticated soundtracks for films and TV, including three-note chords, for example (from the bottom up) C4, E4, and G4 (C major), D4, F4 and A4 (D minor), or C4, F4 and A4 (F major). Such simple triads can be spiced up by (say) alternating one bar of C4, E4, and G4 with a bar of a suspended chord like D4, F#4 and G4 — try it! The C4-A4 register works well for bright-toned chordal horn ostinatos, and you can write as low as G3 (G below middle C) for deeper, more mellow-sounding horn voicings. If you want to go lower still, use trombones, for which I’d suggest a pitch range of C3 up to F#4. You can also successfully apply this technique to trumpets up in the C4 to E5 register, though that would tend to clash with a high-pitched melody — however, it could work very well over a bass-register tune played by low strings and woodwinds!

Brass Harmonies

As mentioned earlier, softly played orchestral brass can be a beautiful texture, either for supporting parts or for stating a quiet theme. Quiet sustained horn chords are particularly effective: you can hear this sonority at work in the opening measures of Thomas Newman’s ‘Voluntary Retirement’ (from the film Skyfall). Played by five horns, it features a simple, descending three-note motif supported by lower harmonies, creating a haunting and melancholy atmosphere. I’ve transcribed the horn parts in Diagram 9.

Trumpets and trombones, traditional purveyors of pomp and ceremony, also sound great playing melodic chordal movements at quieter dynamics. A nice example occurs in Gustav Holst’s ‘Saturn, The Bringer Of Old Age’, the fifth movement of the British composer’s The Planets suite. As shown in Diagram 10, a trio of trombones (two tenors and a bass) plays a stately theme in the key of B minor for two bars, then three trumpets take up the tune while a lone tenor trombone continues underneath, doubling the high trumpet part down an octave. Pizzicato cellos and double basses (not shown in the diagram) pace out a steady descending B, A, G#, E repeated quarter-note figure underneath, adding harmonic movement to the upper parts’ static B minor tonality.

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the BBC’s mini-series adaptation of Mervyn Peake’s Gormenghast. The latter project spawned a piece of music of rare beauty, based on a surreal poem chanted nightly by the schoolmasters of the ancient, decaying castle in a bizarre post-supper ritual.

Given the doomy Gothic setting, I imagine most composers would respond with dark, dirge-y music laden with groaning low strings and contrabassoons. Not so Richard Rodney Bennett — his arrangement starts out light and airy, sung by a boy soprano and underscored by sweeping, heartbreakingly lyrical strings. Producer Estelle Daniel described it as “music of dreamlike quality which plumbed the undercurrents of the soul”. The composer was knighted for Services to Music in 1998 — I’d have given him a knighthood on the strength of this piece alone.

As you can see, the instrumental brass break of ‘Song Of Titus’ features dense, complex harmonies, but you can grasp the music’s essence by playing the trumpet top line over the (greatly) simplified chord sequence I’ve indicated. While the mood of this passage is stately and grandiose, its regal fanfares incorporate fast 16th notes which require precise rhythmic articulation — see the ‘Masterclass Tip’ box for advice on how to best render that with samples.

Masterclass Tip — Staccato Overlay

When programming fanfares and fast-paced melodies, it’s important to have samples that ‘speak’ quickly. First-time users of orchestral libraries often complain that the slow, measured attack of sustained notes doesn’t cut the mustard for action-scene music, where a more urgent delivery is required. One solution is to use short-note samples played with a fast attack, but that approach comes unstuck when you play a long note: the short note having expired, you’re left holding down a key with nothing but glorious silence emitting from your speakers.

Fortunately, there are ways around the problem. Your first port of call should be to delve into the instrument folders of your brass library and check out the marcato, sforzato (sfz) or sforzatissimo (sffz) articulations, all of which are characterised by a strong attack. The ‘fortepiano’ loud-soft style can also work OK, providing notes don’t sustain for too long.

If these articulations don’t sound right, a time-honoured trick is to layer a staccato articulation over a sustains patch. This adds a nice attack to the front of notes; the staccato note dies away quickly to reveal the long note sustaining underneath, thus allowing you to hold notes for as long as you like. The technique was pioneered back in the day by US samplemeister Denny Jaeger in his groundbreaking Master Violin Library collection. It works very well with brass and strings, and is also worth considering for woodwinds, though it might sound a little unnatural with an exposed solo woodwind instrument.

When layering in this fashion, the secret is to achieve a good balance between the two patches. As a rule of thumb, start out with them at equal volume, then fade the staccato patch up and down till you find the ‘sweet spot’.

Playing Styles

As noted throughout this series, today’s sample libraries greatly benefit from the interval-based ‘true legato’ sampling introduced by Vienna Symphonic Library in 2002. This joins up melody notes in a lifelike manner, smoothing over transitions so that you hear an unbroken line rather than a series of disconnected bumps. It sounds very agreeable when applied to soaring brass melody lines and works equally well for ensembles and solo instruments, though it might sound a little unnatural with an exposed solo woodwind instrument.

In addition to the legato, sustain, portato, staccato and staccatissimo...
styles normally provided for strings and woodwind in sample collections, brass instruments have their own particular traditions and vocabulary of playing styles, which can energise a sampled arrangement. As described in last month’s article, double and triple-tongue performances work well for galloping rhythms, rapid note repetitions and hunting-horn calls. Horn rips (energetic, fast chromatic runs up to a short target note) are a great, rousing ear-catching performance style, and for maximum power, the horns’ ‘bells-up’ style (performed, as the name suggests, by tilting the bell upwards towards the audience) produces a louder, brassier and more immediate ‘blaring’ sound.

though a great resource for humorous effects, trombone slides are not well supported in orchestral libraries. Such uninhibited non-classical deliveries are more likely to be found in a pop or jazz big-band horns collection, along with trumpet ‘doits’, falls and shakes. Some pop styles do creep into orchestral collections, though; it’s not uncommon to find over-the-top Mexican mariachi vibraphone trumpet performances, and one library even includes a ‘Zampano’ molto espressivo strong vibbrato style, named in honour of the brutal trumpet-playing critic described it as being “a massive anti-hero of Fellini’s La Strada (who would soon be shown the door if he wandered into a classical concert hall). On a more subtle note, most orchestral libraries include a light vibrato option for their brass instruments, which can help to make solo lead lines sound less formal.

**The Inception ‘Braam’**

On a less subtle note, some of you may be wondering how to create the big, so-called ‘Braam’ slamming low brass racket, which has been gleefully over-used since Inception used it to scare the life out of film audiences. One critic described it as being “a massive...
blast of indistinguishable brass, like an alphorn next to an amplifier”. To my ears, it’s a combination of low orchestral brass samples — bass trombones, tubas, possibly a cimbasso or two — and layered, ambient big drum hits of the type now found in many ‘cinematic’ percussion libraries. The brass sounds detuned, and the percussive elements are also heavily processed. You can get a similar effect yourself by using your sample player’s ‘tune’ knob to down-tune the samples.

Apparently, this monstrous, blatting noise can now be heard ad nauseam on virtually every action movie trailer. A depressing thought, but one consolation is that it might indicate media composers are finally beginning to tire of plagiarising Thomas Newman’s cues!

Coda
A few words on musical expression: the crescendos and diminuendos that occur in some extracts (marked by ‘hairpins’ in the score) may look fussy, but are actually a vital expressive device. As mentioned in the first article of this series, most contemporary orchestral sample libraries offer users real-time control of dynamics via a nifty feature called Velocity Crossfading, whereby the timbre of instrument can be heard changing as its different dynamic layers are crossfaded. This control is commonly assigned to the keyboard’s mod wheel, and mastering the art of dynamically ‘riding’ your MIDI performances is one of the key factors in creating a lively, organic-sounding arrangement. This has become something of a mantra for UK orchestral sample company Spitfire Audio, who advise their customers, “On long notes, make sure you always use your mod wheel”.

I look forward to regaling you with more musical suggestions next month, when our journey through the sampled orchestra continues with an examination of percussion, harp and keyboards.

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Orchestral Brass Sample Libraries

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<td>Metropolis Ark 1</td>
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<tr>
<td>Garritan Personal Orchestra 5 [1]</td>
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<tr>
<td>(The complete orchestral instrumentation of this budget collection is an excellent educational resource for beginners.)</td>
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NOTES
* Includes instruments that may be purchased separately.
po Pre-orchestrated — instruments are blended into single patches, individual sections and instruments not provided.
* Includes different instrument families playing together.

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The difference between a multiband compressor and a dynamic equaliser is subtle and sometimes misunderstood. Their roles do overlap to some degree, and both work by splitting the signal into multiple frequency bands, but they aren’t identical. Multiband compressors use crossovers with a slope that is usually fixed, even if the crossover frequency is variable, and process each of the resulting frequency bands separately before recombining them, whereas a dynamic EQ applies the gain change directly to the gain parameters of a multiband parametric equaliser.

Both designs have their pros and cons. The crossovers used in a multiband compressor can introduce unwanted and audible phase shifts, and a parametric EQ is more ‘tuneable’ than a simple band-splitting system; if necessary, EQ bands can also be made to overlap. A multiband dynamics processor also differs from typical dynamic equalisers in that as the amount of boost or cut increases, the bandwidth remains constant. By contrast, many equalisers exhibit ‘proportional Q’ response, where the bandwidth narrows with increasing gain or attenuation, and this can often sound more natural. Sonnox have incorporated this characteristic into their five-band dynamic EQ, in a design based on their Oxford EQ’s Type Three gain/Q dependency.

Sonnox Oxford Dynamic EQ is authorised to a second-generation iLok key and supports Audio Units, VST and AAX Native plug-in formats, and there’s also a UAD2 version. In most respects the GUI follows the typical Sonnox Oxford template and includes a Preset Manager that allows your presets to be called up in any host application.

Both Sides Now
As with most dynamics plug-ins, the point at which gain reduction in each band kicks in is set using a Threshold control. Gain change can take place either side of this band depending on the Trigger mode selected: in Above mode, dynamic gain change takes place when the signal is above the threshold, while in Below mode, the gain change is applied when the signal falls below the threshold. A Dynamics control sets how the gain reacts to changes in signal level once it crosses the threshold, much like the ratio control of a conventional compressor. The response time of the dynamic processing can be adjusted using familiar Attack and Release Controls, and an Output Trim control helps match the dry and processed signal levels.

Each band also has an Offset Gain parameter, in effect a static EQ gain setting that is applied when dynamic gain reduction in a band is not being triggered, as well as a Target parameter that limits the amount of dynamic gain change that can be applied in a band. (Where bands overlap, the amount of overall EQ gain change can exceed that set for individual bands.) The Threshold control incorporates a 10dB soft knee so that the transition from Offset Gain to Target gain remains smooth. These settings can be made using either rotary controls or by dragging points in the EQ curve display, where ‘mouse-over’ prompts are available for guidance.
When setting up, each band can be soloed or bypassed, and it is possible to process the two sides of a stereo signal equally, the left channel only, the right channel only, the mid signal only or the sides signal only. The EQ types available are low and high shelving or ‘bell’ parametric; all have variable frequency controls, but the Q control is taken out of play in the shelving modes. A zoomable graphic display shows the Offset, Target and dynamic EQ curves while an FFT display tracks the spectrum of the processed signal overlaid on the EQ curve.

The side-chain feed includes its own filter which can be high or lowpass, or a single-band parametric EQ, adjustable for gain, frequency and Q, and there’s a Side Chain Listen mode for checking the effect of the filtering. While most applications use the input signal to feed the side-chain, it is also possible to designate an external side-chain source track or live input using your DAW’s usual side-chaining method. The side-chain can be fed from the stereo input signal, or individually from the left, right, mid or sides signal.

When setting up the equaliser, the user has the option of selecting how the levels are detected. As expected, Peak mode reacts to peak signal level in a conventional way, while Onsets reacts specifically to sudden increases in signal level regardless of peak level. The designers suggest using Onset detection for dealing with transients, staccato notes and sibilance, and it does indeed seem better suited to these roles than Peak mode.

**Peak Practice**

In practice the Oxford Dynamic EQ is very easy to use, and the display shows you exactly what is happening to the gain in each EQ section at all times. I was particularly impressed by just how smooth and natural the results sound, even when a significant amount of EQ is being applied, and I suspect this is due in part to the proportional-Q nature of the parametric sections. In most cases you wouldn’t even know the EQ was being controlled dynamically unless you watched the display. If you have a vocal that just gets a touch harsh on the louder sections, this Dynamic EQ is a useful tool in helping tame it, but it can also work wonders on bass sounds, abrasive guitar sounds, mixes and submixes, pulling back over-dominant elements when they get too loud or lifting out detail that is in danger of being submerged. It is also effective in attenuating narrow-band sibilance and plosives.

I’d expect nothing less from Sonnox but this Dynamic EQ really is a serious working tool that combines ease of operation with exceptional sound quality.
I always look forward to seeing what new plug-ins Universal Audio will add in each update to their UAD system, and the v9.3 selection is a real treat. As well as adding Korg’s classic SDD-3000 digital delay, as used by The Edge, we get the quirky Dytronics Cyclosonic Panner as modelled by Softube, the ENGL Savage 120 guitar amplifier (with Unison support for Apollo owners), the Sonnox Oxford Dynamic EQ (see elsewhere in this issue) and the AMS RMX 16 Expanded algorithmic reverb. The update also brings with it Unison support for the existing ENGL amplifiers.

Bit Brigade
Launched way back in the early ’80s, Korg’s 13-bit SDD-3000 had been discontinued for many years until the company resurrected it as a pedal in 2014 — check out our review of that model at www.soundonsound.com/reviews/korg-sdd-3000-pedal if you’re unfamiliar with the device. This Korg-licensed UAD interpretation has a rack-style GUI complete with numerical display of the delay time, switchable high- and low-cut filters and a modulation section offering a choice of triangle, square, random or envelope-follower modulation waveforms. There are also switchable input and output attenuators, which are important as they allow you to determine how hard to drive the plug-in — the preamp drive characteristics of the original have been replicated. Support for UA’s Unison preamp modelling technology is also included, and replicates the high-impedance input of the original hardware. The delay circuit of the original operated at line level, with an extra gain stage added when the -30dB setting was selected; this was a key part of the unit’s signature sound. In most other respects, the controls offer what you’d expect from a digital delay including a polarity inversion switch for the feedback and a Hold switch to keep the delays going ad infinitum. Additionally, you get all the tempo-sync and rhythmic subdivision options you’d expect from a modern DAW-hosted delay plug-in as well as mono, dual-mono, stereo and mono-in/stereo-out operation. If you don’t invoke tempo sync, the delay time is manually variable up to 1023 milliseconds, as on the original. The modulation section is powerful enough to stray into chorus territory, but used more subtly it adds a very textural dimension to the delays.

Panning For Gold
Modelled here by Softube, the Cyclosonic Panner dates back to 1984 and was sold and branded both under the Dytronics and Songbird names. While the heart of the unit is a two-channel panner, there’s additional EQ, enhancement and phase shift going on to give the pan more of a 3D feel. For me it doesn’t create much of a surround-sound effect, but it certainly has more perceived depth than a conventional panner. Considering how long ago this piece of kit was designed, its panning modes are actually very sophisticated, and sound far more interesting than straightforward automated panning.

Universal Audio UAD 9.3

PROS
• Excellent emulations of the original hardware, with sensible adaptations for the DAW environment.
• Clear user interfaces.

CONS
• Only that you won’t get the benefits of Unison support unless you have an Apollo interface.

SUMMARY
This update is pretty much what we’ve come to expect from UA: more meticulously recreated vintage gear plus the occasional taste of something new and genuinely useful.
Panning can be triggered manually or from ‘above threshold’ input signals, or controlled by an LFO with a choice of waveforms and a DAW tempo-sync option. Two LED displays show the pan movement and position: simple panning follows the horizontal centre line and 3D panning the circular ring. Three panels of buttons set up triggering, type of panning and stereo output mode. Knobs set the Rate, Waveform, Width and Depth of the pan effects, with a further knob at the right-hand side to set the output level. If used in mono, the unit creates tremolo or chopping effects, and if the input is stereo, only the left-hand channel is used in mono mode.

The Savage Beast
Again supporting Unison for Apollo users, the ENGL Savage 120 valve amp head is tonally very versatile, though as its name might suggest, its focus is on modern metal — in which capacity it can deliver some very heavily overdriven sounds. This plug-in was created using component-level modelling by Brainworx, who have also added their own FX Rack, which includes a range of speaker cabinet impulse responses recorded through their own Neve console, plus noise gate, delay and internal power soak.

The amp itself is a two-channel affair where the upper channel can be switched between Clean or Crunch 1 modes and the lower between Crunch 2 and Lead. Each drive type has its own separate gain control followed by EQ, separate volume controls for each drive type, A/B-switchable...
Presence controls and A/B-switchable Master volume controls.

The Bass, Middle and Treble EQ sections plus Dual Presence controls work in tandem with separate Contour switches for the upper and lower channels, an input Bright switch and some EQ pre-shaping switchable for the upper channel. There's also an overall input Sensitivity switch plus Lead Boost and Rough/Smooth tone shaping for the lower channel, and a Hi Balance control that operates only when Smooth is selected.

Despite its metal leanings, the Savage 120 is capable of delivering cutting mids without abrasive highs, and it reacts well to playing dynamics and guitar volume control adjustments. Furthermore, the power amp section can be bypassed if you fancy combining the preamp with another plug-in. However, this probably isn’t the best choice of amp for country pickers! Its clean channel goes from barely clean to nicely smoky, but after that it’s grit and grind all the way down to solid bedrock. In practical terms, you can expect to go from classic rock to just about any style of metal, with plenty of tonal fine-tuning on offer.

**RMX Bandits**

I have a soft spot for the AMS RMX16 as it was one of the first products I reviewed when I took up this writing lark with *Home Studio Recording* back in 1984. It was claimed to be the world’s first full-bandwidth (20Hz to 18kHz), microprocessor-controlled digital reverberator, and found its way onto lots of classic ‘80s records. The algorithmic engine at its heart was fine-tuned by ear and the machine gained a reputation for sounding very musical. UA already offered a comprehensive emulation; in this expanded version, the original nine reverb programs have been augmented by an extension set of nine more, originally an option that required the RMX16 remote control with barcode reader input to add to the character, just as the vagaries of tape made those old delay units sound so interesting. Certainly the market in used RMX16s shows no sign of drying up despite the high prices they still fetch. The extra nine programs don’t stray too far from the core capabilities of the RMX16 but do expand its capabilities to a useful degree.

Add into this the very capable Sonnox Oxford Dynamic EQ, and once again UA have come up with a very attractive package of plug-ins, many of which showcase the advantages of the Unison support in their Apollo interfaces. If you’re on a limited budget, it might be hard to choose, but fortunately you can demo each of the new plug-ins for two weeks to help you decide whether you can live without buying them or not.

**Alternatives**

The Sonnox Oxford Dynamic EQ is also available as a native product and the Korg SDD-3000 is available in pedal format, while SoundToys’ PanMan models a number of vintage auto-panners including the Cyclosonic model.

| Korg SDD-3000 £149; AMS RMX16 Expanded £260; Sonnox Oxford Dynamic EQ £189; Dytronics Cyclosonic Panner & ENGL Savage 120 £115 each. Prices include VAT. | W www.uaudio.com |
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Vernona quadroPOL
Eurorack Module

The quadroPOL is a neat little utility module from Vernona with a surprising number of features, allowing it to be a good utility module even in modular systems, particularly when space is at a premium. Each of its four channels has a single input for audio or control signals, plus CV control of the level, a bipolar level knob and finally a discrete output. Depending on the settings of jumpers on the circuit board, you can position the output of any channel (or all) down to the next. The action of connecting a cable to an output cuts the downward feed.

Already, its functionality covers that of a voltage-controlled four into one mixer — or other configurations as required. The bipolar control doesn’t produce an audible effect on audio, but when you’re blending control signals — e.g. several LFOs or envelopes — it’s particularly cool to be able to invert one or more of them. The level control has a central dead zone to ensure you can easily disable attenuation.

Perhaps an even more important role for many systems would be as a compact quad VCA, where typically you’d plumb in envelopes to drive the signal level. Again, the control’s bipolar nature means you can smoothly morph between a regular and inverted envelope, but the same applies to any modulation source. All inputs operate between -5 and +5 V and since LFOs typically oscillate between positive and negative voltages, the result will be a change of phase at the output, unless you apply an offset (or use a unipolar LFO). You can, of course, combine functionality, such as using one channel as a VCA while the others are a 3-1 mixer. As a VCA, the quadroPOL is clean and responsive, with a dynamic range of 80dB and no discernible crosstalk.

Other uses become apparent as you start to experiment and one of the simplest is as a series of attenuverters for audio signals or control voltages. And when no input is connected, the module generates four static DC signals at the outputs, in the range -5 to +5 V. One use of this might be to supply the offset for another channel, turning a bipolar into a unipolar source. Finally, to round things off, if you connect two audio signals (one to the CV input and the other to the regular input) the quadroPOL will produce the sum and difference of their frequencies, in effect acting as a ring modulator. Here the level control serves to balance between the input signals, therefore changing the ring modulator’s character. There’s nothing to stop you applying this process to a pair of LFOs too, and creating complexity from otherwise plain modulation sources.

The quadroPOL is one of those modules you could easily use in every patch. It meets Vernona’s usual high construction standards, being chunky and fairly accessible, given that jacks are positioned on either side of the central knob column. It works admirably as a VC mixer or quad VCA, with the ring modulation and the polarising qualities adding something extra to sweeten the deal. Paul Nagle

£ £245.99
W www.vermona.com

Liivatera TZ-VCO
Eurorack Module

Eurorack has steadily expanded into so many areas of functionality that it’s sometimes possible to overlook or underestimate modules that were once judged essential building blocks. I’m afraid I was guilty of this when a VCO from Liivatera arrived: I slipped it into my system, expecting the usual stuff. Fortunately, a few days later I remembered it was in there and took a proper look.

The TZ-VCO is 100 percent analogue, and unusual in several respects. First and foremost, the TZ stands for ‘through zero’, meaning that it is intended for linear FM, which is not easy to achieve in the analogue realm. Secondly, it has a rather interesting sync implementation complete with a variable threshold control. With no fewer than seven waveform outputs and a switch to slow it down to LFO rates, there’s clearly more to this blue-knobbed beauty than a cursory glance might suggest.

The module sports a pair of regular pitch inputs (exponential) and two FM inputs (linear), one of which has an attenuator. There’s also a three-way switch governing FM depth, and with this you can flip from gentle vibrato to — at the maximum — glistening and sparkly FM tones. Adjusting FM depth provides hands-on tonal changes and a range of harmonically rich sounds beyond the scope of most analogue VCOs.

The TZ’s top end is particularly good. It tracks way past the limits of my hearing and the through-zero behaviour, reversing phase as the oscillator slows to a stop, is vital to ensure modulation relationships are maintained even at low transposition. When you’re not exploring the delights of analogue FM, the multiple waveform outputs deliver three square waves (a straight square, a variable square and a sub oscillator), plus sine, triangle and two sawtooths, one of which is pitched an octave above the rest. The variable-pulse waveform becomes super-thin at the...
knob’s two extremes but will only disappear on application of external CV to the pulse width. Naturally enough, in LFO mode it’s a great bonus to have all these waveforms available at once.

Liivatera have also prepared a treat for sync lovers. A three-way switch sets sync behaviour so that it operates either on the rising, rising and falling, or falling edges of an incoming signal. The oscillator is based on a triangle core and its sync is designed to reverse rather than hard reset. With the switch in the middle position, cross-mod or ring modulator-type sounds can be generated — often harsh, metallic or gritty.

The Threshold control is used to fine-tune the level at which sync takes place and, given there’s also a CV input for this, you can perform cool tricks such allowing an LFO to gradually transition between sync’ed and non-sync’ed behaviour. Having a variable threshold also means that pretty much any source waveform can yield worthwhile results, while the option to choose the rising or falling edge can lead to tonal changes ranging from subtle to surprising, depending on the source.

This oscillator design has already featured in the Su modules available from www.krisp1.com and perhaps my only criticism of the smaller format is that it can feel quite crowded and the switches difficult to reach. Admittedly, my tests tended to involve most of the waveform outputs and multiple modulation connections, but when you have the option to combine FM and sync, it’s hard not to indulge. I think it was a great decision to include two FM inputs and, if space were no object, I’d have loved a second attenuator too — and octave switches (they’re so often neglected in the world of Eurorack). However, that’s probably being greedy because this is a fun and powerful VCO with many sonic flavours and applications. Paul Nagle

£ 310
W www.liivatera.com
was Eminem, the Shakespeare one was in the style of REM’s ‘It’s The End Of The World As We Know It’, Dick Turpin was Adam and the Ants (right down to having two drum kits), Cleopatra was Lady Gaga, Queen Victoria was a Bollywood soundalike, and Charles Dickens was the Smiths, which was a great one to work out with that Jazz Chorus and thin, clean amp sound!

Increasingly the songs are a pastiche of a specific track, and they are slightly less satisfying to do than a hybrid of an artist’s output or a genre. The Viking Song is an example of a generic power-rock ballad that any number of late ‘80s bands might have done. It has overly reverbed snare drums and so on, but it’s not any particular song or band. It’s distilling something of the genre and a range of bands, rather than being specific. In

“The History Men”

Matt Katz & Richie Webb: The Horrible Histories Songs

Karl Marx predicted that history would repeat itself as farce — but he never saw a role for expert pop parody!

“Increasingly the songs are a pastiche of a specific track, and they are slightly less satisfying to do than a hybrid of an artist’s output or a genre. The Viking Song is an example of a generic power-rock ballad that any number of late ‘80s bands might have done. It has overly reverbed snare drums and so on, but it’s not any particular song or band. Likewise, The Four Georges could be Boyzone, Westlife or any of those boy bands sitting on stools. It’s distilling something of the genre and a range of bands, rather than being specific. In
contrast, Norman Style is a direct send-up of 'Gangnam Style' by Psy.”

The Writing Process

At the time of our interview, Matt and Richie had just finished working on Series 7 of Horrible Histories, which aired from the 15th June 2017 on CBBC. Their impressive list of credits also includes infant favourites Baby Jake and Teletubbies, as well as a huge number of audio productions for BBC Radio 4.

To date, 95 episodes of Horrible Histories have been made, and Richie and Matt have written songs for every one, including the theme tune. They’ve also recorded and produced all of them, in Matt’s own 700-square-foot Noisegate Studios, built within an industrial unit in Warwick. Each song is commissioned by the Horrible Histories production team, which decides during pre-production meetings which element of the episode’s main theme will make the most entertaining lyric, song pastiche and music video take-off.

Richie Webb is the main composer behind the hit TV series Horrible Histories.

“I’m sure there are plenty of seven-year-olds who’ve never heard of bands like Adam and the Ants or the Smiths,” explains Richie, who acts as the duo’s main songwriter, “but there’s enough going on visually that the songs work anyway. For them it’s a funny song with some likeable characters saying funny things and they’re learning about history. If you’re an adult you might think it’s very clever, so it works on many levels.”

Once the theme and song style have been decided upon, the programme’s researchers provide series writer Dave Cohen with a booklet of historical information to help him construct a song lyric, which Richie, armed with the same booklet, turns into a song. “Sometimes Dave provides lyrics that are literally perfect and I do nothing to them,” says Richie. “Other times they need quite a bit of work and I’ll pull things around and write a few lyrics myself. If he’s apeing a particular song then it’s quite easy, but when it’s an original song in a certain style, his lyrics in one verse might lead me one way and I’ll have to make everything else fit. So his first couple of lines will suggest a rhythm and melody and if that clashes with lyrics he’s written later on then I’ll pull things around.

If we’re doing a style of music I’ve not heard a lot of I have to work out how it’s written, but when we are apeing a particular song I’ll just listen to it, pull things around a bit, then write it on the guitar or piano and sing it into my iPhone. I’ve worked that way since I started writing weekly comedy songs for a topical show called Week Ending on Radio 4 in the ’90s. I used to get the lyrics on a Wednesday, write the music on Thursday night and record it at the BBC on Friday, and so I wouldn’t forget the idea I’d to sing it into a Dictaphone.

“The songs never have to be an exact length so it’s fairly flexible. If we’re doing a really big number I know that I might have two and a half or three minutes, but usually they’re about two minutes.”

Bang On

The duo’s production process is, by now, highly efficient, as Richie explains. “I’ll turn up at Noisegate and say, ‘Right, Matt, we need to sound like the Smiths. Make it work!’ and we’ll spend a day recording it. Sometimes I’ll have a really clear idea of what I want, but other times I know that the recording process is going to make the song a hell of a lot better! We’re quite a well-oiled machine now. We just slot into our roles and bang it out.”

“You can’t obsess over every nuance of an arrangement when you only have a day to arrange and produce something in an authentic style that may have taken the original artist a fortnight to record,” adds Matt, “but equally you haven’t got to dream it up from scratch because you have source material to listen to, which is why it’s in any way achievable at all!

“The first stage is to listen to the original and see how easy it is to unpack. If it is an R&B type track, you try to identify whether there’s a Roland TR-808, TR-909 or TR-727 drum machine on there, and work out how they were processed. Then you create a sound set for the song. If it’s a straightforward rock band we’re spoofing, I’ll be looking at the sort of guitars and amps that were used and working out how they were recorded — what reverb was on them and if they sound very ‘roomy’. Anything that’s a real band, we are out of
here mid-afternoon, because we play guitar and piano. Coldplay, for example, is well straightforward. “

Despite the incredibly tight deadlines, the duo very rarely do any pre-production before embarking on a day of recording. According to Matt, the only time he does prepare in advance of a session is when the track being spoofed features a lot of creative synth work. “The Napoleon Bonaparte song is a Skrillex send-up and that’s quite dense, with a lot going on, and so I needed to compile a massive sound set. For that sort of thing I spend an afternoon in advance putting together sounds, otherwise we are pushed to get it done in a day.”

**Virtual History**

Although Matt and Richie have a variety of guitars and amps at Noisegate, for the vast majority of their recording sessions they look no further than their Line 6 Variax physical modelling guitar, which Matt says is brilliant for their purposes. “It models the pickups and hardware of various guitars, so it’s easy to dial up those esoteric, hollow-bodied jazz guitars that you’re never going to get around to buying, or just to flip between a single-coil thing like a Tele for a piercing lead and a humbuckling Les Paul for rhythm tracks. It plays like a Stratocaster, which feels slightly strange when you are making it be a short-scale Casino, but you get over that because sonically it’s so close to the guitars that it emulates. “The Variax doesn’t have any amplification models inside, so on the computer I have Guitar Rig by Native Instruments. My signal path is an SPL Channel One preamp that goes straight into my PC via an RME HDSPe RayDAT card. So we’ll start by dialling in something that works OK, then once the rest of the track is recorded I have the chance to change the amps or the gain level. When you are starting to record something the tendency is to make the sounds too big because there’s nothing else there, so often you end up backing off the settings a bit.”

Sources that can’t be DI’ed are usually recorded with Matt’s AKG C414s. “The acoustic models on the Line 6 are terrible,” laughs Matt, “and I’d never DI an acoustic guitar, so I record my Takamine EN10C with my AKG C414XLI and C414XLS. The XLS is great if you are putting that in a piano or in front of an instrument and want it to be as natural as possible. The XLI is slightly hyped with a presence peak but if you have both you can try them out. I run those through a Focusrite ISA Two which I also use for things like violin because it’s beautifully clean. I also have my SPL Channel One which has compression and a bit of EQ built into it, so I use that for DI’ing guitars and the Variax.”

For bass, Matt and Richie use a Fender Jazz bass guitar, recorded using the same signal path as the Variax, but processed instead with Guitar Rig’s bass amp models. However, their drum tracks are created entirely in the box, mostly using the suite of kits provided by Native Instruments’ Komplete Ultimate.

“I used to be a big fan of Toontrack EZ Drummer,” recalls Matt, “until Native Instruments released some fantastic libraries for Kontakt. Komplete Ultimate includes everything that Native Instruments make and the drums are superb. It’s got the Abbey Road ‘60s, ‘70s, ‘80s, ‘90s and Modern Studio kits, so if it is an acoustic kit you’re after, you can quickly dial up the appropriate drums for any genre.”
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“Neither of us play a wind instrument so they are really challenging in terms of trying to sequence, but sometimes we have the budget to get players in and we have a number of those on hand.”

**Speed Is Of The Essence**

For orchestral work, Matt’s current favourites are the symphonic libraries that are included within Komplete, although occasionally he uses other sources too. “For a long time we used a couple of things from various iterations of Gary Garritan’s Personal Orchestra. That’s so good because of how immediate and processor-light it is. You are able to control how hard things are blown or played with the mod wheel. We used that for a ton of broadcast stuff that involved orchestral instruments. And for a time I was using the exquisitely detailed EastWest orchestral libraries a lot, but you need to have time to use them. The articulations are individual instruments that you load in and that’s a bit of a bore. You get great results with it and I’d use it if I was only doing a small and defined thing, like a string quartet, but otherwise it is a bit of a luxury in our world. “But increasingly we use the Native Instruments orchestral libraries in Kontakt and the latest version of Komplete includes all of their new symphonic ensembles and solo instruments. It is a lot quicker than EastWest because of the way the articulations are nested within keyswitches, and the way you can map controllers to
expression. I started using it a lot when we did a kids' show called Strange Hill High, where each episode was based on a different genre of TV or film, or a specific film, and we had to turn around so much music. For example, there was one where it was The Day The Earth Stood Still so we had to kind of recreate the soundtrack, and there was a Day Of The Triffids one. So we had to unpack how the music worked in those shows and bring some of that to the underscore of this kids show.

“Generally, I find that rather than trawling libraries for something that’s almost right and having to settle for it because you’re out of time, it’s better to give yourself time to wander out with a microphone and record stuff. It’s fun, you get to tailor-make the sound, it’s unique and you get the right result in a fixed amount of time. For example, we did a Radio 4 comedy drama called Storm Chasers where they were chasing tornadoes in an ice-cream van. It’s difficult to find that exact atmosphere in a library so I borrowed a Transit van and drove it aggressively around the estate and up curbs and had my little Zoom recorder in the back!”

Horrible Vocals

Almost all the Horrible Histories songs are sung to camera by characters from history, which means that the actor or actress playing the lead also has to perform the vocals for the recording. These sessions are supervised by Richie and usually take place in whichever studio is currently being used for dubbing and voiceover work, rather than at Noisegate, although the results

When new episodes of the innovative pre-school TV series Teletubbies were commissioned in 2014, after a 12-year hiatus, Richie and Matt were brought in to provide the music.

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~ Jordan Rudess

Keyboard player, composer / Dream Theater, David Bowie, Liquid Tension Experiment, Dixie Dregs, Steven Wilson

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When new episodes of the innovative pre-school TV series Teletubbies were commissioned in 2014, after a 12-year hiatus, Richie and Matt were brought in to provide the music.

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~ Jordan Rudess

Keyboard player, composer / Dream Theater, David Bowie, Liquid Tension Experiment, Dixie Dregs, Steven Wilson

Horrible Vocals

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are always brought back for Matt to edit. Before the sessions, guide lead vocals are carefully mocked up by Richie and mixed to stereo by Matt, who also provides separate stems for the musical backing track and any vocal harmony parts.

“We send the cast copies of the song beforehand so that they can learn it at home, and Matt will give me separate stems to take to the studio,” explains Richie. “We only get half an hour to do a song, or an hour to do a couple of songs, so I’m usually just after something that’s close enough for Matt to be able to fix in Melodyne. Some people need their hand holding, and we go through line by line doing multiple takes of each. Other people like to do a few warm-ups and then they’re good to go. You just adapt to the performer you are dealing with.”

To help Matt edit the vocal files back at Noisegate, Richie writes on the lyric sheet how many takes were done during the session and highlights the ones he thought sounded acceptable.

“We compile the lead, and if it needs to be tracked, then we compile the tracked version or the harmonies,” says Matt. “It’s best when Richie gets a convincing performance from the actors, even if it’s not necessarily quite in time or in tune, because you can’t change the energy level or how in-character something is, but generally you can tune it and put it in time. Sometimes our backing vocals stay in if it’s not really appropriate to that. They’ve all been tuned and put in time and it is not always appropriate to that degree, so you do an element of it with somebody is in character. You can lose the character performance if you back it up to that. They’ve all been tuned and put in time and it is not always appropriate to treat a comedy song like that when somebody is in character. You can lose the character performance if you back it up to that degree, so you do an element of it with backing vocals or support it differently with the music.”

The final stages of the vocal editing process happen after the Horrible Histories episode has been filmed and the actors have mimed to the rough mix, as Matt explains. “It’s always best if they are performing to an edit that we have done of their vocals, but it depends on the filming schedule as to whether or not we’ve had chance to do that before shooting, so sometimes they shoot to the guide vocal

**The Sound Of Children’s TV**

Matt Katz and Richie Webb have worked on a lot of children’s television programmes, and say that the key is to give each one its own distinctive character and sound world.

“When we are starting a new project we try to come up with something that gives a program a specific sonic identity that’s instantly identifiable as that world, rather than it just being generic kids’ TV music,” says Matt.

“Teletubbies already had a distinctive sound when we started working on it, so we had to stick to the same kind of sounds that Andrew McCorie-Shand had originally used,” adds Richie, “although he used a GM-type bottle blow for a lot of his keyboard accompaniment, which we replaced with a sample that we made using an empty bottle of Merlot. We pitched it over four octaves and used it in 120 episodes!”

“Another show that I’ve done with Matt is about a gardener called Mr Bloom, which is a big hit on Cbeebies. It was a matter of finding a sound for him that’s quite acoustic but also a bit brassy and a bit messy, really.”

“We got to spend quite a lot of time on the Baby Jake theme tune and did all of the underscores, but the standout thing was the characters’ voices,” continues Matt. “Baby Jake’s friends are actually Richie and myself making silly noises, and Baby Jake is Richie’s nephew. We gave his parents a Zoom recorder when he was 18 months old, and for a month, they recorded what he was saying under the duvet, so it wasn’t too noisy. We chopped that into discrete collections of sounds and syllables, so that things like the names of family members became Baby Jake phrases. We sent them to production company Darrall Macqueen and they decided how to spell the words, put them in a spreadsheet and came up with a translation of Baby Jake’s language. All of his dialogue is drawn from that.

“Our big achievement was picking out phrases that were rhythmical and had a bit of implied melody to them, and re-pitching the occasional one so that it would fit a simple harmonic structure. We used those to make ‘The Yucki Yacki Yogi Song’, which occurs in the middle of each episode and has had over 147 million views on YouTube.”

**Baby Jake**

Children’s TV is perhaps not where you’d expect to encounter musique concrète!

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and I just have to make sure that I line up their studio vocals with Richie's guide, because at that point I don't usually have the pictures for reference. Sometimes we find that the audio doesn't quite fit the final pictures and we have to tweak it slightly. If they are breaking the fourth wall and addressing the camera all of a sudden, for example, we might dry that out so that it stands in front of everything else.

"So we get a day to do the backing tracks, anything from a couple of hours to half a day to edit and process the vocals, and then possibly a couple of revisits once we've seen the pictures."

**DAW Pressure**

Although many of the *Horrible Histories* songs present production challenges, Matt and Richie are adamant that they haven't yet encountered a commission they couldn't handle. The latest series did, however, require them to create the 'Music Through Time' song, which covered at least a dozen different musical styles in less than three minutes!

"In that history of music one there's an orchestral setup, a big pop band setup, a caveman type setup, plus pop music from the '60s, '70s, and modern day, so it's an example of something with multiple setups with completely different sound sets within the project," says Matt. "That project takes a lot of time to open and that's a lot of pressure on a DAW. We have to make sure that our workstation is powerful enough to handle that sort of thing, because it is not helpful to have sections of the song submixed if you need to come back and tweak things later on. And with it being so complex, in terms of the on-screen action, the lyrics and the music, it changed a few times and we had to revisit it."

"And also there's loads of different people saying things," adds Richie, "so we had to record at different times and drop it in. It took forever to put together, so that was probably our biggest challenge. Another was the 'Kings & Queens' song, which listed every king and queen since William the Conqueror. Just trying to make that work and decide how we were going to do it within the timescale was also a big challenge."

To cope with projects like the 'Music Through Time' song, Matt currently uses a custom Windows i7 PC from Scan Computers, with 12 virtualised cores, six physical cores and 32GB RAM. All mixing duties now take place 'in the box', although Matt still has a Mackie 8-bus mixer that sees action when big radio productions require extra preamps. Monitoring is handled by Adams P33As, which Matt says are very revealing, and the studio's default sequencer is Steinberg's Nuendo.

As far as plug-ins go, Matt explains: "I love Waves' Renaissance Bundle and their Classic Compressors, and I love the Novelttech Character which is an enhancer and dynamic EQ. It was originally a Powercore plug-in but now it's a native plug-in with Plugin Alliance. It has three optimised modes and you can play with your target frequency, and it's great for livening things up. So my default vocal channel setup is a Waves Parametric EQ, Novelttech Character and a Waves Renaissance Compressor.

"On the mixing bus I tickle things with Waves Renaissance compression and EQ, and if things need to be quite aggressive I use the Waves L2 Limiter. I'm also a big fan of iZotope RX audio repair software for getting rid of noise."

**Maintain The Standard**

The working relationship Matt and Richie have built over the years has brought them lasting success in both radio and TV, and is something neither party believes could be replicated with anybody else. Richie has become extremely skilled at composing pastiche songs, while Matt has developed an impressive ability to dissect and reproduce the key elements of song production, and the two together manage to keep to seemingly impossible deadlines. As for the phenomenal success of *Horrible Histories*, Richie's opinion is that it has something to do with aiming high.

"We get people to just write funny stuff instead of trying to write comedy specifically for kids, so we don’t dumb it down. But you can’t design that sort of crossover hit — it happens by accident."
Bettermaker Mastering Limiter
Digitally Controlled Analogue Limiter

Hugh Robjohns

Despite the amazing technology now available to pro-audio manufacturers, I frequently find myself disappointed at the dearth of truly innovative new products — so much of the industry seems committed to looking backwards, and happy to market homages to vintage equipment. Thankfully this isn’t the case with Bettermaker, an inspiring company which grew out of a recording studio set up in 2004. Their mission seems to be to build equipment that’s genuinely useful to fellow recording engineers, and to that end they’re more than willing to move things forward with new tools and new ways of working.

Overview
The latest member of the company’s small but interesting product portfolio is the stereo Mastering Limiter and, as with Bettermaker’s other products, it’s a digitally controlled analogue device that can also be operated remotely through a DAW plug-in. It’s housed in a substantial 2U 19-inch rackmountable chassis, with a large colour screen front and centre — from which you might already surmise that this is rather more than ‘just a limiter’. You’d be right!

At its core, the Mastering limiter is a reasonably conventional but high-performance analogue limiter, with options to operate in either stereo or Mid-Sides modes. The gain-reduction element used for the limiter is a fast-acting VCA with a fixed ratio of \(\infty:1\) (infinity to one) and a maximum attenuation of 20dB. It’s a fixed-threshold design, so the limiting level is determined by the output level control, and the amount of gain reduction by the input level control. The attack and release are adjustable manually, but there’s also a programme-dependent automatic release mode. As everything is controlled digitally, all settings are incremental, precisely repeatable, and recallable, which is ideal for mastering applications.

So far, so ordinary... but Bettermaker go beyond a simple limiter by adding a two-stage analogue clipping section after the limiter VCA (this can be bypassed for those that prefer to use an outboard A-D for clipping purposes). In essence, the clipper takes care of any transients that slip through the limiter, guaranteeing a precisely defined absolute peak level. The relative balance of effort between the limiter and clipper is effectively controlled by adjusting the limiter’s attack time.

And still there’s more! Preceding the limiter’s VCA is a rather intriguing ‘Colour’ section, which allows harmonic distortion to be introduced for a range of tonal thickening/sweetening effects. There are two separate harmonics generators, one that produces odd harmonics, and another that creates even harmonics. Both have independent drive and level controls, which determine how rich the harmonic generation is, and at what level these harmonics are blended with the source signal. Interestingly, each section also has its own band-pass filter that determines which frequency region of the input signal is used as the source to generate the harmonics. This allows odd harmonics to be added at the low end for thickening effects, but even harmonics at the high end to give a warmer, sweeter sound. The control ranges span very subtle to in-your-face grunge, and I was very pleasantly impressed with what this section adds to the unit.
But while being able to mess around with the programme dynamics and harmonic content is all good fun, in mastering the key is knowing precisely what the signal is really doing: what the peak and RMS levels really are, what the stereo image is doing, and — at least until the loudness-wars peak-normalisation practice dies off — whether there are any intersample peaks. To that end, the fourth aspect of the Bettermaker Mastering Limiter is a very comprehensive suite of metering tools (although it’s worth noting that parts of this still appear to be under development at the time of writing).

**Technical**

In line with Bettermaker’s core policy, the Mastering Limiter’s analogue signal path uses high-quality components throughout (THAT Corporation balanced line receivers and drivers, NE5532 op-amps, sealed relays, and so on). An internal linear power supply is configurable for 110 or 230 Volts AC operation, and everything is contained on a single large PCB, built mainly with surface-mount components. Entirely separate ground planes have been maintained across the PCB for the analogue and digital circuitry, to avoid unwanted crosstalk and interference between the two sections. The core limiter circuitry is based around THAT Corporation 4031 chips, which are high-performance Blackmer VCAs. I also noticed a lot of Analog Devices AD5263 ‘digital potentiometer’ chips

Finally, while all of these facilities can be controlled directly from the front panel, with preferred settings stored and recalled as presets, it can also be operated remotely from a plug-in, which is available in VST 2, VST 3 (both 32- and 64-bit), AU, and AAX formats. Not only is this convenient for project recall, but it allows the unit to be bolted into a rack somewhere out of arm’s reach, and still be operated very precisely from the listening sweet spot. Even more usefully, of course, when controlled remotely via the plug-in the unit’s functions can also be automated as part of a DAW project, further extending the creative possibilities.

**Summary**

This is a well-engineered and innovative all-analogue mastering limiter, controlled precisely and repeatably via a digital user interface or a dedicated DAW plug-in. It integrates a fast limiter with a clipper, and throws in a couple of harmonics generators for some controllable colour.

### Technicals

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At the back of the device, two channels of balanced (nominally +4dBu) line-level audio are connected via XLRs, and the unit accepts up to +21.5dBu at the input but can provide up to +23dBu out. This is slightly surprising, as, while this is fine in the European environment, where 0dBFS = +18dBu, the American standard for digital/analogue alignment requires 0dBFS to align with +24dBu... which the Mastering Limiter can’t manage. In most situations I think this can be worked around without too much difficulty, but it strikes me as an odd design decision.

Running a suite of bench tests with an Audio Precision system, the THD+N figure came out at 0.07 percent (ref +4dBu) and the signal to noise ratio is 94dB (ref +4dBu), giving a dynamic range of about 111dB. The frequency response is ruler flat between 15Hz and 50kHz (±0.1dB limits), with -3dB points at 7Hz and something well in excess of 80kHz. Crosstalk at 10kHz is better than -80dB. The harmonic generators are very powerful, and massive amounts of distortion can be dialled in, if desired, with the even harmonics being much less obvious than the odd harmonics, of course. I found it best to keep the drive levels well down, at around 10-30 percent, and then tweak the amount level to taste.

A type-B (square) USB socket on the rear panel provides an optional link to a computer (compatible with USB 2.0 and 3.0) for the remote control plug-in.
and firmware updates, and mains power is connected via the ubiquitous fused IEC inlet (with a slide switch to select the operating voltage).

A very useful block diagram is included in the manual to show the signal path, since there’s a lot going on here. The input signal is split four ways with direct feeds to the metering and a summing node, while the other two go to the even- and odd-harmonic generators (each via an adjustable band-pass filter). The outputs from these two harmonic generators are recombined at the summing node so that the harmonic content can be mixed with the input signal as required before passing into the limiter section.

The limiter section comprises the VCA and soft-clipper, but these are sandwiched between a pair of Mid-Sides matrices so that this dynamics processing can be applied either to the normal stereo signal, or its Mid-Sides equivalent. Both the limiter and clipper have fixed thresholds, as previously mentioned, but the clipper threshold is set 3dB higher than the limiter’s. So more input level means more limiting and, if enabled, potentially more clipping. The idea of the clipper is primarily to catch any brief transients that slip past the limiter (which they will because of its non-zero attack time). Finally, the output from the limiter section (post M-S matrix) is routed through a hard-clipper (for the ultimate in peak level control!) to the outputs, and a split of the output signal is routed back into the metering.

Everything in the Mastering Limiter is managed through a pair of 32-bit micro-controllers, the firmware of which can be updated via the USB connection. Apparently, one of these devices is wholly responsible for the limiter’s VCA operation, while the other takes care of the user-interface, preset store/recall, USB connectivity, and the metering functions. In the course of this review, I had to update the firmware for compatibility with the latest plug-in software, and I was perplexed to find the new firmware listed with two revision numbers — v0.97 and v0.62 (released in August 2017). However, it makes sense when you realise that there are two micro-controllers doing entirely unrelated things, each with their own firmware. Upgrading the firmware is simple enough, but there are two distinct stages in the process, which might catch out the unsuspecting!

Operation

The front panel is deceptively simple, with a large colour touchscreen dominating the centre, sandwiched between two large rotary encoders. The large left encoder knob always adjusts the input level (and thus the amount of limiting), while the one on the right defaults to adjusting the output level but is also used for setting the contribution of the harmonic generators and various other things). Other controls comprise a chunky mains on/off switch, a smaller button on the left which bypasses the unit, and two small rotary encoders on the right that normally adjust the attack and release times. These also feature push-button actions (although I couldn’t find any functions in the current firmware that employed this facility), and are velocity-sensitive to flip between coarse scrolling and fine increments.

On power up, the display features a horizontal bar-graph at the top (spanning -36 to +3 dBFS) for input or output levels, with numeric displays for the peak levels. A blue gain-reduction

Alternatives

The world of mastering limiters is a rarefied one, and most mastering dynamics processors around the price of the Bettermaker are actually compressors, rather than just limiters. I’m thinking of devices like the Maselec MLA2, the IGS Tubecore ME, and the Buzz Audio DBC-M, amongst others. However, none offer the range of precise controllability of the Bettermaker or the adjustable Colour feature, let alone the integrated remote-control plug-in option, all of which make the Mastering Limiter unique. If plug-in controlled analogue compressors appeal to you more generally, also check out the offerings of Wes Audio and Tegeler Audio.
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The beta plug-in software includes an undocumented scrolling timeline display, which shows the input and output levels, gain reduction and clipper action.

The meter below covers an 8dB range, and a second short bar-graph meter shows the action of the clipper (over a 6dB range). To the right is a display of the harmonic (Colour) generator settings.

Five numeric parameter boxes below the meters provide the input level, attack time (0.1 to 2.5 milliseconds), release time (0.1 to 1.3 seconds), and output level, while seven virtual buttons at the bottom of the display access various setup functions such as a preferences menu, switching the main level meter between the input and output signals, and engaging the clipper. Interestingly, when the clipper is activated the attack time parameter becomes a ‘clippiter’ value showing the ratio between limiting and clipping. In essence, small numbers set a fast attack time so that the limiter does most of the work; large numbers use a slower attack time passing more transients through to the clipper. A value of 100 percent disables the VCA limiter altogether.

The middle button in the row engages the Mid-Sides matrix for the limiter/clipper section, whereupon the middle parameter box allows the Sides level to be adjusted ±8dB to alter the stereo width. Next along the row is the limiter’s Intelligent Release function (‘Irel’), which disables the manual release time parameter, controlling the release time depending on the signal content and the amount of gain reduction. Another soft button here resets the parameter values to default settings, while the last button calls up the main unit configuration menu page where, amongst other things, the Colour generator can be set up.

The odd and even harmonic generators can be switched on or off independently, and their drive levels, the centre frequencies of the band-pass filters, and the contribution amounts can all be adjusted using the P1, P2 and output knobs. Also, since adjusting the harmonic generators is likely to affect the overall signal level going into the limiter section, the input knob remains active so that the amount of limiting/clipping can be adjusted as necessary.

In the current firmware, the metering options include a spectrum analyser (labelled FFT), which can be displayed either as a 31-band RTA style, or as an FFT graph complete with a moveable frequency cursor. There’s also a K-meter display option with all three headroom modes, and four different moving-coil meter emulations (PPMs, two different styles of VU, and a digitally scaled meter).

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However, two of the metering modes were not fully operational at the time of writing, and these are the BS1770 loudness display, and a phase/vectorscope display.

Other functions accessed through the configuration menu include preset memory load/save (up to 399 unit settings can be stored, and a QWERTY keyboard appears on the display for titling each preset), display brightness, calibration of the unit’s metering, firmware revision details, serial number, and so on. Being able to calibrate the unit’s metering to exactly match a DAW’s meters (taking into account any interface converter sensitivity variances) is a very sensible idea and the process is very straightforward. The only significant omission I can think of is that there’s no display contrast/azimuth control, and that’s a shame because the screen is quite difficult to read when viewed from above — as it might well be if the Mastering Limiter were bolted into an under-desk rack.

### Plug-in

In the interests of fairness, I should state that while the hardware was the final production version, the plug-in available during the review period was a beta release rather than the final product — so some functions may have changed by the time you read this. Nevertheless, when the hardware was connected over USB the plug-in found it instantly, with parameter changes made on the hardware being reflected instantly in the plug-in, and vice versa. Oddly, I couldn’t find a way of toggling the plug-in’s level metering between input and output, although the plug-in responded to I/O meter selections made at the hardware unit. Also, none of the dedicated metering options were available on the plug-in; only the I/O level bar-graphs, gain-reduction, and clipping meters are shown.

### Impressions

Like all of Bettermaker’s products, the Mastering Limiter looks elegant and refined, is very solidly built, and in its default mode it sounds very clean and transparent. For all of my testing, I plugged the Mastering Limiter into an analogue insert loop of my Crookwood mastering console, and compared its performance mostly against a Drawmer Masterflow DC2476 digital mastering processor, which I often use, as well a variety of plug-ins from UAD and others. In my experience, all limiters, whether analogue or digital, sound different and respond differently to the music passing through them, so making...
direct comparisons is often more misleading than helpful. What matters is whether a limiter can effect the appropriate dynamic control required in any given situation, and whether the various parameters allow the appropriate settings to be found quickly and easily.

The answer to these questions is an unequivocal "Yes!" for the Bettermaker Mastering Limiter. I found it very easy to dial in the required input level and attack times to exert the precise amount of peak control I required. I generally left the release time in the 'Irel' automatic mode which seemed to do a very good job most of the time, although sometimes I found setting the release manually achieved the effect I was looking for more readily.

I’m not personally a fan of peak-normalisation at all, and so using heavy limiting to push the perceived loudness up is not something that comes naturally or comfortably to my ears, and I would never intentionally clip the signal when mastering... but I do recognise that these techniques are still widely employed and often demanded so, gritting my teeth, I explored the clipping options and found that they are implemented very well in the Mastering Limiter. I know many mastering engineers like to introduce some mild clipping through their favoured A-D converters, and that option remains here, of course, but the combination of soft and hard clippers in this Mastering Limiter can certainly be used to good effect to squeeze out those last fractions of a decibel of "loudness" if desired.

The ability to limit and clip signals in the Mid-Sides mode can be very handy indeed, as can the option to widen things a little — although I usually find that this is something that works best when approached in a frequency-selective way, rather than as a broadband tool.

If I had to name one feature that really impressed in this Mastering Limiter, it would undoubtedly be the Colour section. Dialling in a little saturation distortion can often work wonders on a mix, but the level of control provided here is outstanding. Being able to ‘tune’ the spectral regions that generate most of the distortion components, and adjusting how strong those components are and how much of them makes it into the mix creates a very powerful and creative tool indeed. It’s very easy to overdo the effect, but when treated with care and respect this feature almost justifies the cost of the unit on its own — it’s that good!

The extended metering facilities are nice to have, but I suspect are of limited practical use in reality. If I need K-metering, vector-phase analysis and so on, I would much rather use the tools I already have in the computer, where I can also view them on a larger screen. However, in a wholly traditional analogue mastering setup, having these tools would be very useful.

Overall, I think Marek Walaszek and his team at Bettermaker have come up with a really interesting, powerful, and creative tool in the Mastering Limiter, and for anyone looking to invest in a very nice hardware processor to add some polish to their mixes, I can certainly recommend this one.

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Emergency measures iConnectivity unveil ‘bulletproof’ solution for live playback

For live shows that rely on audio backing tracks, virtual instruments or the likes of Ableton Live running on a laptop, the thought of that laptop going down is every engineer’s worst nightmare. That’s why professional acts generally employ two or more identical systems running in parallel so that, should a problem occur, the engineer can simply switch from one to another. However, this does rather depend on having someone constantly monitoring the situation, ready to hit the switch.

Having carved something of a niche out of designing audio and MIDI interfaces that cater to mobile music-makers, live performers and DJs, iConnectivity have now introduced an audio/MIDI interface built specifically for live use, with automatic redundancy features built in. The PlayAudio12 lets you run two laptops in parallel, connected to the twin USB ports on the front panel. Should anything happen to the first machine, from a crash to the USB lead getting pulled out, the interface instantly switches to the second laptop with no audible interruption. The unit provides 10 audio outputs on quarter-inch jacks at the rear, plus a stereo headphone output on the front panel, with 24-bit/96kHz D-A conversion. On the MIDI side, you can connect up to eight USB MIDI devices via a USB hub while a four-port Ethernet MIDI connector at the rear caters for long cable runs onto the stage. A footswitch input lets you add a manual A/B switch and there’s also a control output for slaving additional PlayAudio12 interfaces, two of which will fit side-by-side in a 19-inch rack. This unique interface is also surprisingly affordable, with a price tag under £500. It should be available by the time you read this.

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Weapon X sE Electronics introduce dynamic instrument mic

Further expanding their offering for the stage, sE Electronics have launched a new dynamic mic. The V7 X features a new capsule with an aluminium voice coil, specifically tailored for use on snares, toms, guitar cabs, brass and other instruments.

Like the existing V7 vocal mic, the V7 X features a supercardioid pickup pattern for enhanced feedback rejection, rugged all-metal construction and an eye-catching red internal windscreen (though a black one is also included should you wish to tone things down). However, the new capsule design provides a flatter, smoother overall frequency response with increased low end. Other features include an internal capsule suspension system to reduce rumble and improve isolation, a gold-plated XLR connector and a steel grille, surrounded by a protective ring with bevelled edges that stop the mic rolling around when set down flat. The V7 X is available now, priced £89.

The company have also announced a wireless version of their original V7 dynamic microphone, intended for use with compatible Shure handheld transmitters. The V7 MC1 capsule exists for those who are comfortable with their choice in handheld transmitters, but want to improve the fidelity of their on-stage vocals, say sE Electronics. It will be shipping soon with pricing TBC.

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ME generation Allen & Heath debut ME-500 16-channel personal mixer

The ME-500 is the latest personal monitor mixer from Allen & Heath. Promising plug-and-play setup, a streamlined and easy-to-use feature set and 16 mono/stereo mix channels, the ME-500 is compatible with all of Allen & Heath’s digital mixers as well as third-party systems via the ME-U hub.

Providing a more cost-effective alternative to the ME-1, the ME-500 is designed to be simple for performers to operate, with 16 backlit, dimmable buttons to select a channel and a push-button rotary encoder to adjust volume and panning. There’s also a headphone level knob and dedicated solo and mute buttons, while up to eight user presets can be saved and stored on a USB stick. At the rear, there are separate quarter-inch and mini-jack headphone outputs plus a balanced mono out to feed a stage monitor. Locking Ethernet in and out connectors allow multiple ME-500 units to be either daisy-chained or deployed in a star topology, with no limit to the number of ME-500 or ME-1 mixers in a single system.

The unit can be powered by an external DC mains adaptor but will also accept PoE (power over Ethernet). Other features include a built-in limiter within the headphone amp to protect performers’ ears from unexpected level spikes and a threaded insert allowing the ME-500 to be mounted directly onto a mic stand. This new personal mixer should be available soon, with a street price around £349.

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Mike Crofts

Anyone who has used or been in the market for active portable speakers will more than likely have seen, heard or used an RCF product at some time. The Italian company, based in Reggio, are well-established as designers and manufacturer of transducers and speaker systems, and they have built up a fine reputation for quality and reliability over the years. Their ART range of portable loudspeakers have proved very popular, and the series is currently enjoying ‘Mk4’ status.

The model I’ve been using for the last few weeks is the ART 712-A Mk4, which sits toward the smaller end of the range. The full Mk4 series offers the choice of eight-, 10-, 12- and 15-inch woofer-based two-way driver formats, with additional model variants in the larger versions offering the option of a 1.4-inch exit compression driver and, in the case of the 15-inch models, upgraded drivers, higher maximum output and lower crossover point (in the top-of-the-range ART 745-A).

All the products in the current series use the same amplifier module (rated at 1400W peak, that is 700W RMS split 500W to LF and 200W to HF sections), with the exception of the smallest model, the ART 708-A, which is equipped with something more suitable (800W or 400W RMS) for its size and intended application.

The ART 712-A Mk4 is an example of what nowadays must be the most common format of portable powered speaker, namely the 12-inch, two-way design housed in a moulded cabinet. This particular marketplace is crowded, with makes and models from many manufacturers, and the choice of which one to buy depends on what features the potential user values most, be it back-panel features, sheer power, weight, number of handles or simply price.

The ART 712-A Mk4 speakers arrived at my workshop wearing genuine RCF

PROS

- Smooth audio performance with plenty of power for the size.
- Lightweight and easy to handle.
- Simple control panel, straightforward and uncomplicated operation.
- Neat appearance, and check out the fab covers.

CONS

- None.

SUMMARY

A no-frills but eminently capable speaker that will find a lot of uses in a variety of situations.
tailor-made covers, of the type that unzip at the front and rear to allow the unit to be operated with the covers in place. I was immediately impressed with the quality of these covers, which are very effective and don’t detract from the external appearance of the speakers, but are very easy to remove when not required. After opening the Velcro flap on top of the cover I discovered a large, comfortable top handle with a nice deep recess easily large enough to allow comfortable lifting. Top handles really come into their own when you need to move the speakers around on the floor, particularly when loading them into the back of a van. The ART 712-A weighs about 18kg and is easily manageable, although you obviously can’t get to the side handles with the covers in place. Transporting a pair of these around and lifting them up onto stands presented no problem at all, and they have a pleasing compact appearance that would fit in with most live applications from pub bands to corporate events. The reflex-ported moulded cabinet is angled so that it can be used as a ‘sideways’ floor monitor, and it has a substantial metal pole socket underneath. The drivers are of course RCF products with a crossover point at 1.6kHz; the 12-inch woofer has a 2.5-inch voice coil and the compression driver has a 1.7-inch diaphragm behind a standard one-inch exit on a 90 x 60-degree waveguide.

Inspecting The Facilities

The front of a portable powered speaker generally only tells you who made it, but the back panel gives you a good idea as to what you can do with it. In the case of the ART 712-A the control panel tells you that it is an uncomplicated product that inexperienced or unfamiliar users should have no difficulty connecting up and getting it working. The amplifier module has a large, smooth heatsink with connector and control panels above and below; at the top is the input section with a direct loop-through for connecting another speaker, and simple controls for input sensitivity (mic or line) level and fixed EQ boost. Three LEDs indicate signal present, limiting and system status — essentially an indication that thermal limiting has kicked in. Down below is where you plug the IEC mains lead in and switch the power on, so all in all it’s not going to require a master’s degree in sound technology to get the ART 712-A rigged up and working.

Performance

Having mounted the ART 712-A speakers on poles above a pair of my usual 15-inch subs I ran various recorded material through them at different volume levels. I stuck to my ‘test list’ of tracks, which covers quite a wide range of musical styles, from thumpy ’70s Crusaders through acoustic, jazz and contemporary stuff to orchestral and choral pieces both large and small.

These compact RCFs produced a well-balanced, well-rounded sound with a pleasing crispness to the upper-mid and HF content. Vocals were always clear and the smooth delivery was maintained at higher listening levels, which produced a fatigue-free experience — I’d say these speakers do handle a fairly hot incoming signal well, and appear to be designed to operate with the input level control wound fully clockwise rather than at a nominal ‘centre detent’ position. This is, in my view, a good design point as it lessens the risk of accidentally overdriving them, especially when
being set up by an inexperienced user, and if the input signal is way too low for normal operation there’s always the option of using the ‘mic’ setting for increased sensitivity. The bottom end sounded very well defined and without any unpleasant evidence of DSP ‘humping’ to achieve a decent bass output from a relatively small cabinet. The manufacturer’s web site tells you that there are “no complex menus, no knobs to understand”, so although on the one hand you can’t tweak and twiddle to fine-tune every application, it does mean that there’s nothing to get wrong after the mixer other than placement and playing level. I reckon these speakers will definitely appeal to anyone who has written ‘simple but effective’ on their list of requirements. The only controls are input level, mic/line sensitivity and a flat/boost switch that introduces a degree of lift in the mid and lower ends of the overall response for use at very low levels, for most live-sound use you’d probably never use this feature but it would be handy for, say, low-level audio-visual playback.

I liked and appreciated the simplicity of the control panel — I can’t see how you could go far wrong hiring these out to anyone who can operate a tap. I also very much like the ability to plug a microphone directly into the speaker, which means one ART 712-A plus a mic equals a complete vocal PA, and that’s an equation that works for me.

Live It Up

Taking the RCFs out on a couple of live gigs was both easy and rewarding in a ‘job done’ kind of way. They are — as mentioned already — easy to set up and get running, both electrically and physically, and they are a very good example of plug-and-play PA components. Out of habit I set the level controls to centre position (there’s a mark in the middle of the printed scale that sort of suggests this, although it’s not the ‘unity’ position) and because I didn’t need the system to be too loud I just left it this way.

The first event needed only live vocals against an 18-piece band with lots of unmiked brass, and the results were excellent — plenty of projection and, therefore, clarity across a programme of ballads and uptempo classic pop. I appreciated the light weight and good handle design of the ARTs, both when setting up and packing up, and although we weren’t trying to achieve stadium levels the output levels were more than enough for the 200-seater venue.

As I only had one pair of the RCFs to try I wasn’t able to use them as wedge monitors at the same gig, but I did take them along to rehearse with the same band, and here the simple mic-into-speaker setup does work, but you have to be careful as there’s no monitor EQ facility so I would tend to use them as monitors via a mixing desk to fine-tune every application, it does mean that there’s nothing to get wrong after the

“It maintains an outward simplicity that makes it an easy-to-use and flexible product, whilst having enough clever technology under the hood to achieve impressive audio performance.”

” ﬁrstly, effect is to produce smooth crossover performance and to bring the different frequency components of the output signal into focus by reducing phase ‘smear’, which can be an undesirable product of filtering.

If you want the full story, or just some light bedtime reading on the advantages of recursive least squares over least mean squares, you can download the white paper from RCF’s web site as a good starting point. Or just go and have a listen.

Summary

The ART 712-A Mk4 is a smart little active speaker from a reputable stable. It maintains an outward simplicity that makes it an easy-to-use and flexible product, whilst having enough clever technology under the hood to achieve impressive audio performance. I didn’t find anything to dislike about it and I’d be more than happy to use it for my own events or to put it into my hire list. It seems like a tough unit too, with a generally well-engineered feel to it. The RCF protective covers are also very well made and highly practical for road use, and I’d certainly be bargaining hard with my supplier for these to be part of my purchase deal!

Finally, when contemplating any purchase I also like to consider how any manufacturer stands behind their products, and although these, like most, are produced away from these shores, I am hearing good things about the sales and support being provided by the RCF UK team. In what is probably the most crowded and competitive slice of the portable live-sound speaker market, I’d say that the ART 712-A Mk4 is definitely worthy of consideration whatever the application. As ever, if you want to find out more, then visit the RCF web site, and plan a visit to your local pro audio dealer.

Alternatives

Practically every PA company makes an active, 12-inch, plastic-enclosure speaker — brands to look out for include Mackie, Presonus, Peavey, Yamaha, JBL, Alto... and many others!

£ 489 per speaker including VAT.
RCF UK +44 (0)844 745 1234
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Our S21s offer us the sound quality and robustness we need from a compact console to cover all our touring demands. Plus it’s a DiGiCo for under £5k, that’s amazing.

STEVE WHITE – PATCHWORK LONDON

Patchwork provide professional audio solutions coupled with Tour and Production Management. They currently work on and supply tours for Wilkinson, Skunk Anansie, Machine Gun Kelly, Fickle Friends, Bonzai, VEVO and Sony.
First launched back in 2014, Mackie’s DL32R is an attractively priced, Wi-Fi/iPad-controlled rackmount digital mixer. It features 36 inputs, 28 busses, 14 outputs and Dante connectivity, and possesses a fairly comprehensive list of features and facilities. Although Mackie’s Master Fader iOS app (now in v4.6.1) offers an intuitive user interface that is more than capable of handling the complexity of the DL32R, the absence of a compatible hardware control surface may well have kept it out of consideration for many professional users.

The recent release of the Mackie DC16 Digital Control Surface, which is designed to complement the DL32R and the Master Fader app, has created the Axis digital live mixing system — the modular paradigm of which offers the flexibility to cope with a range of applications, both mobile and installation.

Since my esteemed colleague Paul White looked at the DL32R in detail back in the December 2014 issue of SOS (www.soundonsound.com/reviews/mackie-dl32r) I don’t propose to cover it in this overview of the system, although references to it will doubtless appear.

The DC16 control surface for Mackie’s DL32R digital mixer gives you the best of both iPad and hardware control.

The first thing that struck me on opening the flightcase that the DC16 arrived in was just how physically large it is compared to similar, competitive consoles that carry 17 touch-sensitive motorised faders. On the other hand, its size means that the DC16 does give an exceptionally clear view of its status, due in part to the 26 backlit, full-colour display screens, 16 of which sit at the very top of each fader strip.

Scratching The Surface

The first thing that struck me on opening the flightcase that the DC16 arrived in was just how physically large it is compared to similar, competitive consoles that carry 17 touch-sensitive motorised faders. On the
These screens can display not only each strip’s identifying graphic or photograph but also the parameters of functions assigned to it in text format. A 17th screen, to the left of that of the first fader, displays the identity of the input or output being worked on. A benefit of the screen size is that it allows for fonts that can be quite large, a benefit for older, spectacle-wearing engineers like myself.

The other striking physical feature of the DC16 is that it has no touchscreen display and, as a result, it is a mere 8.4cm thick. Mackie appear to have made the not unreasonable assumption that there will be at least one iPad running Master Fader within the Axis system, so the DC16 incorporates an iPad holder — the Smart Bridge — that sits at the centre of Mackie’s concept of “Surface to Wireless Mixing”.

An iPad running Master Fader ‘knows’ when it is sitting in the Smart Bridge so that, when it is removed — for example, to set up a monitor mix — and then replaced, its screen automatically reverts back what was being displayed at the point of removal. This gives a seamless transition backwards and forwards between surface (Smart Bridge) and wireless (mobile) operation. Although the DL32R in an Axis system allows the wired connection (Dante over Ethernet) of two DC16s and the Wi-Fi wireless connection of up to 20 iOS devices running Master Fader (iPad, iPhone and iPad Touch), a DC16 Smart Bridge provides only space (and USB ports) for three iPads (iPad 2 and above).

One USB port acts as the Control port. A Lightning-equipped iPad connected to it establishes a wired connection that gives it, via its touchscreen, full control of the DC16 and the ability to record and play back 2-in/2-out, 44.1/48kHz, 16/24-bit digital audio streams. If an older iPad with a 30-pin connector is connected to the Control port, then, exactly as with iPads connected to the other two charging ports, it can be connected wirelessly to the DL32R and, when placed in the Smart Bridge, will function both as a touchscreen interface and as part of the DC16’s display.

**Pushing Buttons**

To my way of thinking, the DC16 is perhaps best thought of as an interface to Master Fader rather than being its remote control. In designing the surface, Mackie have created hardware-based editing access to the app’s software functions. On the DC16, a two-row, eight-button block sits at the left-hand end of the line of fader-strip screens, the top row of which selects a single editing function simultaneously across all channels — gain, high-pass filter, send, pan, trim, low-pass filter, delay and a not-yet-implemented user-selectable function. The bottom row activates editing functions that apply only to the selected channel — EQ, graphic EQ, dynamics, effects, assignment and aux sends — and can also bring up the DC16’s setup screens and the transport controls for the two-channel iPad or multitrack hard drive connected to the DL32R. At the other end of the line of screens, up/down buttons not only allow you to step through any additional pages of the function being edited, but also, when pressed simultaneously, to flip the parameters being edited onto the faders, allowing you to use these instead of the encoders.

In addition to the above, a ‘Fat Channel’, sitting just in front of the centre iPad in the Smart Bridge, gives instant access to most of the channel-specific functions whenever any of its controls are moved or pressed. The gain/trim, HPF/LPF selection is identical to that of the left-hand switch block, but the EQ section is expanded.
Two vertical banks of four screens and 16 or across the surface one at a time. VCA and matrices, in either banks of effects returns, aux sends, subgroups, switches to take you through channels, intuitive, with bank and channel up/down.

Navigating around the DC16 is fast and tight. The screen display of the central iPad (the one that could/would be connected to the USB Control port) is set automatically to follow the selected control function, giving you the option of combining touchscreen and encoder to give gain (cut/boost), frequency and Q over four selectable bands. There is no access to the graphic EQ via the Fat Channel, however the dynamics section is expanded to include the gate’s threshold and range controls, and the compressor’s threshold and ratio. Pan and delay share the same encoder and the final two buttons activate 48V phantom power and invert the polarity of the channel input.

A fader strip’s select, mute and solo buttons, all of which illuminate when active, are placed within it. Metering on the DC16 comes via individual mono LED ladders (stereo in the case of the Master fader) in the lower portion of the touch-sensitive fader slots, with three-LED indicators at the top of the slots to display compressor gain-reduction status.

The integration of these physical controls and the Master Fader app is fast and tight. The screen display of the central iPad (the one that could/would be connected to the USB Control port) is set automatically to follow the selected control function, giving you the option of combining touchscreen and encoder to select and recall the fader positions and other settings of the inputs and outputs (assigned to a mix via the Assign button, or selected for viewing as part of a group). In its Mix mode, the right-hand bank gives you access to all 36 possible mixes (LR, aux 1-14, reverb 1/2, delay, subgroups 1-6, VCAs 1-6 and matrices 1-6). In its Masters mode, you can view the positions of master faders for any of those seven mix categories. In addition to the eight mix selector buttons, another four buttons allow you to scroll through the list of mixes, switch between Mix and Masters mode, clear all active solos and activate the microphone plugged into the rear-panel talkback input.

The left-hand bank of screens and switches enables you to recall the fader positions and contents of one of six View Groups, which are freely customisable selections of channels, auxes and subgroups that you want to be able to view together. Similarly, you can create a further six selections that you want to mute simultaneously. In a View Group, channels other than those selected are hidden and, in a Mute Group, only those selected will be muted. Although channels, etc, can be selected as part of View and Mute Groups on both the DC16 and Master Fader app, the groups themselves must be created and named in the app.

The Show display, the 26th screen, sits in the upper right-hand corner of the DC16 surrounded by its associated soft switches. A Show is made up of a series of snapshots, stored in sequential order, that can be manually recalled as the show progresses. A channel safe function allows you to remove channels from snapshot recall, leaving them unchanged. Although snapshots can be stored in a show and a show recalled from either the DC16 or from Master Fader, only the app can create and name a show and name a snapshot. Currently, although snapshots can be freely selected and recalled using a pair of up/down buttons, there does not appear to be any method of re-ordering the snapshot list other than overwriting one with another. However, a More button, as yet unimplemented, holds out the promise of additional Show functionality, so one can live in hope.

Finally, we come to the only analogue controls on the DC16. Talkback sets the input gain of the rear-panel balanced XLR Talkback input, whose output is routed to the DL32R via Dante; Monitor sets the output level of the balanced TRS quarter-inch jacks carrying, in their default setting, Dante-delivered monitor feed from the DL32R, and Headphone controls the output level of the front-panel headphone output, which is paralleled from the monitor feed. A rear-panel stereo TRS mini-jack can accept an input signal up to 16dBu and is intended to be the source of background or interval music. The central, wired iPad connection can also act as a stereo playback source, and the selection is made on the DC16’s setup screens.

**Bold As Love**

Navigating around the DC16 is fast and intuitive, with bank and channel up/down switches to take you through channels, effects returns, aux sends, subgroups, VCAs and matrices, in either banks of 16 or across the surface one at a time. Two vertical banks of four screens and their associated soft switches that sit at either end of the fader area allow you to use knurled encoder knobs and banks to edit parameters such as input levels and pan. The large display shows the channel number and selected parameter, while the encoder lets you adjust the selected parameter. The encoder can be used to adjust any parameter on the screen, such as compressor gain, in a single sweep.

The DC16’s rear panel houses outputs for a stereo set of monitors, a talkback mic input, a stereo aux input, a footswitch socket, two USB charging sockets, USB Control port for connecting the ‘master’ iPad, an Ethernet port for Wi-Fi connectivity, and Dante sockets for connecting to a DL32R mixer.

“For installations in multiple spaces, houses of worship and the like, the Axis system could also offer a very attractive solution.”

Alternatives

If you’re considering a Mackie Axis, you’ll also be looking at comparably priced digital consoles from the likes of Allen & Heath, Midas, Soundcraft and Yamaha. Those companies also offer more affordable alternatives, as do QSC, Behringer, PreSonus and Roland.

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more. Providing that nothing goes wrong
(no Wi-Fi interference, no crashes, no
emergencies) then a monitor engineer
and an FOH engineer with an iPad
each would have little or no difficulty in
setting up and running a relatively simple
PA setup without a DC16 in sight.

Operationally, Master Fader allows
you to set up a mix offline quickly
and easily, so initial preparation for a
show doesn’t require anything more
than an iPad and a channel list. Auxes,
subgroups, VCAs, mute groups and view
groups can be put together and named
as required, and these then appear
on the channel input routing screen
where the necessary assignments can
be made. Master Fader also offers a
useful selection of vintage and modern
EQ, compressor and gate presets that
give you a good starting point for any
mix. On the effects side, selecting two
reverbs and a delay from the available
presets is similarly simple, and you have
the opportunity of editing the reverbs
with some real granularity, should you
so wish. Talking about granularity, the
app’s reference guide takes up 333
pages, giving you some idea of just how
comprehensive and deeply detailed
Master Fader is.

I had intended to use the Axis system
to mix a concert with a four-piece band
— seven drum mics, four instrument
dls, two instrument mics and four vocal
mics — where I was running sound in
a hall that I know well. Although the
concert ended up being cancelled at
the last minute (due to the band leader
being taken seriously ill) so that I didn’t
get the chance to use the Axis system in
anger, I had done all the pre-production
for the concert on it — setting up
channels, monitor sends, output routing,
subgroup and VCA assignment, EQ,
dynamics, reverbs and delay, view and
mute groups, naming and so on. I also
created a new show so that I was ready
to capture mixes during the soundcheck,
and installed Master Fader on a second
iPad so that I would have both available
on the night. That whole process took
less than an hour, at which point I had
a setup that would go in and be up
and running in no time at all and that,
in theory at least, would only require
setting its input gains before I could start
the soundcheck.

In Use
Although the concert had vaporised,
I was supplied with an HDD, strapped to
the top of the DL32K, which was loaded
with recordings of a show and their
associated snapshots, so I was able to
mix recordings, taken pre-EQ, dynamics
and effects, of a rock band’s live gig
in the comfort of my own garage. The B
QUANTUM. Our fastest, leanest, meanest interface to date. Thunderbolt™ 2 gives you direct-to-DAW recording and monitoring with no DSP to slow you down. The result? Unmatched performance at high sampling rates with latency left in the dust.

This means you’re able to perform, monitor, and record with your virtual instruments, live. When tracking, monitor without creativity-burning latency, all in 24-bit 192 kHz quality with 120 dB of dynamic range.

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- MIDI and S/PDIF I/O
- Studio One 3 Artist, and a unique offer to upgrade to the Professional edition for 50% off
- Studio Magic plug-in suite, which includes incredible plug-ins by Lexicon®, Eventide®, SPL®, Maag®, and Arturia®

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input on every channel that by default carries the USB return from the multitrack direct-to-HDD recording can also be used to route a backup input source to the same channel as its prime, allowing you to switch over quickly if the main mic goes down.

Once I was mixing at the imaginary live gig, I began to appreciate Master Fader’s more practical aspects — there’s the RTA and spectrum analyser that can be assigned to any input or output; the 31-band graphic EQ on every output (and the ability to draw a curve on it with one finger); the presets available on every EQ, dynamics and effects processing; the ease with which routing and re-routing can be carried out; the View and Mute Groups; the mixer’s excellent sound quality, and much more besides. On the recording front it’s also worth remembering that the Axis system can also act as a 32-in/32-out USB audio interface and, for installation usage, can record and play back 32 channels to a remote computer via Dante.

To me, the DC16 with an iPad-loaded Smart Bridge really comes into its own running an active live mix with a high channel count. As the hardware interface for Master Fader, it adds ease of access, speed and precision, and enhances the practical functionality of the app. The learning curve isn’t particularly steep, requiring little more than a basic understanding of both the DC16 and Master Fader before you can begin to mix on it. Having an iPad touchscreen available both as an integral part of the hardware mixing experience and as a mobile wireless mix platform is something that I’ve come to rely on in the workflow of my own digital console. As with any digital console, how deeply you get into
the Axis system is entirely up to you, but if you want
to explore the full resources available in Master Fader, they
are there to be utilised in enhancing your mixes and the
experience of both band and audience alike.

**Summing Up**

Looking at Mackie’s Axis as a deconstructed digital mixer — control surface, software and mix engine — there is
much to like. As a total system it works extremely well,
giving you the possibility of connecting two DC16 control
surfaces to one DL32R via Dante, with Master Fader on
two to six iPads welding the whole assembly together and
offering, in addition, touchscreen control from either the
DC16’s Smart Bridge or at mobile locations in the hall. The
modular nature of the system means that hire companies
could offer solutions from a DL32R plus a couple of iPads
for smaller gigs, up to a full-blown six-iPad/dual-DC16
system for larger gigs — provided that no more than 32
inputs were required. For installations in multiple spaces,
houses of worship and the like, the Axis system could also
offer a very attractive solution.

Although the DL32R itself is priced quite attractively,
by the time you add in the cost of a DC16, a DL32R Dante
card and an iPad, Mackie’s Axis system ends up costing
significantly more than a good many of the competitively
priced, all-in-one consoles offering similar functionality
that are currently fighting it out between themselves in
a crowded sector of the market. Whilst none of those
consoles is really directly comparable to the Mackie,
the Axis system, just like its more directly comparable,
professional rivals, will find itself competing against several
of them.

Looking at more professional live-sound and install
applications, to me Mackie’s Axis system stands up well
against its more direct rivals. Its price, features, facilities
and flexibility should prove attractive in this area of
the market, where meeting the requirements of the
application, the needs and expectations of the customer
and the restrictions of the available budget is what
determines success.

Overall, the Mackie Axis Digital Live Mixing System
has a lot going for it and, for many professional live-sound
users, AV and hire companies, it offers a powerful and
flexible system that is capable of covering a wide range of
applications. Anyone
in the market for a
professional-level,
live digital mixing
solution should take
a very close look at
the Axis.
The designers tell us that as the systems are totally encrypted, and able to hop around the 2.4GHz spectrum as necessary, you could have up to eight different systems turned on in the same room, straight out of the box, all set to channel 1, and they still would not interfere with each other! This is particularly important for users of the filmmaker version of their kit — if, for example, a video crew is covering a trade show and they pass somebody else using the same system set to the same channel, then there should be no interference.

It seems that eight channels is the practical maximum for any system working in the 2.4GHz band, so you probably wouldn’t choose a 2.4GHz system for a Broadway or West End musical, but for a typical gigging band, guest speaker or lecturer, it provides a very practical solution. The lithium-ion battery allows around 10 hours of continuous use, or two AA alkaline batteries can be used instead to give a run time of more than six hours; the receiver shows the battery level.

**Rode RodeLink Performer Kit**

*Wireless Microphone System*

Rode’s new performance-oriented wireless set promises ease of use and robust performance.

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**Pros**

- Affordable.
- Easy to use.
- Reliable.
- Good-sounding mic.

**Cons**

- No carry case for the system.

**Summary**

A very capable yet inexpensive radio mic system that works in the licence-free band. Up to eight RodeLink systems can be used at the same time without interference.
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status to avoid unpleasant surprises.

Once up and running, the range can be up to 100 metres in ideal conditions, but in most practical situations, the receiver is likely to be much closer, often at the side of the stage. Walls and other obstacles will reduce the range as they would with any radio system.

**Taking The Mic**

The mic is longer than a typical corded mic, at 256mm, and it is quite wide, at 40mm, but it has a similar weight to a wired vocal microphone and it fits comfortably in the hand. Safely tucked away under the base of the mic is the power button, and this needs to be pressed and held for a few seconds to power up or down. A discrete LED shows when the mic is on, and this shines green when the mic and receiver are communicating with each other.

A lockable slide switch can be used to mute the mic. However, Rode have come up with something sneakily clever in that regard. The slide switch on the mic appears to work as it would on any other mic and has a screw to lock it in place if that’s what you prefer. However, if the performer does switch the mic off when it should be on, it can be reactivated by pressing the mute button on the receiver. Very useful.

Unscrewing the body sleeve reveals a small numeric window showing the channel to which the mic is set, a red pairing button, and a micro-USB port for charging. There’s no dedicated charger supplied with this kit but pretty much any USB power supply will do the job (I used a surge-protected power strip with two integral USB power outlets at gigs that can also charge the iPad I use for mixer control). Gold-plated terminals at the tail end of the mic interface with an optional ‘drop-in’ charging dock — an unusual option for a mic in this price range. Beneath the wire basket, which can be unscrewed for cleaning, is a hypercardioid-pattern back-electret capsule, identical to the one in the Rode M2 wired mic, specified with a 35Hz-20kHz frequency range and capable of handling SPLs of up to 140dB. This particular capsule is voiced to ensure clarity of both spoken word and sung vocals, and to my ears exhibits a more ‘open’ sound than a typical dynamic microphone.

A system dynamic range of 118dB is quoted along with a figure of below 4ms for latency (mic input to receiver output). All radio systems add some latency, which is negligible if using floor monitors, but if you add the latency of a digital mixing desk and a further transmitter to feed in-ear monitoring, the cumulative delay might be enough to perceive as slight ‘out-of-phase-ness’ when what’s heard in the earbuds combines with what you hear through natural bone conduction. This is simply a fact of digital life and something we have to learn to adapt to — but it is worth pointing out that some of the ‘big-name’ European radio mic systems costing rather more introduce considerably more latency than this system.

The RX-Desk receiver operates on the diversity principle, where two detachable external antenna screw onto rear-panel coaxial connectors. A screw-lock connector attaches the power cord from the 15V PSU to the receiver body so it isn’t going to come unplugged by accident. Both line- and mic-level XLR outputs are catered for by means of a mic/line switch, and the connection options are balanced XLR or unbalanced, line-only quarter-inch jack. Overall, the case measures just 208 x 37 x 166mm and weighs 792g.

Around the front there’s a window for showing the receive channel, battery condition, signal status, level and peak indicator. Also on the front panel are a red pairing button, a mute button, a button for stepping through the channels, and two buttons for controlling gain. The gain can be adjusted in 10dB steps from -20 to +20 using the +/- db buttons, and there’s a recessed power button over on the right. Apparently the digital signal path will only clip at a point just above the overload point of the capsule, which is already quite high, so getting the peak indicator to come on is a real challenge.

**Going Live**

Before listening to the mic I wanted to check out the pairing process to see if it was indeed as easy as claimed. First the red button on the receiver must be pressed, which causes the channel number in the display to blink. The Channel button can then be used to step through to the desired channel. All that remains is to slip back the mic body sleeve and press the red pairing button inside, at which point a flashing letter P appears in the mic display and pairing completes automatically, with the P being replaced by the chosen channel number. That all worked as it should. Prior to pairing or when first powering up the system, the small power LED at the end of the mic shows orange but turns green when the mic is paired and in communication with the receiver.

The metering section in the centre of the receiver display is arranged as three blocks, the lower two green and the top one yellow. Below the channel number is the transmitter battery status and to the right the radio link strength and the input gain setting.

Operationally this system sounds as clean as a wired mic, with the latency so low as to be negligible, no perceptible noise and none of the side effects of those horrible compander systems that were used in analogue radio mics to keep the noise down. If you go out of range the mic will mute politely rather than having everything dissolve in a sea of noise, but during my tests using typical live stage dimensions and a brick wall as an obstacle, I never had any issues with the mic fading in or out.

“**This system sounds as clean as a wired mic, with the latency so low as to be negligible...**”

**Alternatives**

Most of the European mic manufacturers offer their own 2.4GHz wireless systems nowadays, as do Line 6. To my tests I used a mini line-array PA system, which has near-hi-fi sound quality, and the RodeLink Performer sounded really solid and clear at all times. It also turned out to be virtually impossible to overload the mic. In fact the only trick that’s been missed is one I saw on a European system I checked out a couple of years ago, where turning off the receiver automatically turned off any transmitters paired with it to save battery life.

So to summarise, the RodeLink system provides an effective and affordable solution for anyone needing a radio mic. It sounds just like a Rode wired M2 mic and works reliably over a more than adequate range without fuss. It can be used in systems of up to eight channels without the hassle or the expense of a licence, and the TX/RX components are fully compatible with other RodeLink, Filmmaker and Newsshooter products.

\[EM|T|W|E|W|W|W|E|T]

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Morris Hayes - New Power Generation - Prince band
LIBRARY WORK
ALL ABOUT LIBRARY MUSIC
PART 8: AVOIDING DISASTER
Like any way to make a living, library music has its pitfalls. We help you look out for some of the biggest!

DAN GRAHAM

This series of articles about library music has, on the whole, been optimistic and encouraging for composers. And so it should be: many seasoned library writers are making five-figure royalty incomes after years of composing, and they are generally enjoying the creativity, freedom and flexibility of their careers. That said, a lot can go wrong along the way, and this month, we present a compendium of sorry tales of liars, cheats, idiots and AI overlords who will try to undermine your daily attempts to eke out a living, library music. This isn’t about fear-mongering, though: it’s to warn you against the worst of them. Let’s get off to a spicy start with cheats, liars and scams.

Composer takeaway: There’s not much you can do about alleged foreign royalty collection scams, but it can’t hurt to keep an extra eye on your friendly local society — who are, after all, exposed to huge flows of cash, close relationships with broadcasters and potential temptation. Beware of getting roped in to anything dodgy: someone may well convince you and yourselves that their bribe is a ‘sales commission’, and their royalty scam is a ‘business model’.

Composer Infringers

As a publisher, I’ve heard of new and inexperienced composers going directly to clients and undercutting high-quality publishers with low prices and bad music. To offset complaints about poor quality, some writers blatantly steal other, better writers’ work, combining loops from different tracks to cover their mischief. They sometimes get away with it, but as the power of tune-recognition software grows, they are starting to get their clients into trouble for copyright infringement.

I’ve also heard of composers writing the same music for two different publishers with whom they have supposedly exclusive agreements, just changing the main melody and calling it a new work. Perhaps it’s the naivety of new writers, but it can cause problems for everyone: remember that the recording copyright exists in all of the recorded music, including the background layers.

A similar story I heard involved a composer reusing the same melody for two different major publishers, a dodge which remained undetected until one of the tracks was used as the backing on a hit single! The composer then ended up caught between warring publishers, who

“TV producers will often forget to fill in the manual ‘cue sheets’ (forms which tell your collection society who wrote the music), partly because they are busy and partly because the TV networks have no great incentive to enforce it.”

Composer takeaway: in some parts of the world they don’t have the centuries of copyright laws that we’ve had in the West, and it will take time for the culture to change.

The Spanish ‘Wheel’

Not that the West is necessarily a beacon of integrity and transparency. In 2017, 18 people were arrested in Spain as part of a police investigation into an alleged scam known as ‘the wheel’ (la rueda). Employees of Spain’s SGAE Performing Rights Society were accused of creating low-quality musical arrangements of public-domain works, while TV stations were listed as publishers; the stations then repeatedly broadcast this music late at night, clocking up millions of Euros in broadcast royalties. The police also accused TV company employees of taking “financial rewards” for helping to favour this music for airtime. This follows years of notoriety, including one former SGAE executive Pedro Farré being jailed after being arrested with eight others for misappropriation of funds in 2011. On his release he published a colourful book about splurging his SGAE expense account on prostitutes, drugs, while complaining about being the only one to be found guilty.

Composer takeaway: There’s not much you can do about alleged foreign royalty collection scams, but it can’t hurt to keep an extra eye on your friendly local society — who are, after all, exposed to huge flows of cash, close relationships with broadcasters and potential temptation. Beware of getting roped in to anything dodgy: someone may well convince you and yourselves that their bribe is a ‘sales commission’, and their royalty scam is a ‘business model’.

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and a couple of

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A tune-recognition software

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scams, so be wary.

Avoid lazy composing

shortcuts. Don’t pass off other

people’s work as yours, or reuse ideas in

different tracks. Tune-recognition software

will catch you and ruin your career.

Get professional indemnity insurance

(Errors and Omissions Insurance in the

USA). Some people mistakenly think that

by operating as a limited company you

can remove yourself from personal legal

and financial risks, but this is not true.

You are personally liable for copyright

INFRINGEMENTS, and the more music you

write, the greater your risk of being

accused, even if the accusation is unfair.

This insurance is essential for library

composers: it will pay out damages to

everyone affected if you are found guilty

and leave your life and home intact.

Chase the cheats. If you catch companies

or composers using your music without

permission, you can write take-down

emails, embarrass them on social media

and use the take-down services of

YouTube and Facebook without any cost.

Clarify fees and deals up front. Clients can

be unrealistic or misinformed about going

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Check your performing rights organisation.

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Peril Avoidance Checklist

• Avoid bribes and scams. Royalty flows run into the

  millions, and where there are small concentrations of

  people handling them (publishers, royalty collection

  societies and TV networks), there is a susceptibility to

  offering ‘commissions’ and ‘ingenious business models’

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• Avoid lazy composing shortcuts. Don’t pass off other

  people’s work as yours, or reuse ideas in different tracks. Tune-recognition software will

  catch you and ruin your career.

• Get professional indemnity insurance (Errors and Omissions Insurance in the USA). Some people mistakenly think that by operating as a limited company you can remove yourself from personal legal and financial risks, but this is not true. You are personally liable for copyright infringements, and the more music you write, the greater your risk of being accused, even if the accusation is unfair. This insurance is essential for library composers: it will pay out damages to everyone affected if you are found guilty and leave your life and home intact.

• Chase the cheats. If you catch companies or composers using your music without permission, you can write take-down emails, embarrass them on social media and use the take-down services of YouTube and Facebook without any cost.

• Clarify fees and deals up front. Clients can be unrealistic or misinformed about going rates, and lowball publishers can have terrible deals up their sleeves.

• Check your performing rights organisation.

Look on their music searches: are your tracks there and with the correct details? Email them and ask how they are checking that TV producers are filling in their cue sheets correctly. How co-operatively will they work with reports from Tunesat, who monitor unlicensed music usage?

• Earn money from YouTube infringement. If amateur video makers are racking up millions of views while using your music without permission, make sure you or your publisher sign up with a company like AdRev who could collect significant advertising income for you.

• Prepare for technical meltdowns. Have your plan ready, with a laptop ready to spring into action. Purchase a new Steinberg USB key every two years to stay in-warranty. Purchase Zero Downtime protection for your iLok dongle. Back up important data on a cloud service like Dropbox. Have at least two recent hard-drive back-ups, including one in a different physical location.

• Avoid bad publishers. Do your best to avoid publishers who are slow, sloppy, offer bad deals and give unhelpful music feedback.

both blamed him. Luckily they agreed a royalty split, but the writer’s reputation was damaged.

Sometimes theft and misrepresentation is more blatant, as composer Deryn Cullen found out. “My husband and I have been composing and recording music in partnership since 2006. Most of our material is destined for libraries, and some we have released, both on our own and through a record label. Until recently we never dreamed that anyone would have the audacity to appropriate, retitle and release our music as their own but we discovered an act of theft, fraud and misrepresentation when we were adding some of our earliest cues to a non-exclusive library, and a couple of them were flagged by their content ID (tune-recognition software) system. After sending sufficient proof of ownership, the library released the information they had against the tracks and we were horrified to discover that someone in North America had released an entire 16-track album consisting entirely of music we had written and recorded between 2006 and 2013. Further investigation revealed that we were not the only artists whose music he had claimed as his own. We are still in the process of disputing and having the release removed.”

Elsewhere, an odd story circulated a few years ago of a young ‘epic music composer’ showing off his music alongside photos of his incredible film score awards. Except none of the awards were his, and none of the music was his. He didn’t try to profit from this deception, so who knows if he was a fantasist, hoaxter or troll, but he recently resurfaced doing exactly the same thing.

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When the eagle meets the dragon... it’s hard to get paid for the trailer music.

Writer Marie-Anne Fischer relates one odd tale of a client who wasn’t so much a cheapskate as a freeskate. “A Canadian composer and the library, I worked on an ending for him which was to be used on a trailer for the movie ‘Dragon Blade’. After asking permission from the company who made the trailer, and asked them to pay for a standard licence. They replied offering us $50, explaining that this was what they pay local composers.

Fifty bucks for something this big was an insult to us and the composer, who gets 50 percent of the fees. A decision was therefore made to fight back, with lawyers chasing production and distribution companies in multiple countries. We also got YouTube and Facebook to take down the video from Jackie Chan’s pages and ran a spirited social media campaign, all the time reaching out to offer them a licence agreement at a reasonable rate, but to no avail.

A few library industry insiders told us that no Western company has ever won a copyright case in China, and we were wasting our time — but out of the blue we got a $25,000 payment and a long and bizarre worded letter of apology from someone, although we didn’t actually know who he was. He assured us that he would use our music as a first choice in future trailers, although we’re still waiting for those!

Composer takeaway: You can’t prevent people trying to rip you off, but if it’s a big-budget film with famous actors and international partners, it seems that you can embarrass someone into coughing up the dough with enough effort. For smaller productions it’s a mixed picture in China and many other countries. We know lots of good Chinese companies and there is progress, but there isn’t currently a strong legal pressure on less reputable companies to bother paying foreign music companies for using their music.

On The Receiving End

The more music you write, the greater your chances of becoming a target of copyright infringement claims. Famous legal cases crop up regularly in the press and highlight the vagueness of what constitutes a musical copyright infringement. Robin Thicke and Pharrell Williams famously had to pay the estate of Marvin Gaye $7.3m in 2015 over supposed similarities between ‘Blurred Lines’ and Gaye’s ‘Got To Give It Up’, which were by no means universally acknowledged. It’s a reminder that legal outcomes are decided by juries and judges who have no expert knowledge, and can be swayed by unfortunate quotes from writers about what inspired them and the testimony of expert witnesses.

Back in the library music world, a current legal case has seen the small family-run Australian company who
represent our music being dragged to court by Eminem. This is over a track called ‘Eminem-esque’, which was written in the US, published by a US library, and only represented in Australia and New Zealand by these sub-publishers. The infringement allegations arose after it was used on a TV campaign, by a New Zealand political party. The press had a field day laughing at the idea of New Zealand politicians taking on Eminem in court, but the upsetting reality is that a small company unwittingly stumbled into liability threats through something they had very little control over.

The case isn’t settled at the time of writing, but it’s a sobering reminder that you never can predict when you might end up in court. As the writer, you have the ultimate liability in cases of copyright infringement, because you sign a publishing agreement guaranteeing that you own the copyright and accept all of the penalties if you don’t.

Composer takeaway: The more music you write, the greater the chance you’ll be the target of an accusation, fair or unfair. As before, your best chance of avoiding financial ruin is to take out professional indemnity insurance cover (Errors and Omissions Insurance in the USA).

YouTube Infringers: Your Best Mates?

As a side note, the bane of composer’s lives a few years ago was their music being used on amateur YouTube videos without permission. Now that YouTube is a big seller of advertising, however, good money can be earned for unapproved uses via advertising revenue. Companies like AdRev in the USA help publishers and composers to recover income that can sometimes be significant; for example, our company earns $1000 per month and rising from unlicensed uses of our music on YouTube.

Composer takeaway: Once a threat to contain, YouTube infringement has become a gravy train to milk.

Rip-off Publishers

A whole nest of vipers is the world of low-rent publishers who will take your music and give you little in return. Much of this ground has been covered in Part 2 of this series, where we talked about the different types of library companies, but, to repeat some of that advice, ask other composers for recommendations, and don’t dismiss what sounds like a bad deal if you know that a company are making good money for their writers.

US composer Vincent Varco gives an example of one publisher who offered a buy-out figure (with no future royalties) of $2000 for 300 cues. Imagine the rate you’d have to write to make that worth your while! He also tells us of publishers who have deals with TV production companies where the publisher gives the TV company all of their future royalties and makes up the loss by taking half of the writer’s share. Arguably the real problem here is the TV company forcing publishers to give up their income, but unless the library-music world has slumped and this is the latest deal you can get, I’d avoid this if you can help it.

Another US composer, Paul Biondi, tells us of a similar experience, perhaps with the same company: “They said that they found themselves having to give away half or all of their publishing to the networks, so taking 50 percent of composing assured them that they would be making at least 25 percent. More disappointing to me, the library owner — a known composer — said he would be using his composing skills and knowledge in making cut-downs, which warrants half of the writer’s share.”

Composer takeaway: Unless you have very strong evidence that it could earn you great money, only give away your writer share to co-writers or other people you’d like to reward (like performers or producers), not publishers or TV networks.

Incompetence

It’s not only swindling rogues who stand in your way. Another huge enemy is incompetence: publishers, collection societies and clients wasting your time and leaving you out of pocket by doing their jobs badly. For example, publishers who give incomprehensible, contradictory briefs and change requests to their writers are massive time-wasters. As one writer tells us: “Despite rarely having tracks rejected, the publisher’s new project manager rejected track after track, constantly asking for rewrites with incomprehensible and contradictory instructions. Halfway through I wondered if she was an idiot and started resubmitting old versions that she’d already rejected. Every single time it was ‘Wow, this is much better, thank you!’ and things ran much more smoothly after that.”

Another writer, Jamie Salisbury, relates the time his track was rejected and over and over, with the mix being blamed. So, he asked a friend to do a new mix.

The publisher initially approved it but then left it off the album, this time blaming the arrangement. He had the last laugh, however: “A couple of months later it became the lead track on a KPM trailers album, and has become my second biggest-earning track, including extensive uses during the Cricket World Cup.”

Writer Guy Rowland has a story involving a famous singer he’s calling ‘Troy Illinois’. Guy was asked to write music exactly in ‘Troy’s’ style — for Troy’s own TV show! Bizarrely, Troy’s own music couldn’t be cleared for this use, but Guy’s every attempt to imitate his style was rejected. “Each time I got the same disappointing feedback from one of the 90 producers working on the thing: ‘It’s just not Troy Illinois enough.’ Finally, out of sheer unprofessional exasperation, I did something you should never, ever do. As a test, I actually submitted an edit of an actual 100-percent-genuine Troy Illinois song itself. No doubt you’re ahead of me, and already guessed the feedback I got. ‘It’s just not Troy Illinois enough.’ That, ladies and gentlemen, is the time to bail.”

Composer takeaway: Try to avoid working for publishers who give incompetent writer feedback. A politically perilous alternative is to complain to the boss about your bad experience. That approach might work if they trust you and have doubts about their staff, but it could easily backfire if they think you’re being difficult.

Errors & Oversights

Human beings are a mess: sloppy, slapdash, too busy to do things properly, forgetful, sleep-deprived, unmotivated, lazy, ill, hangover and working for companies who think they are fantastic because their old back catalogue is making
Significantly less than they are owed. The US company Tunesat are helping to bring this to an end with their tune-recognition software, which monitors broadcasts around the world and gives you reports of music matches, and you can get started with a popular free version monitoring a small number of tracks. AdRev are also helping writers and publishers to catch infringements using YouTube’s similar ContentID software and then raise advertising revenue.

Composer takeaway:
Check the accuracy of your writer and song information on your country’s performing rights organisation’s music search engine and whether your music is available as it should be on your publisher’s web site and those of their sub-publishers (agents).

As a professional composer, library or otherwise, you have to be prepared for technical failures. Software can decide not to work, hard drives can fail, piracy prevention dongles can shut you out of your software and entire computers can die, leaving you out of action just as deadlines loom.

As soon as you can possibly afford it, therefore, you need a plan that will keep you running in the event of any possible fault. To cope with drive failures, you need cloud backups such as Dropbox, as well as on-site and off-site drive backups (you could lose everything in a fire, flood or theft if you keep all your backups in one place, so keep an up-to-date hard drive somewhere else!). To guard against computer failures, you need a backup of your system drive and a spare computer, such as a laptop that you can draft in at short notice to finish a job while your main workstation is being repaired. For iLok dongles, purchase Zero Downtime insurance, and with Steinberg USB keys buy a new one every two years: they are only guaranteed for two years and Steinberg may not replace a faulty one containing all your licences if it’s older than that.

Eventually these nightmare scenarios will happen to you, so you can either be sane and plan ahead, or learn the hard way from a disaster — after which time you will do all of these things anyway. Composer Marie-Anne Fischer has a typical story: “I had a disaster with my computer with all parts breaking at once whilst under pressure to meet a demanding library deadline. Apple said my computer was too old and out of support... that it was best to sell it off in parts. So, I had to invest heavily into a new computer. Having a good back-up plan is very important!”

Technical Meltdowns

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Most of my agents (international library sub-publishers), are great but I am constantly having to beg and browbeat one of them into releasing albums that they’ve been sitting on for up to a year — that’s a year of lost earnings for the writers. We once discovered that a foreign sub-publisher had overlooked some albums and never released them. We found that another was a year behind registering the music with their collection society. All of these delays lose money for our writers.

Meanwhile, TV producers will often forget to fill in the manual ‘cue sheets’ (forms that tell your collection society who wrote the music), partly because they are busy and partly because the TV networks have no great incentive to enforce it. Collection societies vary around the world in how organised they are and how successfully they enforce accurate royalty reporting from TV networks.

It’s hard to guess at the size of these problems, but since huge parts of the international music royalty system require the manual inputting and copying of data and these problems are familiar to anyone who looks closely into it, we can only assume it is rife and all writers are earning significantly less than they are owed. The US company Tunesat are helping to bring this to an end with their tune-recognition software, which monitors broadcasts around the world and gives you reports of music matches, and you can get started with a popular free version monitoring a small number of tracks. AdRev are also helping writers and publishers to catch infringements using YouTube’s similar ContentID software and then raise advertising revenue.

Composer takeaway: Check the accuracy of your writer and song information on your country’s performing rights organisation’s music search engine and whether your music is available as it should be on your publisher’s web site and those of their sub-publishers (agents).
around the world. Get a free account with Tunetastic and never assume that everyone knows what they’re doing. They really don’t.

**Bad Luck**

Sometimes things just don’t work out. For example, as a writer, my main library publisher had a change of personnel. My enthusiastic and inspiring mentor was replaced by someone who seemed to hate me, and the jobs soon dried up. Another misfortune saw my albums delayed for over a year when the company found themselves with too much music and decided to pause releases until they’d cleared their giant backlog.

And then, publishers have their misfortunes. They could get sued or make mistakes and go bust, putting your income in jeopardy. Or they could be bought out by a company that fails to promote your music, or have a few months of low sales, causing them to cut costs and leave your album on the shelf for a few months.

**Composer takeaway:**

Some bad luck is inevitable, but you can avoid the worst of it by working for more than one publisher.

**Outside Your Control**

At the higher levels of the library industry, various battles rumble on, all of which could affect your income. That includes battles between performing rights societies (PROs) and streaming services like YouTube to secure higher royalties; between one country’s PRO and others (such as disputes over who should pay certain taxes); and between PROs and broadcasters, trying to secure better royalty rates or prevent them dropping. Then there are governments taking PROs to court for acting like price-fixing cartels, PRO board meetings that decide to award library music lower rates without asking anyone, and PROs lobbying governments to crackdown on piracy. All of this wrangling and more can cause short-term hiccups or long-term changes in your income levels.

More globally still, there are other common fears. One is that rising competition from new composers and budget publishers is creating a ‘race to the bottom’, where fees are being driven down to the point that no-one will be able to make a living any more. I’ve been hearing this for 15 years and, touch wood, it hasn’t affected the high-end market yet. Another popular note of doom is the rise of AI making automatically composed music, licensed for pennies. This might be a problem one day, but so far, all the algorithmically created music I’ve heard has been laughable drivel.

And, of course, library music could be wiped out by a worldwide economic disaster. Then there are governments secure better royalty rates or prevent them falling; between PROs and broadcasters, trying to stay out of the clink too.

**Composer takeaway:**

What’s The Worst That Could Happen?

To recap, the worst that could happen professionally is that you could end up bankrupt or even in prison. Pretty bad, but having the right insurance should help you avoid bankruptcy, and if you avoid stealing other people’s work and various royalty scams and bribes, you should be able to stay out of the clink too. For these risks and all the others we’ve looked at, the important thing to remember is that most pitfalls can be avoided, especially if you take the precautions in the ‘Peril Avoidance Checklist’ box.

Most composers can just get on with writing music knowing that the worst disasters are unlikely. If you spend your whole time scanning YouTube videos for piracy and rifling through your royalty reports and collection society web sites, obsessively looking for mistakes, you will indeed uncover wrongs and probably raise extra money — but there has to be a balance. If your job has become 20 percent composer and 80 percent royalty inspector, is that really what you want? Couldn’t you just let some of it go and make more money by writing more music?

The answer is moderation: if you’re too lax, people will rip you off accidentally or intentionally, and if you’re too neurotic, you’ll lose money from stressing over the details instead of writing new music.

A reasonable compromise is therefore to spend a few hours every so often drilling down and trying to catch errors without letting it dominate you. Just keep writing lots of great music, and when you’re rich, hire a full-time royalty sleuth to recover your stolen millions. Until then, as Jez from Peep Show would say, perhaps you should just sign and recline — most of the time.

**Composer takeaway:**

Much is in the lap of the gods, but as far as rights issues go, you can turn up to PRO annual meetings or put yourself forward for a role on their committees and action groups if you feel strongly enough.
Teenage Engineering
Pocket Operator PO-32 Tonic
Miniature Drum Synth

Teenage Engineering take the Pocket Operator game to the next level.

SIMON SHERBOURNE

The PO-32 is the latest and most ambitious Pocket Operator in Teenage Engineering’s quirky range of handheld instruments. This one is a programmable synth-based drum machine with a pattern sequencer, performance effects, and parameter lock automation. The ‘Tonic’ moniker is borrowed from Microtonic, the Sonic Charge plug-in whose drum synth engine has been committed to silicon on this device. It’s this plug-in (bought separately) that provides the programming interface for creating your own PO-32 sounds. These can then be transferred to the PO-32 via an audio data burst... you should never expect the ordinary from Teenage Engineering.

Everyone loves a good unboxing, but nothing comes close to opening up a Pocket Operator. The beautiful gold card package protects the main part of the unit, while the integral hook (not something you find on the spec list of much gear) sticks out ready for hanging on a peg. A perforated ‘zip’ in the card is used to unpeel the device in a process my wife described as “an event”; she was dismayed I’d not videoed it and put it on Instagram.

Skin Deep

The Pocket Operator looks like a calculator that’s lost its outer case. If it was a cartoon character it would blush and try to cover its midriff. It’s a slice of circuit board with a surface-mounted screen and a 5x5 grid of controls, two of which are small pots while the rest are buttons. On the rear are clips for attaching two AAA batteries, and a simple metal bar that angles the unit up when on a tabletop, or folds away if held in your hand. At the top there’s a tiny mic for data input, and two mini-jack audio ports providing audio output, data transfer and sync connectivity.

At first it just feels wrong to be holding what feels like a half-assembled piece of electronics, but there’s actually very little that’s exposed. The back is mostly smooth bare board, with just the soldered pegs from the knobs poking through. Cleverly, all the delicate components are safely stowed away behind the screen. My review unit survived some pretty harsh treatment, being chucked in a bag in various planes, trains and automobiles, and put through its paces by two under-10s. If in doubt, though, there’s an optional case which provides both an outer shell and nicer button caps.

Basic Operation

The PO-32 is not the most intuitive device to get started
with — you definitely need the few hints that are printed on the box and on the back of the unit; even simple tasks like changing the volume aren’t straightforward. The screen doesn’t help much: while there are a few small readouts on it that confirm what you’re doing, it’s largely devoted to a cartoon animation that plays along as sounds are triggered, reminiscent of ’80s handheld games.

Sixteen of the buttons make up a 4x4 grid that’s used for playing sounds, setting sequence gates and entering values. The remaining few buttons around the edge are used for changing global settings and modes. One of these toggles whether you’re in Write or Play mode. In play, you can freely trigger sounds from the 4x4 grid. In Write mode, the grid represents the 16 steps of the current pattern, with one sound shown at a time. Steps can be added or removed by tapping the buttons. You can also record patterns in real time (quantised to the grid) if you hold the Write button, although this is a bit fiddly.

The 16 sounds in a Tonic bank share four playback voices. These are not distributed dynamically, each of the four columns of the 4x4 grid is allocated a single voice. When creating a bank it makes sense to avoid placing sounds with a long decay in the same group as things like hi-hats that will cut them off. In fact you can use this architecture creatively. For a start each column is effectively a choke group. Better still, I used this during playback as a way of dynamically muting sounds and creating breakdowns: if you hold down a button from a group it silences all other triggers until you release.

**Teenage Kicks**

Effects are handled in a unique and wonderful way on the PO-32. They’re not so much traditional effects as momentary performance variations that are applied in real time within the synth engine. During playback you simply hold the FX button and then any of the main 16 grid buttons. The effects include things like stutter, reverse, slowdown, shuffle and saturation. If you’re in Write mode anything you do with the effects is captured as automation in the pattern. It’s really effective, letting you add lots of interesting variation.

The two knobs are primarily used to adjust the sounds. Knob 1 always controls pitch, while Knob 2 is a morph between two different sound states. These two states are full presets from the Microtonic engine, so could be completely different sounds. Typically though they are variations of a patch. For example, you could set up a kick sound for the starting point, then save the ‘B’ state with extended release, distortion, etc. These will then be
PO-32 is a hardware version of Sonic Charge’s Microtonic drum synth. The software or plug-in can be used to create and load sounds into the device, including its Pocket siblings.

Simple tempo matching and beat alignment are possible by feeding a click into the audio input. It can also generate a click from one channel of its audio output port. Different Sync modes are available that determine whether the PO-32 is to be a slave, master or both, and how the shared audio/sync ports are configured. Nothing is sacrificed sonically here as the output is mono anyway, but you’d need splitter cables in some scenarios.

**Conclusion**

It would be easy to dismiss the PO-32 as a toy or a ‘collectable’ based on its size, bare board design and tinny built-in speaker. If you dig a bit deeper, and especially if you plug it into some speakers, you’ll find a surprisingly capable drum machine, based on a deep synth engine. While the sounds are not fully programmable from the unit alone, you can get a lot of variation from the sound morphing, automation and effects. The effects in particular make the PO-32 useful, letting you drop in momentary fills, variations and little flourishes of sonic interest on the fly.

If you want to go deeper, you can add the Microtonic software (£99) and gain complete control over the PO-32’s sound engine. And it’s genuinely pocket sized if you like to keep something more tactile than a phone with you for some beat doodling — just be prepared to be asked what it is every time you get it out.

“**It’s genuinely pocket sized if you like to keep something more tactile than a phone, with you for some beat doodling.”**

Transfer is also possible between two PO-32s, again either from speaker to mic, or via an audio cable. It’s also possible to load (and back up) sounds from a data burst recorded as audio, allowing you to share via YouTube, for example, and get access to different banks without splashing out on the software.

**Jam Sync**

The PO-32 can store 16 patterns at a time. These can be temporarily chained into longer loops by holding the Pattern button and tapping out a sequence. Automation of sound parameters can be added to a pattern, either as continuous envelopes by holding Write while adjusting the knobs, or for a single step by holding the step’s button down while adjusting. This, along with the effects gives a lot of scope for interesting movement within a pattern. Rather importantly for a drum machine, the PO-32 can sync to other gear.
History, Refined.

**Trident 88**
16 channel
Starting at $24,999

The Trident 88 is a classically styled, in-line, eight-buss analog console and an advancement in lineage of the revered Series 80 mixing desks. Designed with manufacturability in mind, the 88 is artfully constructed for ease of use and workhorse reliability. Available in stock configurations of 8, 16, 24, 32 and 40 channels, the model 88 is sized to fit the modern, high end facility - whether private, institutional, or professional.

TRIDENTAUDIODEVELOPMENTS.COM
Vertigo Sound VSE-2

Dual-channel Gyrator EQ

Vertigo’s classy new equaliser draws on some intriguing design ideas from the 1970s.

Why Gyrate?

First proposed in 1948 by Bernard Tellegen (as a hypothetical fifth linear element after the resistor, capacitor, inductor and ideal transformer), a gyrator is essentially an active two-terminal device that inverts the current-voltage characteristic of an electrical component. Thus a gyrator can transform a capacitor into an inductor, so it can be used to replace the inductor in an LRC (inductor, capacitor and resistor) filter circuit. In fact, gyrators are sometimes referred to as ‘simulated inductors’, but that can be undeservedly faint praise because a gyrator is actually capable of producing more desirable results than the coil of wire wound around a metal core that makes up an inductor.

One of the attractions of an LRC filter is that its Q increases as the level of cut or boost are increased. Since Q and bandwidth have an inverse relationship, the bandwidth narrows as the amount of cut or boost is increased — the cut or boost becomes more focussed on the centre frequency, reducing the effect on adjacent frequencies. This characteristic, often referred to as ‘proportional Q’, applies to LRC filters whether they employ a real inductor or a gyrator.

The solid-state gyrator, then, has the potential to solve the inductor’s major drawbacks, namely its size and weight, its susceptibility to stray electromagnetic fields, and its cost. However, as with transformers, the non-linearities inherent in the materials and construction of a physical inductor can play a major role in the sonic character of an EQ, and removing these via the use of a gyrator can, in some cases, mean throwing the proverbial baby out with the bath water...

Outside In

As with all the Vertigo Sound products I’ve encountered, the VSE-2 is superbly built. Its hefty 3U 19-inch rackmount steel chassis extends rearwards quite significantly, and it’s fronted by a thick, beautifully finished red panel with silvered detailing and legends. The VSE-2’s dual-mono operation is reflected in the two identical functional areas that together occupy almost the entirety of the fascia, leaving just enough room for the orange and silver Vertigo logo on the left and for the unit’s main power switch and indicator light on the right.

The control layouts for each channel are identical. An upper row of three large rotary switches set the levels of cut or boost in the low-frequency, mid-frequency and high-frequency bands. Other than the maximum ±8dB positions, no level information is given — although the
The manual tells us that the intermediate steps are ±1, ±2, ±3.5 and ±5.5 dB. A corresponding lower row of rotary switches selects the centre frequency in each band. In a playful visual touch, the 18 possible frequencies (see the ‘Freq Scene’ box) are laid out in the form of a sinusoidal wave that seemed to me to make it easier to recognise what frequencies were in use in each band. The initials AIR stand for ‘All Impedance Resonance’, and this final position, centred on 16kHz, allows you to add a sense of ‘air’ to proceedings.

The upper-right corner of each area is occupied by a switchable high-pass filter that’s continuously variable between 10 and 400 Hz, and beneath this sits its indicator LED and in/out toggle switch. In the bottom right-hand corner a red clip LED lies to the left of a similar switch-and-indicator combination that toggles the EQ in and out of hard bypass.

The XLR connectors of the transformer-balanced inputs and electronically balanced outputs lie on the rear panel, along with the fused IEC mains socket. The review unit’s 230V AC operating voltage was printed on the rear panel, and I’ve found no indication of any other available alternative, even though removing the top cover reveals that selecting 110V operation is simply a matter of flipping an internal switch (I imagine that will be the default setting for units sold in the USA and Canada, but it would be worth checking!).

Removing the top cover reveals a similarly high build quality inside, with top-flight, full-size components and evidence of exceptional attention to detail. Each channel’s audio circuitry is carried on separate PCBs: a single I/O board, a stacked twin-board assembly that contains the gyrator itself, and a much smaller board, on which sit the high-pass filter frequency control and its connector header. A second small header board sits on the back of the channel’s XLR connectors. Finally, a seventh board, separated from the audio boards by as large a distance as possible inside the case, is where you’ll find the VSE-2’s toroidal transformer-based power supply. Multi-way cables, crossing at precise right-angles, form the board-to-board interconnects.

The major physical features of the two channel I/O boards are their Jensen JT-11P-1TB line-level input balancing transformers, Vertigo’s proprietary 1976 package of twin, discrete, bipolar op-amps, and the 1646 ICs that drive and electronically balance the channel outputs. The two-storey gyrator boards carry the resistor ladders of the three gain controls on their upper floors, whilst the lower floors are home not only to the bands’ frequency selectors, but also to Vertigo’s 1972 proprietary discrete triple-gyrator package. The 1972 triple gyrator utilises the discrete twin op-amp from the 1976, one side of which handles cut duties, whilst the other takes care of boosts. The 1972’s three gyrator circuits each handle one frequency band and are connected in parallel to each other, in order to minimise phase differences.

In the VSE-2, the maximum gain range for each frequency band is restricted to ±8dB, in order to reduce the overlap between adjacent frequencies that occurs as the filter’s bandwidth widens at low levels of cut or boost. Had the gain range been extended to, say, ±20dB, this would have made the equalisation’s frequency response much more controllable.

“Maintaining a proportional-Q peaking response at 40Hz and 16kHz gives, in practice, virtually the same audible result as a shelving EQ at those frequencies.”

The HF band includes an AIR setting at 16kHz, which appears to function a little more like a harmonic enhancer than an EQ.

The VSE-2 is a superb equaliser that delivers a stunning level of performance at a price point commensurate with its capabilities.
±15dB, the change in filter bandwidth and the area of overlap with adjacent bands at lower boosts would have been correspondingly greater.

The VSE-2’s features and quoted specifications add up to a very impressive package overall: a frequency response of 10Hz-20kHz (+3dB); 122dB of dynamic range; a signal-to-noise ratio of 105dB (20Hz-20kHz unweighted RMS @ +6dBu); a noise figure of -99dBu of transformer-balanced inputs — the VSE-2 is one of those units where your best meters are your ears. In addition to the expected sense of warm richness you’d expect from a tube-like distortion device, it’s just a matter of dialling in the levels of boost and cut that feel intuitive. Once you’ve identified the areas that need to be worked on, it’s just a matter of dialling in the levels of boost and cut that feel right. Simply because it sounds so good, I did catch myself applying more boost than was actually required over the 5-10 kHz range. To be fair, the VSE-2’s manual warned that I might do this, so I followed its suggestion of increasing the level of the AIR mode rather than boosting 10kHz — and in most cases this produced subjectively better results. From the manual’s description of the AIR mode as “pushing artifacts and higher distortion products rather than primarily processing the original signal”, it seems to me that what’s going on here is more akin to harmonic enhancement than a simple 16kHz EQ band.

Speaking of frequencies, unlike other two-thirds-octave (16-band) equalisers of my acquaintance, the VSE-2 deviates in places from the geometric ISO 266 series of preferred frequencies (see box). Taking out the highest (20kHz) and the two lowest (20Hz and 40Hz) ISO frequencies results in six (rather than five) frequencies per band, all of which correspond exactly, or at least reasonably well, with their equivalents on the ISO third-octave preferred list. I was very happy with the frequencies chosen by Vertigo Sound, and I especially appreciated the increased frequency choices in the 40Hz to 2kHz range compared with its alternative equalisers. Here’s how the ISO 266 third- and two-thirds-octave frequencies compare with those of the VSE-2. (All figures are in Hertz, with ‘k’ denoting kilohertz).

ISO 266: 2/3 Octave
Low: 20, 31.5, 50, 80, 125, 200, 315, 500, 800, 1.25k, 2k
Mid: 3.15k, 5, 8k, 12.5k, 20k
High: 31.5, 50, 80, 125, 200, 2.5k, 3.15k, 4k, 5k, 6.3k, 8k, 10k, 12.5k, 16k, 20k

ISO 266: 1/3 Octave
Low: 12.5, 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250
Mid: 315, 480, 640, 1.0k, 1.3k, 2k
High: 2.5k, 3.5k, 5.0k, 8.0k, 10.0k plus A.I.R (16k)

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Running a wide range of material through the VSE-2 revealed a clear and extremely articulate overall sound that, by simply increasing input levels, can be persuaded to reveal more and more of its potential tube-like character. Spending time experimenting with input levels, EQ boost/cut and the cut-off frequency of the high-pass filter to find the EQ profile and the level of coloration that suited the track best was more than worthwhile. Making a note of these settings gave me a reference point I could return to if I later became a little heavy-handed with EQ.

The VSE-2’s EQ is more gentle and smooth than in-your-face. In fact it just doesn’t do hard and aggressive, no matter what level is being forced into it. There’s an extremely ‘organic’ feel to the sound, and a seeming ability to enhance the area that’s being cut. It’s an EQ that starts by applying quite a broad brush to the audio spectrum you’re boosting, while somehow bringing increased clarity to the area that’s being cut. It’s an EQ that sharpens up and focuses on the chosen frequencies as boost and cut levels increase, it’s never going to reach the precision of a multi-band parametric EQ or a 31-band, constant-Q graphic equaliser.

All this makes using the VSE-2 extremely intuitive. Once you’ve identified the areas that need to be worked on, it’s just a matter of dialling in the levels of cut and boost that feel right. Simply because it sounds so good, I did catch myself applying more boost than was actually required over the 5-10 kHz range. To be fair, the VSE-2’s manual warned that I might do this, so I followed its suggestion of increasing the level of the AIR mode rather than boosting 10kHz — and in most cases this produced subjectively better results. From the manual’s description of the AIR mode as “pushing artifacts and higher distortion products rather than primarily processing the original signal”, it seems to me that what’s going on here is more akin to harmonic enhancement than a simple 16kHz EQ band.

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ISO 266: 2/3 Octave
Low: 20, 31.5, 50, 80, 125, 200, 315, 500, 800, 1.25k, 2k
Mid: 3.15k, 5, 8k, 12.5k, 20k
High: 31.5, 50, 80, 125, 200, 2.5k, 3.15k, 4k, 5k, 6.3k, 8k, 10k, 12.5k, 16k, 20k

ISO 266: 1/3 Octave
Low: 12.5, 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250
Mid: 315, 480, 640, 1.0k, 1.3k, 2k
High: 2.5k, 3.5k, 5.0k, 8.0k, 10.0k plus A.I.R (16k)

Vertigo VSE-2 12.17 layout.indd   130
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Alternatives
At this price and performance level, you’re probably also going to be looking at alternatives such as Chandler’s TG12345 Curve Bender, Great River’s MAG-2MV, GML’s 8200 five-band stereo parametric EQ, Manley’s Massive Passive and the Pultec EQM-153 mastering EQ.
the ISO-based lineup available on my elderly, and recently retired, 16-band stereo graphic EQ.

Another interesting aspect of the VSE-2 is the absence of a shelving EQ at either end of the frequency spectrum. Vertigo Sound consider that shelving EQ can add too much energy to the signal thus reducing system headroom, and that maintaining a proportional-Q peaking response at 40Hz and 16kHz gives, in practice, virtually the same audible result as a shelving EQ at those frequencies, since the ‘furthest away’ side of their respective bell curves sit in the less-audible areas of the audio spectrum.

Conclusion

Without doubt, the Vertigo Sound VSE-2 is a superb equaliser. It looks beautiful and, used judiciously, it’s capable of delivering results of the highest quality, and of adding shape, warmth, depth and a quite seductive sheen to a track, bus or mix. Although Vertigo Sound state that the standard VSE-2 is suited to mastering duties (they’re perfectly correct to say so), they also produce a variant designed specifically for that purpose; available to order at no extra cost, which gives ±5dB of boost/cut in 1dB steps.

The VSE-2 costs an eye-watering amount for those of us with modest fiscal horizons, but that’s hardly a surprise given what it’s capable of sonically. Yet, when compared with what’s out there in its own price/performance bracket, the price of the VSE-2 isn’t actually uncompetitive.

If you’re fortunate enough to be able to afford and justify the purchase of a VSE-2, I’d urge you to audition it without delay. The rest of us must content ourselves with dreaming...

£4620 including VAT.

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www.vertigosound.com

There’s a footnote to be added to the history of the proportional-Q LRC filter. Although, at least in my personal opinion, an LRC filter really benefits the sound of a three- or four-band EQ, it became a real liability in multiband units, such as graphic equalisers designed for room equalisation and sound system tuning. What’s needed for those applications are tight bands, with no interaction between them. The first true graphic EQs arrived circa 1963 in the shape of Cinema Engineering’s six-band Type 7080, and Langevin’s seven-band Model EQ-252-A, both of which were proportional-Q units.

Although API had designed a proportional-Q 10-band graphic EQ in 1967, it wasn’t until 1976 that an IC-based circuit, which virtually eliminated interference between adjacent bands, appeared in National Semiconductor’s Audio Handbook. This new development offered a real alternative to the inductor- and gyrorator-based circuits of the day, but its bandwidth performance was not sufficient for third-octave equalisers. It took until the early ’80s for 31-band graphics with ‘constant-Q’ characteristics to appear, and more or less relegate inductors and gyrators to the EQ history books.

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Install Pianoteq on your laptop and connect it to your digital piano to immediately enjoy an amazingly real piano experience. Installation is simple and fast.
Golden Age Project’s Comp-2A is a design that’s unmistakably derived from the legendary Teletronix LA-2A, which has, over many decades, proved endurably appealing. Its combination of electro-optical compression, input and output transformers and tube amplification mean that it’s easy, quick and intuitive in use, and generally smooth sounding. But it can also be driven into a nice bright-sounding distortion that can work well on a range of sources from vocals to snare drums.

In this month’s competition, we’re giving you the opportunity to win a pair of Comp-2As, complete with the optional rack tray — they’re stereo linkable, so in effect this gives you a stereo/dual-mono LA-2A in 2U of rack space.

Don’t be fooled by the half-rack width format — what lies inside is a faithful implementation of Teletronix’s classic design. As with all Golden Age’s outboard products, designer Bo Medin’s approach has been to start with a revered — but for many people prohibitively expensive — piece of equipment, and then try to find the best ways to reduce its construction costs without impacting the signal path. To that end he’s worked out the circuit layout, and reduced the size of the enclosure and had the device manufactured in China. You’ll also find unbranded — but carefully chosen — transformers and valves inside but, cleverly, Bo has designed the circuit so that you can drop in the same tubes as the original. This version features two tubes in the signal path: a 12AX7 dual-triode amplifier, and a 6N6 cathode-follower (also a dual-triode type).

The result is a wonderfully smooth-sounding device in the classic LA-2A tradition, which can be ‘upgraded’ if you wish. But even without such mods, the Comp-2A really impressed our reviewer, who in his June 2017 review (http://sosm.ag/jun17goldenage) wrote: “If you know you want an LA-2A-style hardware compressor and have the money available, I sincerely doubt that you’d regret investing in one (or a pair) of these.”

This beautiful pair of Comp-2A compressors also comes with the Unite Big Rack Kit, an adaptor for mounting two Comp-2A devices in a standard 19-inch rack.

To be in with a chance of winning this awesome prize, enter fill out the form on the SOS web site by Friday 5th January. Good luck!

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AX SERIES

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When it comes to audio editing software, most of the well-known choices have been around for many years and are very mature applications. The same may be true of Sound It!, produced by the Japanese developers Internet Co Ltd, but it’s likely to be less familiar to Western studio types. Could that

Internet Co

Sound It! 8 Pro

Audio Editing Software

If you’re looking for an alternative to the established stereo editing packages, Internet Co’s slick dual-platform offering might pique your interest...

PROS

- Very competent audio editing environment.
- Good collection of bundled plug-ins.
- Easy to use.
- DSD audio support for those who require it.

CONS

- No preset system for the Mixer plug-in chain.
- No surround support at preset.
- Current price might make it difficult to challenge established competitors.

SUMMARY

Sound It! 8 Pro provides a solid, straightforward editing environment for mono and stereo audio, and includes a good collection of bundled plug-ins. However, unless DSD audio support is critical to your needs, there is stiff competition from the established market leaders.
be about to change with version 8? The new release has been fine-tuned for the English-speaking market, and with both Windows and OS X versions available, the company are clearly intent on expanding the product’s reach.

On paper, the feature set looks promising and, with an impressive bundle of plug-ins included and the intriguing option of support for DXD/DSD native audio (with included and the intriguing option of support for DXD/DSD native audio (with included and the intriguing option of support for DXD/DSD native audio (with included and the intriguing option of support for DXD/DSD native audio (with

Sound Principles

Sound It! 8 Pro (which I’ll call SIBP from now on) has a solid set of basic audio editing features. These include typical file editing options, audio format conversion, a broad array of bundled plug-ins (spanning EQ, dynamics, spatial effects, modulation and mastering treatments), basic automation, useful metering tools, batch processing, well-featured CD creation, the ability to create Acid files and, as mentioned earlier, support for handling DSD and high-resolution PCM (DSDX) files. Exactly what you can do with these more specialist audio formats depends upon the specification of whatever audio interface you might have available but, even so, DSD support is a rarity amongst the mainstream audio editors.

The software is available as a download, and installation follows pretty standard procedures under both Windows and OS X. Although I’ve experienced slicker installation and authorisation processes, Internet Co’s gets the job done. It’s also worth noting that there is a fully functional (bar some batch and export options) three-day trial available. This is a sensible policy and will be very welcome for potential new purchasers. The software also includes a comprehensive PDF manual that, despite a few ‘second language’ issues, provides a good introduction to the feature set.

On The Outside

SIBP’s graphical user interface is pretty conventional, with a main menu and toolbar strips that provide easy access to many of the key file-management and editing tools. At the base of the display are the transport controls, which include a jog wheel, and four additional buttons to access the Effects, Mixer, a Timer to make timed audio recordings, and a Media Browser. All these features are kept fairly compact so that the central portion of the display can accommodate floating windows for your open audio files.

Multiple files can be opened at the same time, and are displayed in separate windows, each with an ‘overview’ waveform at the top and a more detailed view underneath. You can use the former to navigate the latter when zoomed in on the lower waveform display. The main windows can be resized, as can both the upper and lower panes of the waveform display. It’s also worth noting here that although there are plenty of options for audio formats, bit depths and sampling frequencies, surround files are not supported.

Routine recording and editing tasks will feel intuitive to anyone with experience of another mainstream audio editing environment. When you create a new empty audio file, you can specify its format and, providing your audio hardware is connected and configured, you just hit the record button and off you go. You can record multiple times into a single file, whether adding new material at the end or dropping in within the middle of a recording; existing audio is simply shifted along the timeline to accommodate it.

Editing options include the usual copy, paste, cut, trim, merge, fade, crossfade, gain adjustment and so on. If you make a selection in one file, then drag and drop it onto an open spot in the workspace, the selection is automatically turned into a new file. Dragging and dropping audio between open files is also supported. SIBP also offers basic automation options for volume and editing and those working in more niche, high-fidelity environments.
apply an appropriate crossfade or gap metadata, apply level matching or global to the track list section, specify some Here, you can add suitable file formats for burning. The Play List if you want to build a CD layout based upon marker positions, or populate the Play List if you want to build a CD layout for burning. The Play List itself provides an unfussy application would be to convert a collection and apply them to all the audio files within a single folder, and export the processed files to a second folder. The most obvious application would be to convert a collection of files to a different format, but you can also apply gain changes, normalisation or fades. Due Process As well as the range of standard editing features described above, SI8P also ships with an extensive range of VST and/or Audio Units plug-ins. There are 49 in total, spanning a pretty wide range of the typical processing and effects options. Some of these, such as the various modulation plug-ins, would perhaps be more obviously at home within a music production environment, but those involved with sound design will, no doubt, be able to put them to good use. The EQ, dynamics, audio restoration and metering options are pretty sophisticated, catering to all sorts of audio editing duties as well as DIY mastering. Taking the latter application as an example, you could construct a suitable processing chain from the eight-band EQ, the Multi-Compressor, Stereo Enhancer and either the Maximizer/Limiter or Sonnox Limiter. While this might lack the integrated approach offered by (for example) iZotope’s Ozone or Wavelab’s MasterRig, the tools are all there.

The last of these plug-ins is just one of several Sonnox processors included within the software. Others include the rather nice Sonnox Reverb, EQ Filter and three audio restoration options: De-Buzzer, De-Clicker and De-Noiser. These versions have been around for a while, but they still do a very respectable job for basic clean-up tasks. Those handling field-based audio recordings made in less-than-ideal circumstances would, I’m sure, find them very useful.

Plug-ins can be applied in two ways. First, you can apply effects or processing directly to an individual audio file or a selection of audio within an open file. While you get the option to audition what the processing will do, once you commit, the processing is applied destructively (as was the case with Sound Forge for many years). Alternatively, you can open SI8P’s Mixer panel, where it’s possible to construct a chain of up to eight effects. These are applied non-destructively in real time. The contents of the Mixer panel effects chain are set on a ‘per file’ basis. 

Place Marking
You can place four different types of marker within an audio file: Generic, Division, Region start/end and Chapter. These are stored if you save the file as a WAV or in SI8P’s own SIW file format. Markers can be created automatically to mark the locations of transients, or points where silence is detected. As well as allowing you to navigate quickly between sections of a file, Division and Region markers, once placed, let you quickly generate separate audio files based upon marker positions, or populate the Play List if you want to build a CD layout for burning.

The Play List itself provides an unfussy system for putting together a CD project. Here, you can add suitable file formats to the track list section, specify some metadata, apply level matching or global EQ, audition the sequence of tracks and apply an appropriate crossfade or gap between tracks before burning to your CD-R/RW drive. The combination of using Division Markers and the Play List (including the crossfade option) might make for an appealing combination for producers who work with stereo recordings from live performances and wish to render them to a CD format. Incidentally, you can also export Disc Description Protocol (DDP) files for delivery of disc masters for duplication. Oh, and SI8P will also import audio from a CD and convert the contents into audio files.

There are a few other core features worth mentioning. First, SI8P can export Acid-formatted WAV files. Although Acid Pro is perhaps not as widely used it once was, the file format is used by lots of other programs too, and unlike WAV or AIFF, provides a way of embedding tempo and pitch information. Second, the software will import a MIDI file and, using a GM-style sound set, will convert the MIDI file into an audio render. Yes, the results are suitably cheesy, but it does work and I’m sure some might find it occasionally useful. Finally, the Tools menu includes a compact Batch Processing option where you can configure a small number of file manipulation options and apply them to all the audio files within a single folder, and export the processed files to a second folder. The most obvious application would be to convert a collection of files to a different format, but you can also apply gain changes, normalisation or fades.

Alternatives
The two most obvious competitors are Steinberg’s Wavelab and Magix’s Sound Forge. The full versions of both of these products are more expensive than SI8P and their prices are a pretty fair reflection of their respective feature sets, though neither supports DSD audio.

The more affordable Wavelab Elements is perhaps closer to SI8P and, while it offers the same Sonnox restoration plug-ins and a decent complement of other processors, it can’t match SI8P in terms of features such as DDP support, comprehensive loudness metering or DSD audio support. It is, however, considerably cheaper.

Tascam Hi-Res Editor, which is a free download from the Tascam web site, does provide DSD audio support but has a pretty modest feature set for routine editing duties.
As a professional, you can leave nothing to chance with sound recordings. With the DR-100MKIII from Tascam, you are able to cope with even the bigger challenges. With its excellent sound quality, simple operation, rich features and mechanical robustness, this audio recorder is designed for exactly the quality and reliability you expect in your daily work.

Linear PCM (WAV/BWF) with up to 192 kHz at 16/24 bits or MP3 with 128/192/256/320 kbit/s at 44,1/48 kHz, -124 dB EIN, 102/109 dB S/N ratio, two built-in stereo microphones (omni-directional/unidirectional), digital input (AES/EBU, SPDIF), input level –58 dBu to +24 dBu, 48 V phantom power, MS encoder/decoder, 4-step low-cut filter, lockable XLR/TRS combo input connectors by Amphenol, stereo line input and output with adjustable level, several auto functions including level alignment and limiter, dual recording feature ...
rather than globally, with the Mixer panel configuration being stored alongside the audio file and recalled whenever the file is reopened. A Freeze option lets you apply the Mixer effects chain permanently.

This all works rather nicely and, as described above, you could easily imagine building a very functional mastering chain. However, as far as I can see, while all the plug-ins themselves offer preset systems, there is not a preset system for the Mixer’s insert effects chain itself. I can’t imagine this would be too difficult to implement but, without it, having to configure your favourite processing chain from scratch every time can quickly become frustrating. That said, the overall bundle is impressive and my only other comment is that the plug-in interfaces lack some of the consistency often found in ‘stock’ plug-in collections.

On The Meter

One of Sound It!’s bundled plug-ins in particular is perhaps worth a few extra words. Now that we have all survived the loudness wars and, with a bit of luck, come out the other side with our ears intact, both hobbyist and professional audio engineers will appreciate the role of appropriate metering in mixing and mastering to appropriate loudness levels. SI8P supplies exactly that in the shape of the Loudness Meter plug-in, which can easily be dropped into the final slot of the Mixer panel to let you see exactly what’s going on.

The meter supports EBU R128 (European) and ARIB TR-B32 (Japanese) standards and offers readings for integrated, short-term, momentary and maximum loudness, while you also get the loudness range displayed. The meter includes three ‘gate’ options to remove the effect of lower-level signals or silence on the meter’s averaging algorithms. You can also switch between relative and absolute loudness displays. While I didn’t do any exhaustive testing, a quick comparison between SI8P’s Loudness Meter and iZotope’s Insight suggested that they produced broadly similar results within the limits of their respective configuration settings and averaging processes. If you need to ensure your levels don’t fall foul of broadcast standards, therefore, the Loudness Meter ought to be a useful guide.

Easy Does It

One advantage of Internet Co’s very straightforward approach to the implementation of file handling and editing tasks is that the initial learning curve is not intimidating, and I think even those encountering audio editing for the first time would find SI8P very easy to get to grips with. I did the bulk of my testing with the Windows version and, overall, the performance was very solid and generally very smooth. I also had access to a beta of the Mac OS version. Visually, and in terms of
One intriguing item in the SI8P feature list is its ability to handle high-resolution Direct Stream Digital (DSD) audio files. For recording and native playback in one of the DSD formats (DSD64/128/256) you need to have audio hardware that’s also equipped to handle these formats. These used to be something of a luxury item but, while the DSD format is still something of an audiophile niche platform, there are now streamlined, low I/O-count interfaces available such as the Roland Super UA at around the £250 mark for those who wish to explore the possibilities.

I didn’t have access to such an interface during the review period, but SI8P does still allow you to open DSD files without a suitable interface connected, offering to convert them into a PCM format at whatever bit depth and sample rate you care to specify, including rates that qualify for DXD (and which would be way beyond the sample rates supported by the vast majority of conventional audio hardware). Equally, you can choose to export any audio file in a DSD format should you wish and, again, you can choose between various formats.

I experimented with loading and saving various DSD files with SI8P and, from a technical perspective at least, the process seemed to work very smoothly. The ability to handle the DSD format is something of a specialist requirement but, for some at least, it might be SI8P’s unique selling point. As far as I’m aware, none of the obvious mainstream audio editor competitors offer DSD support, although there are specialist tools for just this kind of application — Tascam’s Hi-Res Editor, for example, is available from their web site as a free download. Incidentally, if you want a brief introduction to the DSD format, then see Hugh Robjohns’ Q&A response back in the February 2009 issue of SOS.

Based upon my Windows experience, if you need a no-fuss program for routine mono and stereo audio file editing, SI8P is perfectly capable. It provides a compact set of features covering those tasks you might perform 90 percent of the time. The plug-in collection is also pretty comprehensive, with the inclusion of the Sonnox audio restoration tools a nice touch. I also found that, in use, the Play List made CD construction easy. As far as I could take it without specialised audio hardware, I was also impressed with the handling of DSD audio.

On the down side, the lack of a preset system for the Mixer’s effects chain is frustrating; given that the generally straightforward design of SI8P means repetitive tasks can otherwise be dealt with efficiently, it’s a pity that one of the most important, most repeated tasks — configuring your processing chain — can then slow you down. This is a shame and, hopefully, something that can be addressed in an 8.x update at some stage.

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The iPhone 8 models are really — and perhaps surprisingly — impressive, and I wouldn’t hesitate to recommend them. But perhaps surprisingly — impressive, and I wouldn’t hesitate to recommend them. But the big question is whether to wait (or opt) for the iPhone X. No doubt the X will look desirable, with its edge-to-edge display; although since both are powered by the A11 chip, if you can get the same results at a cheaper price, an iPhone 8 model might be a better option. To hijack the last sentence from a recent column written by the historian Niall Ferguson: “watch this Face!”

Mark Wherry
£ From £699
W www.apple.com

Korg iMono/Poly
Synthesizer iOS App

In 2004, Korg embraced their analogue heritage by releasing MS20 and Polysix soft synths in their first Legacy Collection. In 2007, these were joined by the Mono/Poly to form the AE (Analogue Edition) and, with soft versions of the M1 and Wavestation as well as hardware recreations of the MS20 among their current offerings, the company remain unembarrassed when it comes to mining their considerable legacy. Their latest soft synth is an iOS version of the Mono/Poly, which appears to be so similar to previous versions that it can swap patches with them.

The display on the iPhone 8 is of the usual high quality one would expect, and Apple define it as a Retina HD display. Although the resolution is the same — 1334 x 750 for the standard 4.7-inch model, and 1920 x 1080 for the 5.5-inch Plus — the HD appendage reflects the new display’s wider colour gamut and the inclusion of the True Tone functionality from the iPad Pro. This adapts the colour temperature of the display to the environment in which it’s being used, and, as I mentioned when reviewing the 9.7-inch iPad Pro, it’s a subtle effect, although one you notice if disabled.

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In recent years Apple have been noticeably improving the audio quality of both iOS and Mac OS-based devices, and the iPhone 8 is no exception. Apple claim the speakers are 25 percent louder with deeper bass, and I have no reason to doubt them. Obviously we’re not talking about studio-quality monitoring, but any improvements to audio quality in a device you’re going to carry around with you every day are welcome.

The iPhone 8 models are really — and perhaps surprisingly — impressive, and I wouldn’t hesitate to recommend them. But the big question is whether to wait (or opt) for the iPhone X. No doubt the X will look desirable, with its edge-to-edge display; although since both are powered by the A11 chip, if you can get the same results at a cheaper price, an iPhone 8 model might be a better option. To hijack the last sentence from a recent column written by the historian Niall Ferguson: “watch this Face!”

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Of course, its voicing isn’t limited to that of the original synth, and what we have here is a truly polyphonic soft synth based upon the four-oscillators-per-note architecture of the Mono/Poly, but with additional voicing modes, velocity and aftertouch sensitivity, Synchronization of the LFOs and arpeggiator (which offers a random mode that the original lacked), plus vastly expanded modulation capabilities courtesy of an eight-slot modulation matrix.

In addition to the basic synthesizer voicing, the effects that Korg designed for their Legacy synths are also recreated here. The 18 algorithms — comprising dynamics, EQ, modulation effects (including an approximation of the wonderful Polysix ensemble), an exciter, a decimator, delays and reverbs — are well specified and offer a degree of flexibility that you might expect from dedicated plug-ins. Consequently, the results of inserting appropriate effects into the iMono/Poly’s two slots and programming them sympathetically can be stunning. Furthermore, many of the effects’ parameters can be modified in real time using MIDI controllers including velocity and aftertouch, as well as a small selection of MIDI CCs, which opens up all manner of performance possibilities.

The other significant additions are two Kaoss pads, one dedicated to modulation duties and the other to triggering notes and chords. The first of these allows you to emulate the effects of the pitch-bend and modulation wheels, or to control the filter’s cutoff frequency and resonance, or to assign any two of a huge range of destinations to the X and Y axes and then control them by sweeping your finger over the pad. The second allows you to choose one of numerous scales and then control the pitch of the note or the pitch and inversion of the chord produced by touching the pad. They can be great fun, whether used to modify the sounds you’re playing on a connected keyboard, or as sound generators for players who prefer not to play in the conventional manner.

When I first encountered it 10 years ago, I felt that the Legacy Mono/Poly was the least accurate of Korg’s soft synths, and this remains true for iMono/Poly, though sometimes in a good way. Yes, I know that it’s heresy but, because the soft synth’s filter is much tamer than the original’s, it can sometimes sound warmer and less aggressive, which can be ideal for things such as pads and orchestral sounds. If you want to create analogue nastiness, the original wins all the prizes — its filter can be downright belligerent and its X-Mod and FM effects sections still do an excellent job of sounding tortured.

Comparisons aside, you can obtain a huge range of sounds from iMono/Poly and, while there are 128 patches provided with it (and another 128 are available as an in-app purchase) I must admit that I wasn’t hugely enamoured of many of them because I found that the synth was capable of much — and I mean MUCH — greater depth than they offered. If, like me, you want to delve deeper, there are 20 template patches that I found to be good starting points for developing your own sounds.

The iMono/Poly is limited to a maximum of 32-note polyphony rather than the 128 notes of the Legacy version, but I didn’t find this to be a problem, even when playing polyphonically in Unison mode. So, were there any unpleasant surprises? Well, yes... two. The first was the MIDI implementation; there’s no MIDI ‘learn’ for the knobs and switches, and no MIDI Clock sync, the latter of which will be a significant limitation for many potential users. Furthermore, the synth engine recognises a maximum of four MIDI CCs as modulation sources at any given time. You use the setup menu to choose which will be recognised, whereupon they are designated Virtual Patch Control Change 1 – 4, and you can then direct them to the parameters of your choice within the modulation matrix. If you were expecting to be able to automate a large number of parameters and watch the knobs sweeping around while you sat back and sipped your vodka martini, you’re going to be disappointed, but if you treat the VPCCs as performance controls rather than automation facilities, you won’t go far wrong.

Secondly, while iMono/Poly responds to external MIDI, Core MIDI and Bluetooth MIDI, supports both Audiobus and Inter-App Audio, and can be used as a Montpellier gadget (albeit with a somewhat different layout) within Korg Gadget, there’s no AU option. Some potential users are already complaining about this, so it will be interesting to see whether Korg respond.

I’ve never been a fan of the ‘studio in an iPad’ approach to music making, nor a strong advocate of using computers or tablets on stage. Nevertheless, having placed my iPad running iMono/Poly on the laptop tray of an 88-note master controller and having treated it as, in effect, the touch-screen of a large, virtual analogue synth, I was impressed; it sounded every bit as big as the keyboard on which it was sitting. Sure, it has a couple of limitations, but you can buy a top-of-the-range iPad, a controller of some description and the soft synth for less than the price of a beaten-up, 40-year-old Mono/Poly with no memories, no effects, no Kaoss pads and no modulation matrix, but replete with numerous mis-triggering keys and scratchy pots. A fuller test may or may not have thrown up some additional issues but, even if it did, iMono/Poly sounds great and represents excellent value for money.

Gordon Reid
E £28.99
W www.korg.com
By turning your hotel room into a project studio, you can deliver broadcast-quality VO tracks when you travel.
For those of us who make our livings doing voiceover (VO) work, leaving home needn’t mean losing work. In fact, a VO talent with a basic understanding of acoustics can use a laptop computer, a portable interface, and a small studio microphone to deliver broadcast-quality voiceover tracks from just about anywhere in the world. My wife and I both work in the industry, and record mainly in our home studio in São Paulo, Brazil, and at production houses in the region. But we spend a lot of time on the road as well — as an American expat, I make a point of returning home at least once a year.

A trip to the USA in March this year took us both to the VO Atlanta voiceover conference in Georgia, and then to the South by Southwest (SXSW) festival of film, technology and music in Texas, before we visited family and friends in California. That’s a lot of time to take off work when you freelance, so for three weeks we exchanged our home studio for a portable studio that we wheeled around in a carry-on suitcase.

During the three weeks we spent away from our home studio, we gave voice to dozens of audio projects: an in-flight safety announcement for a major airline, a national TV campaign for a furniture and home accessories retailer, corporate videos for companies in and out of Brazil, and case studies for advertising agencies, to name a few. Each project posed unique challenges, but we managed to deliver with speed and quality in every case.

In this month’s Session Notes, then, I want to take you through how we approached our work on this trip, as well as teasing out a few lessons for you (and for us!). Before I do, by way of background, it’s worth mentioning that, compared with music production, voiceover work has some specific recording requirements and aesthetics — you can find out more about that in my two-part SOS feature on recording professional voiceovers at home.

The acoustic treatment in our São Paulo home studio, pictured, follows the principle of thirds: one-third absorption, one-third diffusion and one-third reflection. In our hotel rooms, where pillows and blankets substituted diffusers and acoustic clouds, we ended up with something closer to two-thirds absorption and one-part reflection.

Our room at the Airport Marriott in Atlanta had carpetless floors and an audible drone from the air conditioner. We improvised a gobo, using a sleeping bag, hangers with clips and a nylon tie-down strap with a ratchet and hooks to tame reflections and reduce bleed from the vents on the other side of the room.

In Atlanta, we positioned a pillow, a small reflection filter and jackets inside the closet to make our recordings sound drier. Colleagues at the conference joked that with pillow alters set up in so many rooms, the maids must have thought we were part of a religious sect.
What A Carry On!

Carry-on baggage policies vary from airline to airline and country to country, so if you want to make sure your gear always stays close at hand, make sure you verify carrier regulations before catching your flight. For our trip to VO Atlanta and SXSW, we packed a Neumann TLM 102 large-diaphragm condenser microphone and an Apogee MiC USB microphone that I used with my iPhone. We brought along an iPad from which to read scripts, a MacBook Pro running Adobe Audition CC for recording and editing the audio, and a Mac Mini, which we connected to hotel TVs with an HDMI cable, to act as a second workstation. We also took a Universal Audio Apollo Twin Duo audio interface, a pair of Sennheiser HD 380 Pro headphones, WindTech PopGard 2000 and Sterling Audio PF1 pop filters, a Proline PLDMS1 desktop microphone stand, a Sterling Audio UMS Utility Microphone Shield, and a 20-foot XLR cable — that would give us the freedom to record and edit across the room from our vocal booth.

Booking Your Room

Making hotel reservations early is an important first step to building your project studio on the road, especially if you’re attending a convention or a festival. We booked our hotels four months in advance and opted for rooms with two queen-size beds instead of a single king, because that would allow us to erect a ‘mattress pyramid’ over the space between the beds, if required. In theory this creates a spacious voiceover booth, though in practice it’s very much a last-resort — you never know what you’re going to find when you flip up a hotel mattress!

Over the phone or on a web site, I explained our unique situation to hotel staff and made some special requests. Hotels will usually do their best to help you out if you mention you’ll be recording voiceover in one of their rooms — for all they know, you might be a famous YouTuber! What requests did I make? For starters, at each hotel, I asked for a room far from the elevators and at the end of a hallway, which would limit foot traffic outside our door. I also asked for a room facing away from the highway, the hotel pool, or any other obvious sources of noise, and for a room on a high floor — in fact, on the top floor, if possible. “What about rooftop machinery?” I hear you ask. Well, in my experience, loud upstairs neighbours tend to cause far more trouble than rumbling air conditioners. Finally, I confirmed that the room would have both a safe and Wi-Fi access to the Internet.

While a hotel closet may not be as dry as a pillow fort, it can offer benefits. For instance, a voiceover actor has more space to move around. And, if the room is quiet, the acoustics of this setup sound surprisingly close to those of a professional studio.

Project Studio Setup

As soon as we checked into each hotel, I took a moment to study our new recording space. How quiet was the room? Could we switch of the refrigerator and central air-con if needed? Did outside noise bleed through the windows and walls? In other words, just how low could I get the noise floor for our recordings? Then, I walked around each room clapping my hands and grunting like an ape to get a notion of acoustics. Which part of the room was the least resonant? Could the closet serve as a kind of booth, or would we be better putting a luggage rack, ironing board, sofa cushions, or chair on top of a desk or bed? Were there plenty of pillows and blankets to deaden the space around the mic? Did the room have a remote control for the TV (not just for the cable box) so I could adjust the screen settings and use the TV as a computer monitor? If a room didn’t pass my inspection, I’d get keys to look at one or two more options before unpacking our gear.
ORCHESTRAL SWARM

BEAUTIFUL LONG SONIC TEXTURES CREATED FROM LAYERS OF MICRO-MOMENTS

RECORDED AT BRITISH GROVE

SPITFIRE AUDIO
We spent five nights at the Airport Marriott hotel in Atlanta. The voiceover conference took place inside the hotel’s convention centre, so it made sense to stay there, despite the air traffic overhead. We were relieved to learn that the room had thick, triple-paned windows to keep out jet noise! But still there were other challenges, not least the carpetless floor, a buzzing refrigerator and a droning central air conditioning system...

The refrigerator could be unplugged, but there was no way to switch off the air vents. Since the hotel was fully booked and as other rooms would likely present similar problems, we set up our booth in a closet, which was close to the door and far from the vents by the windows. We hung jackets inside the closet, tucked a pillow in the corner behind the mic, and connected a small reflection filter to the mic stand. We also suspended a sleeping bag across the room’s entrance hall, using nylon straps and hangers with clips. This improvised ‘gobo’ would help to tame reflections and attenuate noise from the vents. In Adobe Audition, I set a high-pass filter at 80Hz and used iZotope RX5’s DeNoise feature to give our voiceover tracks a few more decibels of dynamic range.

At our hotel in Austin, where we spent a full week, we took a similar approach. The AT&T Conference Hotel sits on the edge of the University of Texas, so I imagined it would be quieter than the airport hotel. However, despite making an early reservation and sending our list of special requests, we got stuck in a room overlooking a construction site — you need a pretty loud musical bed to mask the sound of a jack hammer! Much to our relief, although the hotel was at maximum occupancy for SXSW, the front desk staff found an unoccupied room in a separate wing of the hotel. There, we set up our gear in a large closet, draping our sleeping bag over the open doors on each side to act as an absorbent fourth wall. We also took advantage of the room’s broadband Wi-Fi signal and big TV screen to set up a second workstation with our Mac Mini, using an ironing board as an improvised desk.

The final leg of our trip, in California, demanded a bit more ingenuity, since we drove from city to city and didn’t always have time to set up a well-built recording space. We rented a mid-sized SUV in Los Angeles to act as a mobile booth, and we spent a night in Santa Monica with our Brazilian friends Eduardo and Bia, who moved to the United States a few years ago and who also work in voiceover. They’ve repurposed the closet of their main bedroom into a voiceover booth, adding Auralex panels to the walls and ceilings and recording with a Shure SM7B, an Apollo Twin and a MacBook Pro with Adobe Audition CC. Dynamic mics such as the Shure SM7B and the ElectroVoice RE20 are popular in home voiceover studios, but I find them a little bulky to take on the road.

Recording & Editing

In addition to Adobe Audition CC and Twisted Wave, I often used iZotope RX5 on the road. If you need to reduce background noise and mouth crackle, this software does magic (and the newest version, RX6, which I’ve not yet grabbed, introduced a stand-alone feature called Mouth De-click.) I also had Plugin Alliance’s SPL De-Verb installed, just in case we had to record in a highly reflective environment, but I rarely needed it. Instead, when away from our hotel room, we’d find a way to record in our rental car, in somebody’s closet, or in any quiet and relatively reverb-free space that was available. In situations like these, we recorded with our Apogee MiC, using the...
Twisted Wave app installed on my iPhone. Twisted Wave has basic editing features, so I could upload edited tracks to Dropbox without a laptop or Adobe Audition CC.

I recorded all our tracks as 48kHz, 16-bit, mono, uncompressed PCM audio files and, unless a client requested something different, that's how I uploaded them. I would sometimes do some light processing (for instance, a high-pass filter and a touch of noise reduction) before sending tracks to ad agencies or to end clients directly. But if the track was for a production house, I’d do little more than delete mistakes before uploading the unprocessed audio file.

When working with a sound engineer, consider including a description of your gear and a photo of your booth with the link to your file. Also, give the engineer a few seconds of room tone at the start of the recording. Set your levels, hit record and sit silently for a moment so the engineer will have a clean noise print of your recording environment. This will come in handy for noise-reduction algorithms, or for

The New Normal

I spoke with JJ Jurgens, Creative Director for CBS Daytime On-Air Promotion, about voiceover tracks recorded on the road. I wanted to get a feel for whether networks are bothered by the idea of the talent working while they travel, and what the implications are if they’re unable to work when they’re away.

JJ: “In the past, VO talent used to come into CBS and do their sessions in-house. That is not the norm any more. Almost all our sessions are done via ISDN or Source-Connect, which makes it easy for talent to travel. If they take their road gear and can connect with us for sessions, we don’t look to hire replacements for them while they’re gone. Usually their agent will give us a travel schedule, letting us know the hours when the talent won’t be able to connect, like while on an airplane, and we work around this schedule. If we need a read done and the talent doesn’t have a travel kit or isn’t able to go into a studio to record while travelling, in this case we would have to turn to other talent because, unfortunately, in the promo world, we must get spots out so quickly that we don’t have time to wait for the talent to return.”

The Mix With The Masters Program offers the unique opportunity to work closely with the world’s top music mixers, producers and engineers and to improve one’s skills in music production. Since its successful debut in 2010, it has hosted over 1000 participants and 85 seminars.

In addition to these seminars, Mix With The Masters has now launched an exclusive online community for engineers and producers, offering an incredible array of exclusive online services, enabling its members to discover the approaches and techniques of A-list music mixers and producers. You can now become a member to get an unlimited access to hundreds of videos, and to take advantage of our unique services such as Webinars, One-Day Seminars and much more.

In like manner, the Mix With The Masters residential seminars are held at the Studios La Fabrique in the South of France, which is the perfect setting to both learn and relax. Each weeklong seminar includes, among other activities, a series of discussions about production techniques, mental and philosophical approaches towards mixing, career advice, workshops, during which the guest speaker tracks a band, and mixes in front of the attendees, as well as giving them feedback on their own projects.

The process of greatness fostering greatness has long been recognized, and is the reason why master classes are organized. The Mix With The Masters Program is proud to be part of this tradition, both through our online community and weeklong seminars.

www.soundonsound.com / December 2017

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On The Go With Joe Cipriano

While attending VO Atlanta, Simone and I met voiceover superstar Joe Cipriano, who does promos for comedy shows like *The Simpsons*, *Two And A Half Men* and *Mike And Molly* on the Fox and CBS networks, and does in-show announcing for top-rated shows like the *Emmy Awards*, the *Grammy Awards* and *America's Got Talent*. Joe often records on the road and he generously shared his insights with me for SOS readers.

Joe: “I try to limit the amount of equipment I take with me when I travel, so it takes up as little space as possible in my carry-on and when I set up to work. I use a MacBook Air as the heart of my remote recording studio. I take a Sennheiser MKH-416 microphone with me, with an audio cable to connect the mic to the interface. My interface for getting in and out of the MacBook Air is an Apollo Twin Solo, which connects to my laptop via Thunderbolt. Previously, I was using a USB interface, but found that using the Apollo Twin Solo frees up a USB port on the computer and allows for all sorts of sound treatment via the Universal Audio device and plug-ins. I also bring low-profile ear buds made by 1MORE. I use Pro Tools to record and edit, both in my studio at home and on the laptop when I’m away. So, it’s important to have that extra USB port free for the iLok that holds the Pro Tools license; I would have a problem if it wasn’t for the Thunderbolt interface.

“As far as where to set up, I use various methods. In a room, I’ll usually put a towel down on top of a desk or bureau. I place a luggage rack or an ironing board on top of the towel to be used as infrastructure for my makeshift voiceover booth. I drape whatever is there in the room, such as a blanket, over the top of the luggage rack or ironing board, ensuring that those three sides are covered. Then I set up pillows inside this ‘fort’ that I’ve built, and I place the microphone, laptop and interface on the top of the luggage rack or ironing board on top of a desk or bureau. I place up pillows inside this frame for my hotel voiceover booths. He positions this frame and directional microphone.

Joe also records broadcast-quality audio from the backseat of his car. Parked in a quiet area, an automobile makes an effective vocal booth, since the air-tight interior provides a balanced acoustical environment.

Joe also records broadcast-quality audio from the backseat of his car. Parked in a quiet area, an automobile makes an effective vocal booth, since the air-tight interior provides a balanced acoustical environment.

Joe Cipriano uses a luggage rack or ironing board as the frame for his hotel voiceover booths. He positions this frame on a desk or bureau, drapes blankets or a duvet over the top, stuffs pillows inside, and leaves enough space for his laptop, interface and directional microphone.

Joe Cipriano was on a tight deadline. I arrived via airplane at Columbus, Ohio, and was picked up by my friend AJ McKay. I told him we needed to find a relatively quiet place to park, so I could do a live session back to CBS in Los Angeles. As we pulled over into a quiet parking lot, and as I was setting up, suddenly it began raining. It wasn’t just raining; it was coming down in buckets, and the sound of the downpour hitting the roof and windshield of the automobile was deafening. My friend was a quick thinker, and he found a petrol station that had a covered roof over the area where you pump the gas. As soon as we pulled under that roof, the rain stopped pelting the car and I was able to hook up to the studio in LA live. The recording session went without a hitch. And when the client asked what studio I was working in, I told them it was a little place in Ohio called the Chevy Studio.”

First, even though it’s important to let the sound engineer know you’re recording on the road, you don’t need to tell the end client. During our trip, we had a producer and his client on the same Skype conference call. Prior to this session, the producer asked us not to mention we were recording in a hotel. He trusted that we’d send him broadcast-quality audio, but was worried that his client wouldn’t be so understanding. Second, when recording a job at your home studio before a trip, consider using the same mic that you will be taking on the road, or at least set up your ‘road mic’ next to your studio microphone and record the job with both. This way, if you need to deliver retakes from the road, you won’t be adding silence between sections of speech. (Complete silence in a voiceover track can sound unnatural, especially if the track was recorded in a non-studio environment — cutting and pasting ‘room tone’ generally works far better than muting intervals or using a noise gate.)

On the topic of recording and editing, bear in mind two final considerations.
have to record the script all over again, and the retakes will sound much closer to the original recording.

**Getting VOs Ready To Air**

Before wrapping up, it’s worth saying a few words about how Comando S Áudio, a production house in São Paulo, processed and mixed one of the voiceover tracks we recorded and delivered during our trip — I’ll use a TV commercial for SBP mosquito repellent as a case study.

My wife Simone recorded this voiceover inside the spacious closet of a quiet top-floor suite at La Quinta Inn in Paso Robles, California. It was one of our better hotel booths and the track sounded great. A few days later, however, we got a call from Comando S Áudio: they needed a line correction. We were in Santa Barbara, having lunch at the house of friends, so we excused ourselves from the table to set up and record in their bedroom, placing our TLM 102 on top of their bed. The small room was quiet and the bed itself, plus a dresser stacked with books, kept the space relatively reverb free, so we didn’t need any pillows or cushions. Simone took care to match the volume and tone of the original recording and she matched her distance from the microphone as well.

Fabricio Mary, the Comando S Áudio sound engineer who mixed the commercial, later sent me an explanation of the steps he took when processing our tracks. For the first recording in Paso Robles, he used Avid’s EQ III parametric equaliser to clean up the low end and reduce problematic resonance by running Simone’s voiceover through a high-pass and low-shelf filter set at 240Hz and narrow Q notch filters set at 240Hz, at the harmonic of 480Hz, and again at 900Hz. Then he added brightness and compensated for a dip in the TLM 102’s frequency response by bringing up the region between 4.5 and 8 kHz. Fabricio didn’t use any noise-reduction or de-reverb plug-ins, but he did point out to us that the hotel-booth recording would have benefitted from an extra pillow on each side of the mic. Regarding the line correction we sent a few days later from Santa Barbara, Fabricio didn’t boost the high-end frequencies, but he did keep the high-pass, low-shelf and Q notch filters in place to attenuate resonance and match the tone. He also ran the second recording through a de-crackler to clean up some mouth noise. With the volumes matched, the takes blended nicely together and the line correction was imperceptible in the commercial that aired. You can listen to our raw files and to Fabricio’s processed track here: [http://sosm.ag/sos-1217-media](http://sosm.ag/sos-1217-media), and you can watch the result at: [www.youtube.com/watch?v=KtxmlCsf-E](http://www.youtube.com/watch?v=KtxmlCsf-E).

**Final Thoughts**

As high-end gear gets cheaper and smaller, it has become easier and easier for the professionals of our industry to work remotely — whether from a home studio, a hotel room or a rental car. So, if you’re a VO talent who likes to travel, why not pack your gear and book your rooms? After all, one of the perks of working in voiceover is that your instrument is easy to carry when you hit the road!
SPL at 1kHz — an impressive figure — and switching in the pad gives another 10dB. This means it won't get upset being in front of loud guitar amps, wind instruments or drums. In addition, there's a switchable low-cut filter. The verdict from Paul White in his 2009 review (http://sosm.ag/apr09at2035) was that the 2035 “produced a slightly flattering result that I found easy to listen to, but didn't go so far as to make the sound obviously coloured when used on vocals.”

Also part of the kit is a pair of critically acclaimed ATH-M40x headphones. These offer professional performance, with 40mm drivers and a 12Hz-24kHz frequency response.

Finally, the iD4 Black rounds out the package. Audient’s portable two-in, two-out desktop USB interface features high-performance converters, zero-latency monitoring, a J-FET instrument DI and a Class-A mic pre with switchable 48V phantom power that was described as “exemplary” in our December 2016 review (http://sosm.ag/dec16id4). This input also doubles for line level duties. As well as being compatible with Windows, Mac and iOS, the iD4 Black has a neat trick up its sleeve. With a press of the iD button, the volume encoder becomes a mouse scroll wheel, meaning that it can be used to adjust the value of virtually any control or virtual knob your mouse is hovering over.

This line-up of home studio favourites is contained in one kit that’s bundled with over £500 of software and services from Eventide, Steinberg, LANDR and Producentech, making the whole bundle worth over £900.

To be in with a chance of winning one of three AT2035-Studio kits, enter the competition on our website by Friday 5th January. Best of luck!

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Win! Audio-Technica AT2035-Studio Kit

THREE TO BE WON!

This month, we’re giving away three awesome Audio-Technica AT2035-Studio kits. The kits have been carefully compiled by A-T so you can set up the ultimate home studio from just one box.

The kit pairs Audio-Technica’s AT2035 cardioid condenser mic and ATH-M40x headphones with Audient’s limited-edition iD4 Black USB interface, plus there’s an additional £500 of software to get your studio setup started.

Designed for recording, the AT2035 cardioid microphone is based around a large-diaphragm capacitor capsule. There’s a small presence peak around 12kHz, but otherwise the frequency response is broadly flat for a neutral sound. Without the pad, the AT2035 can tolerate a punishing 148dB SPL at 1kHz — and switching in the pad gives another 10dB. This means it won’t get upset being in front of loud guitar amps, wind instruments or drums. In addition, there’s a switchable low-cut filter. The verdict from Paul White in his 2009 review (http://sosm.ag/apr09at2035) was that the 2035 “produced a slightly flattering result that I found easy to listen to, but didn’t go so far as to make the sound obviously coloured when used on vocals.”

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Until the recent release of their first floor pedals, the TranZformer GT (for guitar) and TranZformer LX (for bass), getting hardware with the sound of an API mixing console required a significant financial investment. But calling these new creations ‘pedals’ is a bit like calling Mount Everest a hill — they’re both much closer to being floor-mounted channel strips.

Each TranZformer is physically quite large and imposing, and the brushed metal front panels of both models carry the same controls: six API-style knobs for the detented input gain and output level attenuator, the compressor and the three-band EQ, plus three, red LED-equipped footswitches that activate the Tone (EQ), compressor and hardwired bypass. Another red LED, next to the gain control, flashes if either the input or output stages start to clip. Quarter-inch TS jacks provide the unbalanced, high-impedance instrument-level input and output, and an XLR connector deals with the transformer-balanced, line-level out. Both outputs feature polarity switches, and a ground lift is provided on the XLR.

Power to the pedals comes via an included 18V/250mA wall-wart power supply. The input jack feeds an API 2510 discrete op-amp, whose output is controlled by the front-panel gain control. From there, the signal passes to the one-knob, API 525-derived, ‘feedback’-type compressor. The compressors in both TranZformers are identical, with an attack time of less than 5ms and a release time that’s quoted as averaging 100ms, as high-frequency/full-bandwidth content is released faster than low-frequency signals, to give more natural envelope tracking.

The compression ratio is fixed at 2:1, and the six-position switch both lowers the compressor’s threshold to increase compression as it is turned clockwise, and applies make-up gain to compensate.

Next in line are the pedals’ EQ sections, which reveal the only differences between the GT and LX models.

The EQ circuitry consists of a ±15dB, three-frequency, inductor-based circuit “inspired by” API’s classic 553 program equaliser. In the GT version, the low-frequency and mid-range bands have a peaking response, and the high-frequency band a shelving response, whilst on the LX all three bands have a peaking response. The other difference between the sections is that the GT’s EQ acts at 200Hz, 1.2kHz and 5kHz, whereas the LX sees action at 100Hz, 400Hz and 2kHz. From the equaliser, the signal passes to the input of a true audio legend: an API 2520 discrete op-amp that, as in the API 553, drives the output balancing transformer. The signal path splits post-transformer, one branch travelling to the balanced output XLR, and the other to the unbalanced output. In bypass mode, the input and output jacks are directly connected.

API put the sound of their classic consoles right at your feet!
connected, in order to eliminate any loading by or interaction with the TranZformer’s internal circuitry. This also effectively mutes the XLR output, as its signal is taken post the transformer.

**In Use**

Setting up the TranZformer GT and LX to feed amps via the quarter-inch jack output requires a little forethought. Putting either pedal into bypass removes any gain from the 2510 and 2520 op-amps, so unless you’ve taken care to match the volume of the pedal-enhanced signal with that of the bypassed original via the output level attenuator, hitting bypass will result in an abrupt downwards level change at the high-impedance jack output.

Although the TranZformers are described by API as guitar and bass pedals, they’re more versatile than that since, before clipping, the input can handle +13dBu, the unbalanced output can deliver +21dBu and the balanced output can manage +27dBu. This means that either pedal can not only handle inputs from other pedals, preamps, keyboards and synths, but also can drive pedals, guitar amps (modelled or real), effects loops, inputs and inserts.

The results I heard when plugging into either model, with both compressor and EQ bypassed and monitoring the balanced XLR output are, to me, worth the price of admission on their own. With each one, I heard the distinctively warm, punchy, detailed, dynamic sound and solid bass that I remember from API’s 512V 500-series preamp (which I reviewed in SOS December 2016). Winding up the input gain brought in more character, the sound feeling somewhat bigger and richer as the transformer began to saturate and the edge of distortion started to creep in.

**Easy Squeeze**

The compressor is also rather special, the combination of lower input gain levels and its first three settings producing gentle compressions that don’t begin to intrude until the gain is pushed harder. Positions 4-6 are where you start to become aware of the compression, and driving higher levels into these settings causes the compressor to clamp down harder. Although at these higher compression levels the compressors can sound a bit over-fierce for my own taste, on the TranZformer GT it’s capable of delivering compression and length sustain without unwanted pumping. The LX’s compressor is identical and performs to the same high standard, levelling out note volumes, controlling slaps and adding sustain to both acoustic and electric basses, again without any extraneous pumping.

The action and output level of the compressor depends on a combination of input gain, the compressor setting and the frequency spectrum of the signal, which means you must use your ears to determine the compression that you want to achieve. Although the TranZformers’ compressors have a built-in make-up gain structure that can swing the output level from -4dB below the input level to +6dB above it, I found myself wishing for some additional way of altering the make-up gain to more closely match the uncompressed signal level. In a studio situation, this doesn’t matter, as I’d be unlikely to be switching compression up, down or off in the middle of a take; on stage it could be a different story.

**Tone Tweaks**

Whoever specified the centre frequencies and response curves of the TranZformers’ EQs made great choices. On the GT, a 200Hz boost fattens up single coils and cutting it cleanly up humbuckers and acoustic guitars; a 1.2kHz boost opens up the mid-range and adds edge and bite, whilst a cut smooths out over-aggressive guitar sounds without neutering them; and boosting the 5kHz shelf adds a sense of air that’s not over-bright, with a cut dialling back anything untoward in this area and beyond. In addition, because there’s a wide overlap between the response bands at 1.2 and 5 kHz, you can create interesting EQ curves by combining a cut and boost.

The same goes for the all-peak EQ in the LX, for which the response at 100Hz can either flatten up a bass’s low end or thin it out to leave room for a keyboard or synth bass. Boosting at 400Hz adds punch and aggression, while a cut can give a more traditional jazz or country-style sound. The 2kHz peak increases clarity and could be used to emphasise articulation, or be dialled back when necessary. In the LX, the overlap between the three peak response curves again gives you additional tonal options when combining a cut and boost on adjacent bands.

**Thinking Time**

Thought of as API console channel strips for acoustic and electric guitars and basses, the TranZformer GT and LX can do no wrong as far as I’m concerned. They offer results verging on pedalboard guitar and bassist looking to upgrade the front end of your DAW or to enhance your pedalboard you should definitely try the appropriate TranZformer. Highly recommended.

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**Alternatives**

For bass or acoustic guitar, a number of pedals offer integrated EQ, compressor and DI facilities, but electric guitarists will most likely need one or two pedals to duplicate the TranZformer GT’s facilities — and even then, nothing will give you quite the sound of the TranZformers.

---

**PROS**

- Superb performance and API character.
- Effective, musical compressors and EQs.
- Quarter-inch jack and balanced XLR outputs.
- Built to last.
- Great value, given what’s on offer.

**CONS**

- More suited to studio than to stage.
- Bypassing mutes the XLR output.

**SUMMARY**

These pedals are, in essence, floor-mounted channel strips of the highest quality for acoustic and electric guitars and basses.

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**API TranZformers £589**

£589 each including VAT.
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It’s hard to believe that it’s been nine years since DSI reintroduced the Prophet name to a world crying out for recreations rather than digital emulations of classic analogue synths. Looking back, the chattering classes had significant misgivings when DSI did so, but the Prophet 08 won over many (although, if we’re honest, not all) of the doubters, and paved the way for the Prophet 12 and the Pro 2, the Prophet 6 and the OB-6, all of which have offered different flavours of hybrid analogue/digital synthesis. I’ve reviewed all of these, comparing them with their ancestors — the Prophet 600, the Prophet 5, the Pro One and even the Oberheim 4-Voice — and, without exception, I’ve been impressed. So today’s question is, will the Prophet Rev 2 be another good’un?

The Voicing

The Rev 2 keyboard and its desktop module are each supplied in two versions: an eight-voice model and an otherwise identical 16-voice model, and you can upgrade the former to the latter using the Rev 2 Expander Kit. Both versions are bi-timbral, allowing you to create a separate sound, each with its own effect, arpeggio and sequence, in Layers A and B of each Program. These can then be layered or positioned either side of a user-defined split point. For the smaller model, layering reduces the polyphony to four notes while, for the larger, it’s reduced to eight notes, as you would expect. You can audition and edit either Layer individually or while listening to the composite sound, and copy or swap Layers within a Program. Initially, you couldn’t copy sounds from one Program into another, but this was corrected in v1.0.7.2 of the OS. If you switch off Layer B, then Layer A is played with the instrument’s full polyphony.

Each voice offers two detunable, mixable DCOs with optional key sync (which initialises the waveform when you press the key) and the ability to be disconnected from the keyboard. There’s also a sub-oscillator an octave below osc 1, and a white-noise generator. You can waveshape any of the oscillators’ waveforms (not just the pulse wave), picking an initial shape and modulating it as you choose. In addition, an oscillator ‘slop’ parameter allows you to add errors to the pitches of each oscillator to turn your nice, stable DCOs into horrid, inconsistent VCOs. Oh yes... and you can hard-sync osc 1 to osc 2 for the usual range of effects, which the Rev 2 does rather well.

The output from the oscillator section is then presented to the same Curtis low-pass filter used in the Prophet 08. This offers resonant 12dB/oct and 24dB/oct modes, the latter of which will self-oscillate at high resonance settings. Its character is as you would expect — warm and flexible, but perhaps not as aggressive as some people might like. The filter cutoff frequency is affected by inconsistent VCOs. Oh yes... and you can hard-sync osc 1 to osc 2 for the usual range of effects, which the Rev 2 does rather well.

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its journey through the ADSR stages isn’t long — around six seconds — but the maximum attack, decay and release times are languorous; around 25 seconds for the attack and decay, and around 40 seconds for the release. At their fastest, the A and D times are reassuringlyclicky. You can also modulate the cutoff frequency using the output from osc 1. The possibilities of this are enormous: you can sweep the pitch of osc 1 to create all manner of clangorous timbres and other sophisticated effects and, when the filter is self-oscillating, you can even create 2-op FM sounds this way.

Each voice’s audio amplifier is controlled by a second velocity-sensitive DADSR contour. The amplifier section has just one additional parameter: Pan Spread, which places alternate voices either side of the centre by an amount determined by the parameter value. In addition, there’s a VCA Level (‘Initial Level’) parameter tucked away in the Miscellaneous menu, and this allows the synth to drone if that appeals to you.

The Rev 2 is replete with modulation options, not least of which is an eight-slot modulation matrix that allows you to direct your choice of 22 sources to your choice of 53 destinations, with each slot having an individual, bipolar amount control. Programming this can be simpler than you might imagine — just hold down the Source button and adjust the control that you want to define as the modulation source, and then do the same with the Destination button. This doesn’t provide access to the complete list of sources or destinations, but it’s a quick way to programme the most common ones, and you can access the rest on-screen in the usual fashion.

Four of the modulation sources are LFOs that offer five waveforms, key sync and master clock sync, and you can apply these using their dedicated Destination knob or via the modulation matrix. Their maximum frequencies didn’t seem to reach the 500Hz promised in the specification but, at the other end of the scale, I found that I could force them well below the claimed 0.022Hz for some incredibly slow sweeps. There’s also an auxiliary envelope that can be routed using another dedicated Destination control or through the matrix. You can set this one to loop if desired, which means that you can use it to create unusual waveforms. With the attack, decay and release set to zero, the maximum frequency attainable is around 400Hz, which suggests some interesting possibilities. At the other end of the scale, the lowest frequency I obtained could be measured in minutes/cycle!

Other voicing facilities include individual Program volumes in addition to the overall master volume control, octave transpose, hold, and a flexible portamento (the rate of which can be determined separately for osc 1 and osc 2) with equal rate and equal time modes, and both legato and non-legato options. There are also two pan modes that determine how the sound is affected when pan is a modulation destination.

Dave Smith Instruments
Prophet Rev 2 £1434

PROS

• It retains the character of the Prophet 08 but with much greater power and flexibility.
• It’s small, light and convenient, but feels solid and robust.
• The keyboard feels good, and offers both velocity and aftertouch sensitivity.
• It’s cheaper than its siblings.
• It can sound superb.
• No wall-wart power supply.

CONS

• There’s no polyphonic unison mode.
• You need to use two hands to transpose the sequences.
• There are no Program selection shortcuts.

SUMMARY

Sometimes a polysynth appears that ticks almost all of the boxes and makes the transition from mere synthesizer to musical instrument. For me, the Rev 2 is one of these. Well done, Dave Smith and the team.
The first affects the spread of individual notes, to determine how ‘wide’ the sound is in the stereo field, while the second allows you to move the whole Layer left or right, just as when using the panning knob on a mixer. The Rev 2 also offers the usual range of key modes: low-, high- and last-note priority, with or without multi-triggering, and these are particularly relevant when using the monophonic Unison mode, which allows you to play a single note with up to 16 voices. Strangely, there’s no polyphonic unison that would, for example, allow you to play four notes, each with four voices. But on a more positive note, there’s chord memory and, like other Prophets of recent years, it offers numerous alternative tuning scales that, if you wish, you can replace with alternatives downloaded from the Internet.

The Arpeggiator, Sequencer & Effects

Each Layer includes an arpeggiator offering up, down, up/down, random, and ‘as played’ modes over one, two or three octaves, with each note repeating up to three times, as you choose. Timing can be derived from the internal clock, MIDI Clock, an external trigger source or external audio and, if you want, a neat re-latching feature allows you to hold an existing arpeggio and then, when you play a new chord, create a new arpeggio rather than adding the new notes to the existing pattern. It’s simple, and works well.

There’s also a dual-mode sequencer in each Layer. The first mode offers a single, 64-step, six-voice polyphonic track, and you can play along with this using any unallocated voices. Programming notes, ties and rests is straightforward and, if you make a mistake, it’s possible to edit each note individually. You can copy and swap sequences between Layers, but not to or from another Program, which you might find frustrating. Unexpectedly, you can change Program while a sequence is playing, and it will continue with the new sound. Unfortunately, transposition on playback requires that you press the Record button and the desired key, which is impractical because it can require two hands, so DSI really should find a better way to accomplish this. The second mode is a four-track, 16-step sequencer called the ‘gated sequencer’, and you have to press and hold a note on the keyboard or via MIDI for it to run (hence the name). This allows you to create and replay up to four modulation tracks, although you can make it play notes by routing one or more tracks to the oscillators’ frequencies. Programming is parametric but straightforward; select the desired track and destination and then enter values on each step using the Value knob. The only complexity is encountered when using the Slew function available on tracks two and four, which allows you to apply programmable amounts of glide to the transitions of values in tracks one and three. Interestingly, the four tracks can have different numbers of steps, which mean that you can create sequences that take a very long time to repeat. Whichever mode you use, you save the sequence by saving the Program containing it.

Each of the Rev 2’s Layers has its own effects processor and each of these offers 13 algorithms; three flavours of delay, a chorus, three phasers, two flangers, a reverb, a ring modulator, distortion...
SEPARATE SPEECH AND MELODY FROM A MIX

Isolate speech and melodic content from noise, music and more

Isolate speech and melody from background elements in mono and stereo recordings. Remove noise, improve speech intelligibility in archival recordings, seamlessly clean up dialogue in film post-production without the need for ADR, and much more.

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The ADX Speech Volume Control (SVC) plug-in allows +/- 12 dB of independent volume control over both speech and background elements on a mono or stereo track.

VVC
Volume and pan control of melodies within a mixed song
ADX VVC automatically identifies the main vocal or lead melody line from a mono or stereo mix, then allows you to adjust vocal volume and pan position – all without requiring the original multi-track session!

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and a resonant high-pass filter. Each algorithm offers just two parameters. The first defines the delay time, the modulation rate, and so on, while the second determines the magnitude of the effect. These, together with the wet/dry mix, are destinations in the modulation matrix, so you can do far more with them than you might expect. Of course, these facilities are far from unique to the Rev 2, but there’s something about the way that the controls and menu options are arranged that encourages experimentation. Consequently, I would treat the effects as part of the sound design process rather than as substitutes for external processors. Alternatively, you can defeat them to ensure a pure analogue signal path from the oscillators to the outside world.

**In Use**

Do you remember the famous café scene in *When Harry Met Sally!*? If so, you’ll be able to picture my reaction when I removed the Rev 2 from its box. Yes, yes, yes... YES! Five octaves. Not three, nor four, but five! If I were allowed to level just one criticism at DSI’s recent releases, it would be the widths of their keyboards, which render them of limited use to me without hooking them up to a 61- or 76-note controller. But, like the Prophet 08, the Rev 2 provides a sensible complement of black & whites based upon the high-quality velocity- and pressure-sensitive keyboard mechanism found on the Prophet 6 and O8-6. Mind you, it wasn’t just the width of the keyboard that elicited that response; it was the whole look and feel of the instrument, and its build quality. I’ve loved my Prophet 600 since I bought it in the mid-1980s and, for some sounds, I used it in preference to a Prophet 5 because there was (and still is) something about it that felt ‘just right’. Today, with the exception of the position of the pitch-bend and modulation wheels, the Rev 2 feels much the same to me. At this point, I must admit that I was surprised when I heard that DSI had named their new baby the Prophet Rev 2 rather than the Prophet 16. But when as that of the Prophet 08, to the extent that it’s even compatible with Prophet 08 Programs. For me, this is great, but if you’re not a fan of the ‘08, you might not agree. Anyway... I selected ‘jump to current position’ as my preferred method of using the knobs (it also offers pass-through and relative modes) and, in the absence of a Live Panel mode, invoked the basic template patch, and started programming. I immediately found that the diminutive size of the Rev 2 belies the power within, and was soon programming pads and ensembles of real depth, aetherial patches such as ghostly voices based on filtered noise (the low-pass filters tracked almost perfectly over a wide range), organs that use the filters as the fourth ‘drawbar’ per voice, electric pianos and other keyboards, and even sounds reminiscent of early digital synths. Of course, the Rev 2 also creates superb leads and basses, and does ‘big vintage analogue’, although an overdrive would have extended its capabilities in these areas even further. I also had great fun with the effects, for example, invoking a BBD delay with maximum feedback (so that it looped indefinitely) and using the MIDI Note

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**The Rear Panel**

The Rev 2 boasts independent stereo outputs for Layer A and Layer B, so you can use it as two independent analogue mono/polysynths. If you don’t connect anything to the B outputs, both Layers are output from the A outputs, as you would expect. Next to these, you’ll find a quarter-inch TRS stereo headphone socket that carries the same signal as the Layer A outputs, meaning that, if you have cables inserted into the B outputs, only Layer A is heard, which is a shortcoming. There are three analogue control inputs. The operations of two of these are obvious: Sustain accepts a standard TRS sustain pedal, while Pedal/CV allows you to use an expression pedal or external CV as a source in the modulation matrix. The third — Sequence — can be used with both a pedal or an external analogue signal to provide on/off commands for the sequence and the arpeggiator, or triggers to step through a sequence, or triggers and gates for the synth’s contour generators. Next to these you’ll find a conventional set of MIDI in/out/thru sockets and a USB B socket that carries bi-directional MIDI but not audio. The Rev 2 is USB 2 class compliant, so no drivers are needed. Finally, there’s an IEC socket for the internal, universal (100V-240V, 50Hz/60Hz) power supply.

---

**The gated sequencer window.**

I started to play it, I understood the decision. Despite boasting up to twice the number of voices, waveshaping, an expanded modulation matrix, digital effects and more, its underlying architecture and character is the same as that of the Prophet 08, to the extent that it even compatible with Prophet 08 Programs. For me, this is great, but if you’re not a fan of the ‘08, you might not agree. Anyway... I selected ‘jump to current position’ as my preferred method of using the knobs (it also offers pass-through and relative modes) and, in the absence of a Live Panel mode, invoked the basic template patch, and started programming. I immediately found that the diminutive size of the Rev 2 belies the power within, and was soon programming pads and ensembles of real depth, aetherial patches such as ghostly voices based on filtered noise (the low-pass filters tracked almost perfectly over a wide range), organs that use the filters as the fourth ‘drawbar’ per voice, electric pianos and other keyboards, and even sounds reminiscent of early digital synths. Of course, the Rev 2 also creates superb leads and basses, and does ‘big vintage analogue’, although an overdrive would have extended its capabilities in these areas even further. I also had great fun with the effects, for example, invoking a BBD delay with maximum feedback (so that it looped indefinitely) and using the MIDI Note
number to control the delay time while arpeggiating suitable sounds. This not only changed the repeat speed as I played but also the pitch of the existing notes in the loop, which was fascinating. Imagine playing early Tangerine Dream tracks on a tape machine with an oval capstan while bouncing up and down on a boat in a force 10 storm. Exhilaration or seasickness? You decide.

The Rev 2 sports a genuine bi-timbral mode over MIDI that allows you to play the A Layers of any two Programs using adjacent MIDI channels, with voices 1-8 allocated to the first, and voices 9-16 allocated to the second. This makes it possible to play two sounds independently across the entire MIDI Note range, which is quite different from the splitting and layering available from the keyboard itself. When testing this, I encountered a few minor bugs, but I understand that these are about to be fixed in the latest OS, so there’s no need to recount them here. Oh yes, and I experienced during my time with the Rev 2 was on a single occasion when the oscillator and filter calibration routine froze. Power cycling unlocked it, and I was able to calibrate the synth without ignored. The only worrying moment was far more significant than the bleed. Consequently, I think that this can be appreciated the distinctions between them (see table). With its VCOs, Poly-Mod, mono-timbral architecture and 1970s sound, the Prophet 6 is the model that most resembles the earliest Prophets and will appeal to those who lust after a Prophet 5. In contrast, the digital oscillators and enhanced facilities of the Prophet 12 are more appropriate for players who require contemporary sound design capabilities. Between these sits the Rev 2; it’s more advanced than the Prophet 12. Given its price, I can see it becoming the Prophet of choice for many players. Indeed, if money were tight and I were buying one for myself, the Rev 2 is probably the Prophet I would choose.

Conclusions

The Prophet 6, Prophet 12 and Prophet Rev 2 share many attributes but, having spent considerable time with each, I now appreciate

<table>
<thead>
<tr>
<th>Which Prophet?</th>
<th>Prophet 6</th>
<th>Rev 2</th>
<th>Prophet 12</th>
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<tbody>
<tr>
<td>Timbryality</td>
<td>Mono</td>
<td>Duo</td>
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<tr>
<td>Voices</td>
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<td>8 or 16</td>
<td>12</td>
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<td>4 x digital oscillators offering &gt; 20 waveshapes including noise</td>
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<td>52 destinations</td>
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<td>64-step, 6-voice polyphonic sequencer</td>
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</table>

E Eight Voice £1434, 16 Voice £1859. Prices include VAT.
T Dave Smith Instruments +1 415 830 6393
E mail@davesmithinstruments.com
W www.davesmithinstruments.com
With a name like The Orchestra, it might be tempting to dismiss this as just another orchestral sample library; however, that would be missing out on something quite special. In fact I’ve found it quite hard to stop playing with The Orchestra and actually start writing this review.

First off, I should explain what it is not. It’s not a deeply sampled, highly detailed rendering of every possible articulation of every possible orchestral instrument, recorded in some swish Hollywood studio in umpteen mic positions. At a mere 6GB in size that would be expecting a bit much. What it does manage to squeeze into that relatively small footprint are the essential components of an orchestra, with just enough articulations to cover the most commonly needed performance variations. These instruments are presented as playable patches, but they serve a greater purpose: The Orchestra Engine. Users of phrase-based sample libraries such as the Sonokinetic range (Minimal, Tutti, Capriccio, etc) or Native Instruments’ Action Strings will be familiar with the concept: real, pre-recorded musical phrases or rhythmic patterns that conform to the chords you play on a keyboard. However, the content of those phrases always remains the same. What if you could make your own content? That’s what The Orchestra Engine is all about. The main difference is that there are no pre-recorded phrases in this library — you use the sampled instruments as raw material. We’ll dive into the Engine’s concept and methods in due course. First, let’s take a look at how The Orchestra’s presets are laid out.

Preset Structure

Under the Instruments tab are five folders representing each orchestral section: Strings, Brass, Woodwinds, Percussion and Choir. Strings, Brass and Woodwinds have subfolders for each instrument type in that category, e.g. Violins 1, Violins 2, Violas, Celli and Basses. Within these subfolders are the instrument patches: one that loads all available articulations, the remainder that load only single articulations. For a full count of instruments and their articulations, see the ‘Full Instrument List’ box. The Percussion folder contains four patches: Non-Pitched Percussion, Orchestral Harp, Timpani and Tubular Bells. The Choir category is provided as a ‘bonus’, providing male and female ‘Oh’ and ‘Ah’ sustains, male and female staccatos, each with 11 different keyswitchable syllables, and a sustained ‘Elven Choir’ that morphs through the vowels ‘ooh, aah, eh and ee’. You’re unlikely to render convincing versions of Mozart’s Requiem or Carmina Burana with the Choir, but it’s a useful addition nonetheless. The final item at the Instrument tab’s root level is a patch simply named ‘The Orchestra’, which will be covered in detail later, for it’s here that the magic of this library lies.

Instrument Basics

Instrument patches are very simple: no envelopes, no filters, in fact no clever scripting at all. An EQ with low- and high-boost/cut and a convolution reverb with a choice of 10 impulses and a reverb mix knob are all there is to play with. The ‘All Articulations’ patches add buttons for selecting articulations. These are also pre-assigned to keyswitches C0 to F0 —
this convention is (laudably) consistent across all keyswitchable instruments. One nice feature is that multiple articulations can be layered by pressing two or more keyswitches simultaneously — handy for layering a staccato attack with sustained violins, for example. Articulations common to all the String, Brass and Woodwind are staccato, sustain, marcato and legato. The Strings benefit from two extras: pizzicato and tremolo. Marcatos are all of a usefully extended length, decaying nearly to silence. Sustain, legato and tremolo are all dynamically controlled by the mod wheel; all others are velocity sensitive. Sustaining sounds and staccatos all have three dynamic layers; staccatos have up to five round robins depending on the instrument, whilst non-pitched percussion can have up to eight dynamic layers and five round robins. Sonuscore have clearly worked hard to squeeze as much into 6GB as possible, so there are inevitable compromises — there are no trills, for example — and whilst the legato transitions do a reasonable job of smoothly bias, but cutting a few dB at around 1.4kHz sweetens them up nicely. Similarly, a boost at 1.9kHz gives added grandeur to the french horns — to quote from Bates Motel, “it’s all good, Norman.” The instruments were recorded in their traditional seating positions at Studio 22 in Budapest. Only a single stereo mix for each instrument is provided. There are no alternative mic positions, so the room’s ambience is cast in stone, contributing significantly to the distinctive sound.

There’s no denying that this library has a very distinctive sound of its own. Sonuscore describe it as “a bit rougher and more ‘honest’ than the common orchestral Hollywood sound” — and indeed it does have a certain brashness that some may feel doesn’t sit well with their more polished, ‘specialist’ libraries. An open mind to a different sonic approach is required here. How pointless it would be if all orchestral libraries sounded the same? Sure, the strings have a heavy mid-range bias, but cutting a few dB at around 1.4kHz sweetens them up nicely. Similarly, a boost at 1.9kHz gives added grandeur to the french horns — to quote from Bates Motel, “it’s all good, Norman.” The instruments were recorded in their traditional seating positions at Studio 22 in Budapest. Only a single stereo mix for each instrument is provided. There are no alternative mic positions, so the room’s ambience is cast in stone, contributing significantly to the distinctive sound.

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connecting notes with no uncomfortable gaps, there are no tools for fine-tuning their speed or volume, or adding portamento. Whilst it would take up too much space to describe each instrument in detail, a number of key characteristics are worthy of mention. The sustained/legato Viols and Violas are fairly light on vibrato, especially in the p and mf dynamic layers, but it becomes more apparent in the f/ff layer. Celli and Bass sustains are played in a non-vibrato style. Brass sustains cover a moderate dynamic range with enough gusto to carry a tune, but never quite hitting ff. The staccatos and marcatos, on the other hand, deliver some satisfyingly rude and fruity tones when played hard. Star of the bunch is undoubtedly the Low Brass, having enough Wagnerian heft and lip-flapping rudeness to mark the End Of Days. In the Woodwinds, the Clarinet and Contra Bassoon sustains are played in a typically classical non-vibrato style, whereas the Flute, Oboe and Bassoon bring in vibrato as the note progresses. My personal star of the bunch is the Oboe’s Marcato, with a rather cheeky, animated vibrata — it’s a Venetian masked ball waiting to happen. Combining sustain and marcato articulations works very well with all the instruments; the marcato element lends a positive attack which decays neatly into the sustain element, which can then be further shaped with the mod wheel. Ideal for performing custom-shaped sforzandi.

The Orchestra Engine

On loading The Orchestra patch, the GUI presents a very different face. If you can tear yourself away from playing the immediately absorbing default preset ‘A New Life’ (a Baroque-style string section playing a chugging eighths pattern) you’ll see the Main tab’s screen — one of three — showing five rows and three columns. The concept here is simple enough, but the creative possibilities are, quite literally, endless. In the left-hand column, clicking on any row’s instrument name guides you through an expanding directory of all the instrument categories, instruments and articulations. The idea is to select one specific articulation — of whatever instrument — to fill that particular slot. The middle column has small dials allowing each row to be octave-shifted relative to the others, by up to +/- four octaves. Conveniently, these dials are also MIDI learnable. The right-hand column is where all the action happens; clicking a down arrow brings up one of two menus, depending on whether a row’s assigned sound is sustained or staccato in nature. The choices are one of three arpeggiators for staccato, or one of two envelopes for sustains. ‘You can also choose ‘none’, so that row plays its sound set as it were an ordinary (polyphonic) instrument patch. To fully understand the arpeggiators and envelopes, we need to switch to the Engine page.

Feel The Motion

At the top of the Engine page are five tabs for accessing each of the three arpeggiators and two envelopes. All five of these ‘modulators’ are independent from one another, but the arpeggiators’ editing parameters are identical across all three, as are the ones for the envelopes. The arpeggiators are the most complex, and are divided into two sections: Arpeggiator and Rhythm. Arpeggiator parameters begin with the note order, which includes a wide selection of up/down, zig-zag and inward/ outward movements, as well as whole chords. Playback rate ranges from quarter to 1/64th notes, and notes can be transposed by +/- 12 semitones and +/- two octaves. Notes can be made to repeat up to four times before moving to the next step, and variable amounts of swing can be applied, from zero all the way to a 3:1 ratio (eg. dotted crotchet, quaver). Then comes the clever stuff. Note Selection defines which notes in a chord will be used in the arpeggio. The options are: off (ie. they’ll all play), lowest, lowest two, middle, top two and top. So you could, for example, assign arpeggiator 1 exclusively to play staccato bass strings and/or celli, and have them play their own bass pattern that responds only to the lowest note of a chord. Patterns can also reset, either to every bar or every second bar, in a choice of time signatures (compound and odd signatures are all available). The Rhythm stepper controls the tempo of each step and sequences can be up to 32 steps long. It follows the standard bar graph format — just alter the height of each bar to change the velocities. There’s also a handy tempo multiplier to halve or double the speed of both the arpeggio and rhythm stepper. The potential variations are staggeringly huge when you put all of these functions together — and then add the other two arpeggiators into the equation.

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a customisable, repeating dynamic curve to sustained sounds. Drawing curves is straightforward, just drag the mouse freehand across the bar graph or right-click to draw straight lines. The cool part is that the overall dynamic of the shape you’ve drawn responds to the mod wheel, so you can manipulate its strength on the fly. The time scale can be set to run from between half a bar to four bars, in compound or odd time signatures, and like the arpeggiator, Note Selection specifies which notes of a chord will sound. Notes can either retrigger at each repetition of the envelope cycle, or carry on playing in a continuous loop. Once you’re done setting up all the instrument assignments, arpeggiators and envelopes, your creation can be saved as a new Orchestra .NKI preset. Bear in mind that snapshot presets are specific to a particular .NKI patch, so if you do save your work as a snapshot you must also save the whole Orchestra instance (preferably as a new .NKI), otherwise your snapshots will vanish into the aether.

**Orchestra Presets**

The presets that come embedded within The Orchestra’s .NKI should not go without discussion; they’re entirely responsible for my finding it hard to stop playing and start writing up this review. If there was a CCTV camera monitoring my studio, you’d have seen me with a constant gormless smile on my face. These presets can be loaded either via the snapshot menu or the larger drop-down menu on the Main screen’s GUI. They fall into three categories: Orchestral Colours (60 presets), Orchestral Rhythms (60) and Animated Orchestra (30). The Colours presets are the simplest, being pre-made layers of instruments in various combinations, and make no use of the arpeggiators, envelopes or tempo-sync’ing. Rhythm presets comprise a wide range of arpeggiated staccato patterns in various time bases, ideal for providing a driving backbone to any arrangement. There’s a lot in common with NI’s Action Strings here. Patterns focus both on instruments from a single orchestral section, or combinations of instruments from different sections. The jewels in the crown that afford such endless inspiration and entertainment, however, are the Animated presets. These are mini-arrangements in their own right, featuring a mix of sustains and staccatos from different instruments, frequently making use of all the arpeggiators and envelopes. Titles such as ‘Driving Force’, ‘Icy Lake’ and ‘Building An Empire’ suggest their general mood — they’ll doubtless inspire different reactions in different people.

**Mixer Page**

Five simultaneous layers of noise need some form of management. The mixer page deals with the volume and panning of each layer, with essential solo and mute buttons so you can check on what each layer is doing. Three effects act globally on the whole mix: EQ with two selectable fixed curves, Compressor with threshold and gain controls, and convolution reverb.

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**Full Instrument List**

<table>
<thead>
<tr>
<th>Strings</th>
<th>Staccato</th>
<th>Sustain</th>
<th>Marcato</th>
<th>Legato</th>
<th>Tremolo</th>
<th>Pizzicato</th>
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<tbody>
<tr>
<td>Violins 1</td>
<td>X</td>
<td>X</td>
<td>X</td>
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<td>X</td>
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<td></td>
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<tr>
<td>Basses</td>
<td>X</td>
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<td>X</td>
<td></td>
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<tr>
<td>Men Sustain</td>
<td></td>
<td>Ah &amp; Oh keyswitchable vowels</td>
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<td></td>
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<tr>
<td>Men Staccato</td>
<td></td>
<td>11 keyswitchable syllables</td>
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<tr>
<td>Women Sustain</td>
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<td>Ah &amp; Oh keyswitchable vowels</td>
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<td>Women Staccato</td>
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<td>11 keyswitchable syllables</td>
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<tr>
<td>Percussion &amp; Harp</td>
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<td>Plucked, three velocity layers</td>
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<tr>
<td>Concert Harp</td>
<td></td>
<td>Hits: three velocity layers, five x RRs; Tremolo with mod wheel dynamics</td>
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<tr>
<td>Timpani</td>
<td></td>
<td>Hits: two velocity layers, two x RRs</td>
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<tr>
<td>Tubular Bells</td>
<td></td>
<td>Swells: six variations; Rolls with mod wheel dynamics; Hits – three velocity layers</td>
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<td></td>
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<tr>
<td>Suspended Cymbals</td>
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<td>Hits: five velocity layers, five x RRs; Rolls with mod wheel dynamics</td>
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<tr>
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<td></td>
<td>Hits: five velocity layers, five x RRs; Rolls with mod wheel dynamics</td>
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<td>Snare Drum</td>
<td></td>
<td>Hits: eight velocity layers, five x RRs</td>
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<tr>
<td>Tam Tam</td>
<td></td>
<td>Long Hits: eight velocity layers, five x RRs; Muted Hits: four velocity layers</td>
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<tr>
<td>Piatti</td>
<td></td>
<td>Hits: four velocity layers, five x RRs</td>
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<td>Drum Ensemble</td>
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<td>Hits: four velocity layers, five x RRs</td>
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<tr>
<td>Taikos</td>
<td></td>
<td>Hits: four velocity layers, five x RRs</td>
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</tr>
</tbody>
</table>
The Mixer page, with controls for the level, pan, reverb send level, solo and mute status of each layer. Reverb is the only effect engaged here.

with a choice of 10 impulse responses. Each layer has its own reverb send level, and there is a master reverb return level for the whole ensemble. All the mixer controls are MIDI learnable (bar the solo and mute buttons) so you can easily control the density of activity, bringing instruments in and out on the fly.

Multis

There’s nothing to stop you loading several instances of The Orchestra at once to create ever more complex interactions and layers of sound. The 15 Multis demonstrate this well, occasionally employing four instances to do the job. There’s a point of diminishing returns at play here — sometimes there’s so much going on it’s overwhelming — but the potential is demonstrably clear. It’s up to the user to exercise common sense and good taste!

Conclusion

Whether or not you approve of this sort of phrase-based orchestration, it has to be conceded that The Orchestra is totally absorbing, and more fun than a fun thing. I even like its tonal character — quirky, brash and occasionally unsubtle — yet it has an attitude that evades some other orchestral libraries. Traditional orchestrators might baulk at using it, especially in a situation where a playable score has to be produced — imagine the raised eyebrows of the poor copyist faced with the prospect of unravelling what they’d consider an undisciplined sonic fur ball. But for purely MIDI musicians, for whom the means is irrelevant and only the end result matters, The Orchestra will score highly (pun intended). It may require a little more forward planning than the aforementioned Sonokinetic libraries and their ilk, but it’s all doable. Ultimately, the sheer joy of this library in contrast to ‘fixed phrase’ libraries is that you can create your own content, and make it uniquely yours.

The only challenge now is for me to stop playing around and write some music with this thing!
Kesha Lee at LA's Conway Studios.
This is the 132nd article in Sound On Sound’s Inside Track series, and it’s a sad reflection of the state of the music business that only three female engineers have featured in the first 131. What’s more, persuading the fourth to take her turn in the spotlight proved surprisingly difficult. Kesha Lee explains that she is uneasy about doing an interview because she’s only been doing this job for four years, and that “as an engineer coming up in Atlanta, the setup or process is a lot less involved gear- and session-wise than it may be in other places”.

In that time, however, Lee has worked with the likes of God 1st †, Gucci Mane, Travis Scott, Playboi Carti, Future, Pharrell Williams, TI, Jeremih, Young Thug, Migos and Chance The Rapper — and now has a US number one album, courtesy of Luv Is Rage 2, the debut album by Atlanta rapper Lil Uzi Vert. It took a bit of persuading to make clear that having engineering and mixing credits on a chart-topping album make Lee a perfectly valid interview candidate! Eventually a Skype conversation ensued, in which Lee shared her story and that of the making of Luv Is Rage 2.

Second Chances

Born 28 years ago in Alabama, Kesha Lee’s first musical exploits involved playing violin at middle school, and later playing the clarinet. Her father was into music, and encouraged her to play the piano, and to get to grips with the modern world of DAWs, even going as far as to buy her a laptop with Pro Tools. However, they both found it rather hard to make sense of Pro Tools. Other DAWs were tried, but Lee drifted into non-music jobs, like working in restaurants.

“I really wanted to work in music,” Lee recalls, “but did not know where or how. I was friends on Facebook with someone who was a radio personality at the local radio station, 98.7 Kiss FM in Birmingham, Alabama, so I asked whether they were hiring. This led to me doing an internship and later being offered a job for one day a week. I ended up cutting in commercials using Adobe software. People’s voices were sped up for the commercials, and then pitched down again, so I learned about that. I realised that I didn’t really want to continue with broadcasting, because I was in a room by myself all the time, and I looked into audio engineering. Around that time my mother bought a Mac for my brother, and I started playing around with GarageBand on that, and it was much more simple and straightforward to use than the programs my Dad and I had been working with.

“What attracted me to engineering was to be able to add all these effects to audio. I did not know that you could do that, but GarageBand really got me into that. So I started looking around for a school, and ended up doing an 11-month audio engineering course at the Atlanta Institute of Music. In my last quarter there I began sitting in on Gucci Mane’s sessions with his engineer Sean Paine, at Patchwerk Recording Studios, and also on Future recording sessions with Seth Firkins. I wasn’t really an engineer yet, but Seth invited me to sit in and taught me a lot of stuff. Once I graduated he figured that I should, instead of sitting in, be working, but I’d still regularly go round and he’d allow me to practice on his setup and ask questions about things I didn’t understand.”

Social Scene

After graduating from the Atlanta Institute of Music in September 2012, Lee continued to benefit from the fact that the Atlanta scene is exceptionally vibrant, with new rappers regularly breaking through nationally and internationally. At the same time, the scene is also small enough for everyone to know everyone, and hanging out on each other’s sessions is the norm. In fact, Atlanta recording sessions are social gatherings, worlds away from the hushed and almost sacred ‘don’t disturb’ atmosphere that has often been the norm in recording studios. This even leads to the ironic situation of Lee regularly having to don headphones while recording a rapper who is in the booth, because the guests in the control room are making too much noise!

“Yes, everything in the studios in Atlanta is really chilled and vibey. There never really is a problem with artists and...
producers popping in on each other’s sessions and stuff like that. The rappers and producers are for the most part very open-minded, and don’t have a problem bouncing ideas off each other. They also like to work very fast and as an engineer in Atlanta you have to keep up with them. You can’t be too concerned and technical about signal chains and sound customisation. You may want to switch the microphone for another rapper, or try a different vocal chain, but they don’t care about that, they just want to get their ideas down. And yes, if there are a bunch of people in the room, and they’re watching a game and don’t want to turn it down, and I really need to hear what the rapper is doing, I will put on my Sennheiser HD280 headphones.

“I haven’t actually encountered any problems working as a woman in the studio, because I’m at the desk, focused on a screen, and everybody is behind me. People don’t really talk to me when I am working. There have been two or three instances when people were offended to find a girl engineer, and assumed that I didn’t know what I am doing — but they came round. I have the hardest time with people who know how to use Pro Tools, because instead of telling me what they want, they tell me how to do it, which takes me out of my workflow. There are three ways to do everything in Pro Tools. I know many hotkeys and am really fast with them. So it is: ‘Just tell me what your end goal is!’”

On The Road

The social and interactive nature of the Atlanta rapping scene helped Lee to quickly gather a very impressive list of credits after graduating, as mentioned above, which also includes being a recording engineer for well-known Atlanta beatmakers like Metro Boomin and Zaytoven. A meeting with Atlanta beatmaker and producer Don Cannon, who is close to Symere Woods, aka Lil Uzi Vert, resulted in Lee working regularly with the rapper. Since 2015, Lee has engineered all three EPs and four mixtapes Uzi Vert has released to date, as well as Luv Is Rage 2. She also mixed four songs on the rapper’s debut album; the others were mixed by well-known mixers Jaycen Joshua and Leslie Brathwaite. However, as we shall see, and as Lee herself already hinted at, being an engineer and mixer in the Atlanta rap world is a rather different proposition from most other places.

Preliminary work for the album took place in December 2016, with the core team of Uzi Vert, Don Cannon and Lee. Next up was a trip to Hawaii at the beginning of 2017, suggested by the label, Atlantic. “The first days after we arrived we were jetlagged, and the studio where we were based to work, Island Sound, was rather far from the house where we were. So we ended up just setting up a work station in the house. The audio wasn’t the best, but the vibe was great.

“We were supposed to be in Hawaii for a month, but only stayed for two and a half weeks. After that we spent a couple of weeks in LA, and then we came back to Atlanta, where we worked on the album until August, at Means Street Studios. It was full-time. My schedule was insane. The last two months I had two days off each month, and one of the days was because I was moving into my apartment, and the other day I still had to work. There were times when I didn’t sleep for days! Throughout, the only times we weren’t working was when Uzi would do a show. Those times gave me a chance to watch some ideas I had for the songs, editing certain sections and mixing, but after a while I stopped, because I wasn’t sure what songs he was going to pick.

“The reason the recordings took so long was because it was Uzi’s first album, and he wanted to top his last body of work. He knows what he wants with regards to the beats he raps over, but we often can’t figure out what sound he is going for. So there was a lot of recording, and at some stage the songs he’d recorded while back were old to him, and he wanted to record new stuff. Sometimes he uses a beat as it is, sometimes he’ll like a certain part of the beat and he will ask for it to be looped, so there was a lot of arranging the tracks how he wanted them. Or I’d do an effect that he would like, and he would want that for the song. Or we would have a producer come in and make a beat on the spot. There were a couple of occasions where a producer would come in and Uzi would help them make a beat, or he’d do the drums. I think he did the drums on ‘Twisting’, the album’s opening track.”

Starting With The Beat

Rewinding a little, Lee elaborates on the process by which Uzi Vert, Cannon and she created the album. “People either email Uzi beats, and he forwards them to me, or I will reach out and say ‘Hey, we are in the studio, can you send some beats?’ I will pull up these beats, and often he wants to hear everything before he picks something. He doesn’t really like listening to each beat in too much detail before he goes into the booth, he just goes in and records. He doesn’t do like a whole verse, instead he’ll do things line by line, and I’ll be punching in.”

Project Management

One of the biggest challenges that faced Kesha Lee during the eight-month production process for Luv Is Rage 2 was organising, saving and archiving everything. “This was my first album project from beginning to end, on which I was both a recording and a mixing engineer, so I had not handled a project of this size before. Each day I created a folder with the date and the time, and inside each folder we would have the name for each song, the name of the producer, and Uzi’s name. If the song wasn’t finished, I would mark what he had done, and I soon also put the tempo and the key in the file name, because they often ask me for that, and I did not want to have to open the session just to get that information. At the end of the month I would have organised the sessions into finished, unfinished and collaborations. Because often Uzi would say something like, ‘Can you put up that song that I did not finish with Metro?’ It was just easier to find stuff that way.

“I would do the best rough mix that I could do while recording, because I knew that we would be moving onto another song, so I tried to do as much as I could during the sessions. If I went back to a song it was because I had an idea for something I could fix or improve, but not only would I never be sure whether a song would be used or not, Uzi would also listen to the songs a lot, and he’d get used to hearing it a certain way, so I had to be very careful about changing things. Plus he did go in again to change certain songs, because he had new lyrics or wanted to add stuff.

“If I had the track-out, I might mute a certain instrument, or I’d chop the beat up. I like doing snare hits or duplicating the kick or 808 in certain spots, or use SoundToys to change the sound. I’d always make notes of what I did, so I could replicate it, if required. We did get the track-out for all the songs, but in some cases the track-out was not the same as the two-track, and because Uzi gets so attached to the version of the song he’s worked with, we then simply mixed using the two-track. Also, even if I did mix with the track-out, I’d try to match that to the two-track.”
The producers send two-track MP3s, and depending on who the producer is, and if Uzi really likes the song, I will get the track-out. But I don’t want to get too far into editing a two-track if I later have to replicate those edits on a track-out.

“Uzi will have all his friends in the studio, but just the three of us work on the music. Cannon makes beats or he will play more of an executive producer role, where he gets beats from other people. Uzi does his own thing. I will even let him do his own drops, where you only hear him and there’s no music playing. I will put the track on latch automation, and I will let him press the mute button so he can do his drops when he wants them, and then when he clicks mute again, the track will come back in. He will do this totally intuitively. Many of the drops will come in on the one, but if the latch automation didn’t catch that on the one, I will go back in later and I will move everything to where it is supposed to be, so the drop is clean.”

Obviously, ‘track-out’ is the Atlantan studio word for exporting all the individual tracks or stems from a DAW recording session. Working with just the two-track MP3 instead greatly simplifies Lee’s job.

It turns out that the Atlantan approach to recording is unusually straightforward in other respects as well, at least in a rap environment. “The standard for Atlanta seems to be to use Yamaha NS10s or Augspurger monitors,” Lee explains, “and the recording signal chain consists of an Avalon VT-737sp mic pre and a Tube-
Tech CL 1B compressor. The microphone is normally a Neumann U87 and the headphones are Sennheiser HD280s. We briefly used the U87 when we got back from LA, because that’s what’s in Means Street Studio A, but Uzi likes the Neumann TLM103, which is in Studio B, and which we use most of the time."

Lee did not record any musical instruments on the album, apart from an electric guitar overdub to one song called ‘The Way Life Goes’, during a session at Germano studios in New York. An almost obligatory addition to any rap album these days appears to be the involvement of a few celebrity singers, with Pharrell Williams and the Weeknd particularly popular. Both feature on Luv Is Rage 2; the Weeknd sent in his vocal contribution, while Williams’ vocals were recorded during the lay-over in LA. Lee had expected to just be watching Williams’ regular engineer record him, but instead she ended up behind the controls at Lee’s mixes for Luv Is Rage 2 were done entirely on her laptop, often with just the built-in audio I/O and a pair of Sennheiser headphones. This photo was taken at Means Street Studios in Atlanta.
Keeping It Simple

Apparently, the intention was for Lee and Don Cannon to do all the final mixes, but time pressure meant that many of the album songs were given to Joshua and Brathwaite to mix. However, since Lee’s mixes were very largely based on the two-track file of the original beat, her mix work involved essentially doing a vocal mix and balancing that the beat. Lee in fact conducted many of her mixes at home, using just the basic Mac soundcard and her $99 Sennheiser headphones.

“I use a three-year old MacBook Pro, and Cannon recently bought me the UA Apollo Twin, so I want to get into using that. In general, I want to be able to be completely work on my laptop wherever we go, because even the Pro Tools systems in the bigger studios don’t have certain plug-ins. I also don’t use the desk for anything more than the volume button for the monitors and the switch for speakers. I’ve tried mixing during the sessions, but there’s so much going on that it’s hard to concentrate and really listen to things. So I prefer to mix in my own space, away from people. And I figured, most people listen to music on their laptops or phones using earbuds, so why not mix how they are going to hear it? I will check my mixes on the speakers in the studio, but I am really comfortable with my Sennheiser HD280s.”

Talking Mixing

The mix session for ‘How To Talk’ is modest by modern standards, containing only 33 tracks — of which almost half were not used. These break down in the stereo mix of the beats at the top, 14 beat audio tracks, one beat aux, 10 audio vocal tracks, two vocal aux tracks, three effect aux tracks for the vocals (delay, fast delay, reverb), a mix print track and a master track.

Lee: “The two-track at the top is colour-coded purple and green, for organisational purposes, so it’s really easy to spot where the hook is. The hook is always green. This also helps Uzi, so he knows where the hook ends and where the verse comes in. The two-track has three plug-ins. The ‘Q’ is the Avid EQ3 seven-band, which I used to turn down the volume of the track. The ‘V’ is the Waves Vitamin Sonic Enhancer, and the ‘P’ is the UAD Precision EQ, which was added by Cannon. There is not much I can do to the two-track, but I added the Vitamin to play with some of the frequencies and bring out the highs a little bit. Below the two-track are the track-out tracks, all in blue. We did not go with that track-out, so I muted the aux for it.”

Vocals

“The intro track was a voice note Uzi had recorded on his phone, and he played it in Nomad Factory’s MCL-2269 Limiter & Compressor was used to warm up the beat, along with RN Digital’s unusual Detailer plug-in.
default, with a Retune speed of 20, but lately he has been like: ‘Give me more Auto-Tune!’ so now we have the Retune Speed set to anywhere from 12 to 5. The ‘D’ after Auto-Tune is the Waves De-Esser, the ‘Q’ the Avid EQ3 seven-band, and the ‘1’ is the Waves C1 gate. All vocal audio tracks also have sends to the delay and the reverb aux tracks. The delay aux track has the Avid Mod Delay II set to half notes, with feedback at 43 percent, the Waves Renaissance Reverberator, set to ‘Hall 1’ reverb, with the highs cut on the reverb EQ, and the Waves S1 stereo imager. The reverb aux has the Renaissance Reverberator.

“All audio vocal tracks go to the vocal aux track. I had two vocal aux tracks in this session, because I wanted to try something different, using plug-ins I don’t normally use. That’s why one of the aux tracks is muted. The vocal aux track that I did use has the Waves De-Esser acting around 4230Hz, then the EQ3 seven-band which has a high-pass at 96.4Hz, and I’m dipping out muddiness at 200 and 500 Hz. I’m also adding some high end at 6.52kHz. I don’t normally add EQ with the seven-band, but Seth would add some high end on Future’s voice and that worked well, so I tried it here. Next is the Waves Renaissance Compressor, to keep the dynamics in check, and then the Waves SSL E-channel, on which I am again dipping out various frequencies. The latter plug-in is more for colour and character. The Waves CLA-3A is more for the sound, and the RN Digital Detailer made Uzi’s vocals sound fuller and wider. The final plug-in in the chain is the Nomad Factory MCL-2269 limiter and compressor, again for the sound and for more volume.

“We always go for a warm, full, loud, in-your-face vocal sound, also because we like the vocals to be louder than the beat. I always turn the beat down 1-2 dB. There are no plug-ins on the master track, because I used to work for a producer who didn’t want that, as the mix would go to the mastering engineer. So I’m still used to doing it like that. I turn the master volume down anywhere between -7 to -9 dB before it goes to mastering, so they have room to work with.”

**Not A Bad Start**

Luvs Is Rage 2 was released on August 25th, and debuted at the top of the Billboard 200 chart. Despite the fact that this surely is a life-changing event for a young engineer and mixer only four years into her career, Lee appears remarkably modest about her achievement, preferring to focus on the fact that she still has a lot to learn. “Of course I think it’s cool, and it’s exciting to have worked on something that many other people like. But it’s mostly others who get excited about it. They make you realise that having a number one is a big deal.

“I freelance, so when Uzi goes out promoting the album I continue working, but I also am taking care of stuff I’ve neglected while I was working on his album. It would be easy to just continue working as I have been here in Atlanta, but actually, I still have a lot of growing to do. Mixing kind of fell into my lap, and because of what happened I want to learn more about it and get better, but I actually love recording, and also want to get more into editing. I’m also really excited to go to the AES in New York this Fall, and I want to find new gear and try out more microphones and different preamps. So mostly I’m getting my life organised and getting ready for all these upcoming trips. And when Uzi gets back, I’ll be working with him again. He spoke of a few projects he wants to do.”

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**www.soundonsound.com / December 2017** 173
Warm Audio's new mic is claimed to offer the sound of an original Neumann U87 at a fraction of the price. Does it succeed?

**PROS**
- Affordable.
- A good-sounding, all-round studio microphone with a sound character that recalls a brighter U87.
- Comes with a nice box and a comprehensive selection of accessories.

**CONS**
- If you want something that sounds exactly like a vintage U87, you’ll have to fork out for a vintage U87.

**SUMMARY**
Warm Audio have issued a hostage to fortune in marketing the WA-87 as a precise copy of the vintage U87; it’s perhaps best to ignore the hype and think of it as a versatile and affordable studio all-rounder in its own right.

Demand for items of classic studio hardware is high and supplies are low, meaning that the originals are out of reach to many home and project-studio owners. This situation has created a niche for companies like Warm Audio and Golden Age, who use low-cost Far Eastern manufacturing to bring replicas of this gear to market at much more affordable prices. Thus far, Warm Audio have focused mainly on studio outboard, but the WA-87 is their first foray into the world of microphones.

There are no prizes for guessing which classic item of studio hardware the WA-87 recreates. Introduced in 1967 as a solid-state successor to the equally classic U67, the Neumann U87 is a microphone that properly
deserves the term ‘iconic’. It is still a market-leading product 50 years later, but today’s U87Ai is the result of a 1986 design revision, and differs from the original in various respects that some people think are significant.

The original U87 used the 48V phantom power supply to polarise the capsule directly, and as a consequence, the electrical design of the capsule had to be modified slightly compared with that of the U67, to isolate the front and rear backplates. This revised capsule was called the K87. However, in the U87Ai a DC-to-DC converter was added, which allowed Neumann to revert to using the original K67 capsule found in the U67 (which was redesignated the K870), and also to raise the polarisation voltage to 60V, delivering a 3dB improvement in signal-to-noise ratio and a hotter output.

Some feel that in making these modifications, Neumann compromised the sound of the original U87, and there are those who say that the Ai version is slightly harsh or brittle in comparison. This point of view is by no means universally held, but Warm Audio have chosen to produce their own recreation. It too is made in China, but under the important-sounding Germanic brand name of Lens Kondensator, and it apparently features a diaphragm made from “new old stock Japanese mylar”. Naturally, they have also closely recreated the original U87 circuit, using a Cinemag USA transformer and what are described as “all discrete, premium components” (since the U87 itself has never used any integrated circuits, it’s hard to see how else you could do it!). There is, though, a noticeable difference in sensitivity between the WA-87 and the U87, with the Warm mic’s output being several dB hotter.

87 Varieties

Until last year, I was the proud owner of a very nice-sounding original Neumann U87. However, pride comes before a fall, and owning vintage mics can be an expensive business. As is not unusual in a 40-year-old mic that has seen plenty of use, the capsule failed, and on the advice of Funky Junk’s mic repair team, I opted not to have it replaced with a new K87 from Neumann, but with an older K87 re-skinned by Thiersch Elektroakustik in Germany. This was a significantly less expensive option, but has undeniably changed the sound of the microphone: it’s brighter and perhaps superficially more exciting than it used to be, but I’m not convinced I like it as much.

For comparison, I also had available an unmolested U87 and a Neumann U77. The latter is a much less common copy the one-inch, centre-terminated, dual-backplate capsule Neumann developed for the U67 and related microphones: the design has been a staple of Far Eastern mic manufacture ever since companies like Rode first brought affordable capacitor mics to market in the ’90s. However, what some manufacturers failed to account for was that the frequency response of the capsule itself shows a strong high-frequency pre-emphasis. In Neumann’s own mics, this is tamed electrically to deliver a broadly flat overall frequency response, but many early Chinese-made mics used simpler electronics that failed to equalise the HF lift, and thus gained a reputation for sounding harsh and over-bright.

Although there are plenty of cheap K67 and K87 replicas available off the shelf, Warm Audio have chosen to produce their own recreation. It too is made in China, but under the important-sounding Germanic brand name of Lens Kondensator, and it apparently features a diaphragm made from “new old stock Japanese mylar”. Naturally, they have also closely recreated the original U87 circuit, using a Cinemag USA transformer and what are described as “all discrete, premium components” (since the U87 itself has never used any integrated circuits, it’s hard to see how else you could do it!). There is, though, a noticeable difference in sensitivity between the WA-87 and the U87, with the Warm mic’s output being several dB hotter.
The Real Thing

There is plenty of demand for original U87s, to the point where second-hand prices are noticeably higher than those commanded by the current U87Ai version. Depending on condition you can expect, at the time of writing, to pay anywhere from £1500 to £2000 or more for a U87 in good working order. The mic is relatively easy to work on and all components are available as spare parts, but they are not cheap, so buying a broken one in order to have it fixed up is not always an economic alternative. The inclusion of non-original parts, especially a third-party or re-skinned capsule, will have a significant impact on value.

If you are comparing the relative costs of the WA-87 and the U87, one factor that needs to be taken into account is the supplied accessories. Warm Audio’s version includes a very nice wooden box, plus equivalents of the Neumann EA87 elastic shockmount and SG287 swivel mount too. Most of the U87s you see on the second-hand market don’t come with the Neumann originals, which are notoriously pricey to buy separately, and are only included with a new U87Ai if you buy the more expensive Studio Set package.

On acoustic guitar, for example, I might actually prefer the added gloss of the WA-87 over the bite of an original U87...”

“On acoustic guitar, for example, I might actually prefer the added gloss of the WA-87 over the bite of an original U87...”

or are there still aspects of Neumann’s own manufacturing process that no-one has yet successfully replicated? My money’s on the latter, but it would be fascinating to hear how the WA-87 would sound if it could be fitted with a new Neumann K87.

I should also point out that there is some variability even within the sound of new U87s. The frequency response of the K870 capsule is specified within a tolerance of ±2dB, so in theory, two Neumann capsules could differ by as much as ±4dB across the frequency range and still be within factory specs.

Does It Matter?

Everything about the WA-87’s marketing demands that it be judged primarily on how well it mimics the Neumann U87. You could argue that this just sets the WA-87 up to fail, because it challenges everyone to pit it against the original in exactly the way I’ve done. Anyone who compares the two on enough sources will surely hear a difference sooner or later, and the WA-87’s positioning as a U87 clone means that any difference we do hear will be experienced as a negative. This positioning also makes me a little uncomfortable from an ethical point of view. It’s one thing to clone a Pultec or Fairchild, where the original manufacturers are long gone and the designs are more or less in the public domain, but quite another to ride the coat-tails of something that remains a market-leading product with only minor variations.

However, if you can free yourself from the mindset that sees the WA-87 as good or bad only inasmuch as it matches or doesn’t...
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Dynaudio, the Danish manufacturers of high-end loudspeakers for the home audio, professional studio, automotive and multimedia markets, are celebrating their 40th anniversary this year. Rooted in the world of hi-fi, it wasn’t until 1992 that Dynaudio launched their Professional division to develop monitor loudspeakers for the recording studio and broadcast markets. Dynaudio’s professional monitors are highly regarded worldwide, particularly the BM series which, for many years, have been (and remain) extremely popular in a wide range of applications.

Dynaudio are not a company that rests on their laurels, and April 2016 saw the release of the first models in the all-new Lyd range of active near/midfield studio monitors: the two-way Lyd 5, 7 and 8. This line-up has now been augmented with the recent release of the range-topping Lyd 48, a compact 3-way active studio monitor built on the same basic platform as its two-way siblings.

**What You See**

When I described the Lyd 48 as compact, I wasn’t joking! Dynaudio have managed to shoehorn a full three-way, active monitor into an enclosure with exactly the same overall dimensions as that of the Lyd 8. Since this ‘compression’ results in the Lyd 48’s orientation being landscape rather than portrait, the fronts are mirror-images of each other, so there are left and right versions of the loudspeaker. The front baffle, which is available in either the “studio standard” basic black or in gleaming white (which I really like the look of), carries the same eight-inch woofer as the Lyd 8, a four-inch mid-range driver sitting in its own separate, sealed enclosure, and the 1.1-inch soft-dome tweeter that is common to all Lyd models. As with the others in the range, the Lyd 48 sits in a bass reflex enclosure, whose port exit is a slim, vertically oriented rectangle.

**Three-way Active Monitors**

Dynaudio’s Lyd series of nearfield monitors has been a great success — and now the range has a new, three-way flagship model.

### Dynaudio Lyd 48

**Price:** €2637

**Pros:**
- Capable of delivering an accurate, detailed and revealing audio performance across all genres of music.
- Can create an expansive deep and precise stereo soundfield.
- Easy to work with over long periods.
- Physically compact with a compact footprint.

**Cons:** None, unless you don’t like three-way monitors.

**Summary:**
An impressive, compact three-way monitor that delivers an assured audio performance at an attractive price.
Dynaudio Lyd8 12.17 layout;8.indd   181

running (because of the change of orientation) the full height of the bass side of the rear panel, and through which the flare of the port itself can be seen. The rear panel carries the same inputs and controls as the others in the range, and here you’ll find the IEC mains input and power switch, the auto-standby mode selector switch, the analogue-only inputs (on balanced XLR and unbalanced RCA), together with four switches that modify the loudspeaker’s response. Dynaudio have taken the view that, in reality, there are only a few adjustments that can, or should, be made to the response of a loudspeaker in situ and as a result the entire Lyd line presents the user with a limited number of DSP-driven options.

The first switchable option is [input] Sensitivity that allows you to optimise the gain staging into the loudspeaker in 6dB steps between 0dBu and +12dBu. The remaining three switches affect the Lyd 48’s frequency response. A Bass Extension selector sets the limit of the loudspeaker’s bass frequency response in a 10Hz step above and below 40Hz. Studio engineers typically work at relatively moderate levels of 70-85 dB SPL, and this allows the Lyd 48 to achieve its quoted 32Hz lower limit at the expense of a -5dB reduction in its maximum SPL of 112dB. Increasing the playback level may require that this maximum bass extension be limited in order to avoid overdriving the woofer itself. This extension-restriction facility also helps the loudspeaker to meet its design criterion of delivering a consistent tonal balance at the loudspeaker to meet its design criterion of delivering a consistent tonal balance at the loudspeaker to meet its design criterion of delivering a consistent tonal balance at the loudspeaker to meet its design criterion of delivering a consistent tonal balance at the loudspeaker to meet its design criterion of delivering a consistent tonal balance.

The Sound Balance switch acts in a similar fashion to the Tilt control found on the long-discontinued Quad 34 hi-fi preamp, where one knob would tilt its frequency response around a central, pivot frequency. In the Lyd 48, the knob becomes a two-position switch whose Bright position puts in a boost of 1.5dB at 20kHz and a cut of 1.5dB at 20kHz via linear-phase filters to avoid phase problems at the listening position, and its Dark position swaps around these boost and cut frequencies. If the Lyd 48 is positioned less than 50cm from any wall, the Position switch’s Wall option applies filters that (in Dynaudio’s own words) “will help with anomalies created by reflections coming off the wall, especially in the lower frequencies.”

The Sound Balance filter, if necessary, to tilt the frequency response tailoring outlined above. With the Lyd 8, the Lyd 48’s tweeter and woofer are driven by 50W and 80W amplifiers, respectively, with an additional 50W amplifier to take care of the mid-range driver. The frequency response of the Lyd 48 is quoted (unfortunately without qualification) at 32Hz to 21kHz.

The Lyd 48’s crossover frequencies are set at 490Hz and 5.6kHz, which requires the mid-range driver to cover a far wider area of the audio frequency spectrum than I’ve come across in other three-way designs. The ability of the Lyd 48’s mid-range driver to cover this wide frequency band would appear to be largely down to its MSP (magnesium silicate polymer) cone, given that the two-way Lyd 5’s five-inch bass driver crosses over into its tweeter (which you’ll remember is identical to that in the Lyd 48) at 5.2kHz.

Dynaudio’s proprietary composite MSP membrane combines low mass, high rigidity and ideal internal damping properties and allows the distinctive, instantly recognisable cone geometry that characterises the company’s bass and mid-range drivers. As regular readers may recall, my colleague Phil Ward reviewed the Lyd 5 and 8 in the November 2016 edition of SOS and he went into some detail on the construction and underlying theory behind Dynaudio’s cone-based drivers. If you’re interested in that aspect, I can recommend not only that you read that section of design, development and build of their loudspeakers carries right through to the Lyd-series owner’s manual, where you’ll find detailed instructions on how to them set up prior to listening.

If you’re going to match Dynaudio’s commitment to helping you get the positioning and orientation of your Lyd 48 monitors spot on, you’ll need a tape measure to ensure that the listening position is no more than 2m from the loudspeakers, string and a marker pen to help create the ideal equilateral triangle spacing between listening position and monitors, and an iPhone or iPad running the Dynaudio Meter iOS app.

The Dynaudio Meter offers an RTA (Real Time Analyser) display, a sine-wave/square-wave/white-noise/pink-noise generator, an SPL meter and a setup page where you can calibrate your iOS device’s internal microphone, and its line-out and line-in levels. Naturally, as well as being a useful tool for setting up your Lyd loudspeakers, the app can be used in other setup applications and it’s well worth downloading, especially if you don’t already have something similar on your iPhone or iPad. Having set the geometry of your listening environment, the next step is to feed the pink noise from the headphone output of your iOS device (which should be positioned at the listening position) to each speaker in turn and to use the Sound Balance filter, if necessary, to tilt the...
RTA measurement and your own subjective methodology and use both the objective these resonances. If you follow this absorbers (aka bass traps) to dampen these, you'll need to employ low-frequency room's dimensions. To deal with any of multiples of, one or more of the listening with wavelengths equal to, or integer resonances that occur at low frequencies as peaks or dips in the RTA display, are some kind.

Having found that point, you can deal with it using a sound absorbing panel of with it. The idea is that if, from your listening position, you can see the loudspeaker in a mirror placed against any hard surface in your studio then you know you’ve found a first reflection point. Having found that point, you can deal with it using a sound absorbing panel of some kind.

Room modes, which will show up as peaks or dips in the RTA display, are resonances that occur at low frequencies with wavelengths equal to, or integer multiples of, one or more of the listening room's dimensions. To deal with any of these, you'll need to employ low-frequency absorbers (aka bass traps) to dampen these resonances. If you follow this methodology and use both the objective RTA measurement and your own subjective listening tests to identify and deal with any issues, you should end up with a monitoring environment that enables you to record and mix with the confidence of knowing that what you are hearing is what is going down.

What I Heard

Having run through the above procedure, with the Lyd 48s positioned some four feet away from any wall, I ended up with the Sound Balance switch set in the zero position. Since I usually monitor at an average of around 80dB, I was able to set the monitor's bass extension to its maximum and settle down to some serious listening. As always, I began with the EDM of Deadmau5 and COH, whose recordings test a loudspeaker's ability to reproduce extreme bass with power and precise timing, whilst at the same time delivering clarity and transient detail across the mid-range and treble. The performance of the Lyd 48 in the bass end was exemplary and I was very impressed by the level of definition and transient detail that it delivered in the mid and treble frequencies, even when the bottom end was shaking the metaphorical floorboards.

Two of my current favourite CDs are the 1969 original and 2015 re-recording of Deadmau5 and COH, whose recordings test a loudspeaker’s ability to reproduce extreme bass with power and precise timing, whilst at the same time delivering clarity and transient detail across the mid-range and treble. The performance of the Lyd 48 in the bass end was exemplary and I was very impressed by the level of definition and transient detail that it delivered in the mid and treble frequencies, even when the bottom end was shaking the metaphorical floorboards.

The compact size, performance and price of the Lyd 48 means that it offers serious competition not only to its nearest direct competitors, but also to similarly priced two-way active monitors that might be surprised to find themselves competing against a three-way system. If you’re looking for compact, high-performance active monitors at around their price range then a pair of Lyd 48s should very definitely be on your audition list.

**Conclusion**

The Lyd 48 is an impressively accurate and capable speaker that delivers a high level of audio performance at a relatively modest price for a three-way monitor of this quality. The superbly recorded musical performances whose dynamics, density of sound and harmonic complexities have proven to be a stringent test of a loudspeaker’s ability to place instruments in a soundfield, which the Lyd 48 succeeded in passing confidently and convincingly. I have not previously come across any other monitor with a single mid-range unit covering such a wide bandwidth, and the absence of any crossover-related anomalies in the mid-range gave the reproduction of voices and instruments in that area an impressive level of clarity and articulation that never sounded harsh, but that took no prisoners when it came down to revealing problems. The Lyd 48’s soft-dome tweeter, common across all the Lyd range, also deserves a mention, its smooth, detailed and inherently musical performance making the monitor untrysting to listen to and easy to work on for long periods.

With the Lyd, the feeling that I came away with was of a very self-controlled monitor that always sounded smooth and unflustered, no matter what I threw at it. Good productions shone and lesser productions had their shortcomings ruthlessly revealed. The Lyd 48 stereo soundstage was expansive, tangibly solid and gave a real sense of depth, with voices, instruments and other sources being reproduced with accuracy and placed precisely within it. Its reproduction of low-level detail was also impressive, with the hanging tails of the real world reverberations in Cantus’s CD Spez holding on into silence.

**£2637.60 per pair including VAT.**

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**DigiTech CabDryVR IR-based Amp/Cabinet Simulator**

Speaker emulators have evolved from simple analogue filter circuits to using convolution and impulse responses captured from real mic, speaker and cabinet setups. We’re used to seeing convolution technology in plug-ins, but recent years have seen this increasingly appearing in hardware form.

DigiTech’s take on this is a compact stompbox-format dual-channel cabinet emulator. It allows full dual-mono operation, processing two inputs separately. Alternatively, a single guitar amp can be processed using two different virtual speaker cabinets, or the pedal can process the guitar on one channel, while also outputting a dry signal on the other.

The CabDryVR input can only accept line/instrument signal levels, such as from your guitar, from your pedals, or from a preamp line output or effects send. It can’t receive the output directly from a power amplifier, though — you’d need to route the power amp output to the CabDryVR via a DI box that can accept speaker-level signals. It’s also worth noting that there’s no dummy load built in for use with valve amplifiers, the speaker or a separate dummy load needs to remain connected.

The CabDryVR is built into a sturdy cast metal case, and it comes with a StompLock rubber protector which slips over the jacks, and it comes with a StompLock rubber protector which slips over the jacks, and it comes with a StompLock rubber protector which

Lawing Zexcoil Z-Core

**Noise-cancelling Stratocaster Pickups**

Lawing Musical Products have been building their own take on noiseless replacement pickups for Stratocasters for some years now, assembling their Z-Series and Legacy Series pickups in their own custom shop based in Newark, US. Most noiseless Strat pickups use either a stacked coil or two narrow blade pickups side by side, which might tackle the noise but can be difficult to voice in a way that matches the single-coil pickups they replace. That’s not to say it’s impossible — Australian company Kinman make a range of stacked-coil noiseless replacement pickups which sound very authentic, but they use some pretty complex assembly methods.

Scott Lawing decided to go about solving the problem in a different way. Instead of stacking the coils, he used a separate coil for each string, each one surrounding a flattened pole-piece. The coils and magnet polarities are arranged in a three-plus-three, series-wired arrangement, to work as a humbucker. The six coil assemblies are angled and arranged side by side in a way that doesn’t cause level- or tonal-change problems when strings are bent into the zone between the G and D strings, providing the pickups aren’t set too far away from the strings.
For the hotter pickups in the range, there's the option to wire the two sets of coils in parallel, for a brighter output.

The reasoning behind this design approach is very deep and it's most definitely not a case of smoke and mirrors — Scott Lawing sent me a copy of his technical notes, which target the real reasons that pickups sound the way they do, with a great deal of importance being attached to the way eddy currents are controlled in the pole-pieces. Rather than use a bar magnet as a pole piece, as in a typical Fender pickup, each metal pole piece sits on top of its own magnet.

Recently Lawing Musical Products added the Zexcoil Z-Core range, which is offered at a lower price than their previous custom-shop ranges. This is made possible by manufacturing in the same US factory that winds all their coils, and they say more models will be added to this more affordable line when practical. We're told that the aim is to deliver a classic AlNiCo 5 tone, while offering a range of output levels. The pickups are offered as a set, either on their own or already wired, with the controls mounted on a cardboard plate, ready to transfer to your own pick-guard. (The quality of the pots, switch and capacitor is first-rate.) As the pickups are understandably quite expensive to manufacture, the company currently sell direct to keep the end-user cost down, but they will ship worldwide. Note that left-handed Z-Core pickups are currently only available from the Custom Shop, and orders may take up to three weeks to process.

The voicing of the pickups is clearly vital and, according to Scott, that has much to do with the electromagnetic properties of the pole piece around which the coil is wound. Once this has been established, different output versions can be created using the same pole-piece design by changing the number of turns of wire wrapped around it and by changing the strength of the magnet. Scott describes this as their Tone Tuning Technology, and he's made a lot of detailed measurements to ensure his pickups behave the way he wants them to in terms of their resonant frequency and Q.

The Z-Core range covers five voicing options, comprising Vintage Underwound, Vintage, Vintage Hot, Modern and Overwound. Cosmetically, the pickups don't emulate traditional Strat pickups as the pole pieces are hidden completely behind a cream plastic cover (which may offend purists but not me: they look really neat). They weigh a little more than standard pickups, and seem to be a few mm deeper, but they fit a standard body with no need for modification.

I was sent a pre-wired pickguard to review, and it took me only a few minutes to swap out my existing pickguard (fitted with a Fender Tex Mex pickup set) for the Z-Core Vintage set. As delivered, the Z-Core pickups were rather a long way from the strings but it was easy enough to raise them until the output level was close enough to what I was getting from my Fender pickups. Checking with the manufacturers confirmed that the Z-Core Vintage pickups work best when set fairly close to the strings (between 2.381 and 3.175 mm) and, unlike some single-coil pickups, you can get them very close without creating atonal overtones.

A quick tune-up and I was ready to check out the clean sounds, which I found to be bright with plenty of Fender-like snap to the note attack. (Of course, Tex Mex pickups are not vintage voiced, so I couldn't expect an exact tonal match.) My first impression was that the sound was in some way a little more hard-edged than I was used to, rather like the sound you get when switching from nickel-wound to pure steel-wound strings.

I discovered that this was due mainly to the slightly non-standard way the pick-guard assembly is wired. The bridge pickup has a dedicated tone knob, while the middle tone knob is wired to both neck and middle and has a no-load position when fully open, so that the output isn't loaded by two sets of tone controls in the bridge-plus-middle position. Backing it off by a small amount brings in the pot resistance and tames the highs. The pots are the standard 250kΩ used with conventional pickups, and there's no treble bleed wiring around the volume control.

There were no noticeable level or tonal changes when bending the G string high up on the neck and the pickup combinations sounded as they should with just the right degree of 'Strat-ish' hollowness. Patching in a high-gain overdrive pedal revealed just how effective these pickups are at rejecting hum. Even standing close to the amp there was no noticeable hum — just a bit of circuit noise from the pedal itself. Overall then the pickups do just what they should — they recreate an authentically bright, single-coil sound without the single-coil's inherent hum susceptibility. They're definitely up there with the best of the noiseless options; it's clear that a lot of thought and research has gone into the design.

While the cost of Lawing's custom-shop pickups might make you think twice about switching, these Z-Core pickups compete on price with single-coils — while delivering the same benefits as their custom-shop cousins. Tonally, they give you the bright, open sound you'd expect of a single-coil pickup, their hum rejection is impressive and there's a good menu of voicings from which to choose. All of which makes them a very practical proposition for recording, or when playing in venues where electrical interference manifests itself as a hum problem.

Paul White

£ $225 per set of three Z-Core pickups. Wired sets with controls and switch from $320.

W https://lawingmusicalproducts.com
Vox MV50 Series Guitar Amplifiers

The designers at Vox must have had one of those ‘just because we can’ moments when they dreamt up the MV50 Series of amplifiers — we’ve all seen ‘lunchbox’ amplifier heads, but if you had to put your lunch in, 60W of these you’d probably starve within a couple of days! Little larger than many stompboxes, these tiny amps are built into tough metal enclosures and run from an external PSU (which is, at least, rather more robust than most of the wall-warts and carpet carbuncles we come across). The PSU has a separate mains lead so doesn’t clog up your power strip, though it has one of those clover-leaf connectors rather than an IEC plug, so you’ll need to remember to take the right cable to a gig.

Although the outward appearance and underlying technology are the same, there are three different amps in the series, each with a different voice. But there’s no modelling going on: the preamp stage is all analogue, combining solid-state components with Korg’s new ‘flat’ 6P1 Nutube triode valve, which is fitted via a socketed connector to allow replacement if needed. A 50W Class-D power amplifier makes the small form factor possible.

There’s a helpful power percentage meter on the front panel, and while the number of controls may seem limited, there’s actually everything you need to get a great sound.

A rear-panel switch acts as a Standby power switch and there’s also a switchable Eco mode that puts the amp to sleep if left unused for about 15 minutes. There’s a speaker impedance selector on two of the models (4Ω, 8Ω or 16Ω), which means most speaker cabinets will be happy to be fed by them (the MV50 Clean model has a three-way power setting instead). There’s also an EQ switch to add a little low end when using the amps with smaller speaker cabinets. Because of the size, as well as the usual live and studio uses, it would be well worth considering one of these amps as a backup in case your main amp fails — though it’s possible you’ll find that you prefer the sound of the MV50 to your regular amp!

A headphone/line output with speaker emulation doesn’t mute the speaker feed — so you could use it with the speaker disconnected for silent recording, or could conceivably use it live to feed a mixing desk at the same time as a speaker, allowing the FOH engineer to apply further effects as needed.

The only real drawbacks are that these amps don’t include reverb and have no effects loop, and that might limit your processing options when gigging. In practical terms, if you want to add reverb or delay for live performance without doing so on the PA desk, you’ll need to run the amp clean and get your filth from pedals. In the studio, of course, you can add delay-based effects after recording.

I have to admit that my favourite of the three amps is the MV50 AC, which serves up chiming cleans imbued with that characteristic Voxy jangle, lovely touch-sensitive blues tones, and plenty of overdrive when the gain is turned up full. There are only two controls — Gain, Volume and Tone — but that Tone knob has a long reach, with the middle of the range sounding pretty ‘dialled in’ to me. You may need to add pedals if you’re after a Brian May tone, but with the gain up full it isn’t far off. Does it sound like an AC30? It’s hard to tell exactly — I’ve had a couple of AC30 Top Boost amps over the years, and by the time you wind them up to their sweet spot, the volume is punishing! But it is fair to say that the MV50 AC sounds the way I always wanted my old AC30 to sound when playing at sensible volumes.

The MV50 Clean is perhaps an amp for the pedal lover, as it stays true to its name no matter how far up you crank the volume control, only showing a hint of breakup when pushed really hard. There’s no separate Gain control, but you do get separate Bass and Treble controls plus that three-way power switch. Clearly, the sound is Fender-inspired, and as a pedal platform the amp does a great job; the two tone controls give you a little more tonal range to work with.

Lastly, we have the MV50 Rock, which follows the same control layout as the MV50 AC but is voiced to deliver more of a Marshall-type Brit rock tonality, with little respect for clean headroom. It also has more gain available than the MV50 AC and the sound only approaches cleanish on the very lowest gain settings. Crank up the gain and you’ll immediately get instant rock gratification with a generally fatter, more grinding tonality than the MV50 AC. Both the MV50 AC and MV50 Rock respond well to your guitar volume controls.

Though you can use these amps with any suitable extension speaker, Vox offer a ported eight-inch cabinet, the BC108, that’s specifically designed to be used with these amps. One of these didn’t come with the review amps, though, so I conducted my tests with a 1x12 Celestion Greenback-loaded, open-back cabinet, and found both the volume and the tone quality well up to the task of serious pub gigging or studio work. The analogue front end manages to make them both sound and, importantly, feel like a valve amp — and that Class-D power section can go mighty loud.

I like the idea of using the MV50 AC in the studio, where you can add effects such as delay, chorus or reverb in your DAW, and you also have the choice of DI’ing, miking, or both at once. That particular amp can do both clean and dirty sounds, making it particularly flexible. All the amps sound great at low volumes too, which is such an important consideration for the typical home studio. I’d gladly pay a bit more for a slightly larger box with a built-in reverb, but these little amps are just too cute to resist — in fact, I think I’ve just talked myself into buying the MV50 AC! Paul White

£199 each including VAT.
W www.voxamps.co.uk
Found On Reverb
Player Grade Vintage: 1970s Sony C-38B Large Diaphragm Condenser Microphone

The world’s largest marketplace for musicians, Reverb.com/UK.
SAMPLE LIBRARIES

Soniccouture
The Canterbury Suitcase
Kontakt Instrument

For many, the ultimate Rhodes electric piano was the Mark I Suitcase model. This is the hip, sophisticated keyboard sound we’ve heard gracing countless soul, R&B, jazz and pop tracks for nearly half a century, combining a clear, bell-like attack with a satisfyingly long sustain and a distinctive auto-panning stereo tremolo effect, ideal both for funky workouts and for sensitive, late-night ballad accompaniment. And that’s precisely the sound you’ll get if you buy Soniccouture’s Canterbury Suitcase, featuring 8GB (installed) of deep-sampled vintage Rhodes timbres. This iconic instrument resides in Toronto’s Canterbury Music Company studio, named by its owner Jeremy Darby after the Kentish cathedral city of his birth. Having conducted a sampling session at the studio in 2012, Soniccouture fell in love with its in-house 1976 Rhodes Suitcase 88 and returned four years later to sample the hell out of it: we’re talking 21 to 25 velocity levels per note with multiple random robin sounds simultaneously recorded through a direct line out, stereo speaker and room mics. Result: an exquisitely touch-sensitive virtual instrument comprising over 11,000 samples.

Without further ado, let me say that this piano is one of the best sampled Rhodes I’ve heard. I was immediately taken by its strong, pinging ‘bell’ attack, and pleased to find that both the attack and body tone remain consistent across the piano’s seven-octave range, with no dead notes, buzzes or volume dips — the mark of a well-maintained Rhodes. All that remained was to turn off the room mics, use the built-in EQ to enhance top end sparkle, and slap on some chorus and reverb: after that, I was lost to the world for a few hours, happily improvising celestial harmonies on this supremely playable and dynamic instrument.

The library requires Kontakt 5.6.8 and runs on the free Kontakt Player as well as the full version of Kontakt, with activation done online via the efficient, if somewhat intrusive NI Native Access system. There’s only one patch, but once it’s loaded you can access around 90 ‘snapshots’ divided into ‘Cleaner’, ‘Dirtier’ and ‘Experimental’ folders. Some of the latter presets (notably ‘Teutsuri’ and ‘Westminster Ghosts’) generate beautiful, haunting and ethereal overlapping overtones from a single key press, great inspirational timbres for sound-design excursions; the dirty distorted presets would benefit from some extra aggression, readily available from today’s manic guitar processor plug-ins.

In addition to an impressively authentic, head-spinning stereo tremolo, users can add up to six programmable effects, including chorus, phaser, flanger, delay, rotary speaker, filter/auto-wah, distortion, lo-fi and a good speaker cab emulation. There’s also a global reverb featuring a selection of plex halls, rooms, plates, springs and some vintage hardware emulations, including the mighty AMS RMX16 reverb. While these studio-quality effects are a welcome garnish, the main dish is the superior, unadulterated sound of this excellent Rhodes Suitcase piano.

Dave Stewart
£139
www.soniccouture.com

Sample Logic
Trailer Xpressions
Kontakt Instrument

If there has been an obvious trend within the sample library/virtual instrument world over recent years it has been the rise of the products aimed fairly and squarely at media composers. Sample Logic have been an active part of that trend and products such as Cinemorphx, Bohemian and Morphestra 2 offer some pretty sophisticated and powerful tools in that regard. Many of these products cross the border between music composition and sound design. However, Sample Logic’s latest release — Trailer Expressions — while appealing to the same potential audience, is perhaps more firmly situated on the sound design side of that line.

The library consists of a collection of WAV-based samples (just over 4GB in total) spanning atmospheres, drones, risers, stingers, scrapes, whooshes, impacts and other all-out sonic mayhem. You can work directly with the WAVs or, if you have the full version of Kontakt, via a dedicated Kontakt front-end that organises all the sounds into a number of themed Kontakt instruments and provides a number of additional ways to manipulate the sounds for maximum flexibility.

Each of the Kontakt instruments contains a number of related samples mapped across the keyboard, and patch names such as Hits, Risers, Scrapes, Pulses and Stingers give you a good guide as to what to expect within each of these collections. The front-end provides various global tools that are then applied to each sample within that patch. This includes the ability to adjust the playback start point in the waveform, playback pitch (variable over two octaves), attack and release controls, delay and reverb options, and a low-cut and high-cut EQ. Towards the top of the UI are the Energizer and Polisher options. These are both types of multi-effects, with the former offering a combination of compression, saturation and distortion while the latter combines EQ, saturation, transient enhancement and stereo width. Both are very effective to make the already ‘big’ sounds even bigger. All these on-screen controls are available for MIDI Learn and automation within your DAW or sequencer.

In terms of actual content, I’m not sure there is a huge amount that’s really new amongst the raw samples but they are all pretty impressive. Indeed, impressive enough that a little most certainly goes a long way; just two or three of these sounds layered in any sound design cue is going to pack a serious sonic punch. Where Trailer Xpressions does really score (d’oh!), however, is in just how easy it makes the task in hand. It might not be the cheapest sound design instrument you can buy, but it sounds great and makes it rather too easy to create a Hollywood-ready sound design cue in a matter of minutes. And, for working media composers up against yet another deadline, that could quite well be reason enough to meet the asking price.

John Walden
£199
www.samplelogic.com
Virz2

Cinematic Thunder: Epic Orchestral Toms v1.5
Kontakt Instrument

The sound of orchestral percussion as a backdrop while bad guys get chased, gigantic robots fight and time runs out has become something of a cinematic cliché. There is a reason for that — it works. There are a number of really good sample libraries that can supply the necessary level of sonic excitement, but a relatively new (and modestly priced) option on this front is Virz2’s Cinematic Thunder: Epic Orchestral Toms.

Well, I say new but, in fact, v1.5 represents a significant update on the original Cinematic Thunder released some four years ago. This revision comes with a substantial overhaul of the sample base, new samples added (the library now spans around 5GB), a brand-new Kontakt UI (the library requires the full version of Kontakt) and a pattern-sequencing option added. Of course, the heart of the library is still those multi-sampled (up to 14 velocity layers and three round robins) orchestral toms, with 18-, 16-, 12- and 10-inch drums sampled and new Surdo samples added for this version.

Four master patches — Sticks, Mallets, Group Hits and Group Big Hits — are included, each bringing a somewhat different sound. Within the Sticks and Mallets patches, the samples are mapped across the keyboard based upon drum size, with two keys for easy manual playing. However, you also get extra keys for two-, three- and five-stick hits and a ‘roll’ key that responds to the Mod Wheel to build crescendos. The two Group Hits patches are somewhat simpler and map various pairs of toms to single keys across the keyboard.

The main page of the new (and rather nifty) UI allows you to mix and match between three different microphone placements (from dry to ambient) or to use a ‘processed’ option (a pre-configured mix of the three main mics plus some added ambience). However, do be reassured that these sounds can get impressively big if you want them to and the multiple velocity layers make for a very expressive playing experience. The option to add a little pitch and velocity randomisation also helps here.

The Effect screen offers a convolution reverb, delay, limiter, EQ, distortion, stereo widener and compression. However, the Sustain Master is really worth exploring as it can dramatically alter the character of the hits to suit your needs. The final page — the new Sequencer — allows you to program a single sequence of up to 32 steps and with variable note divisions to sync to your project’s tempo. This is a doddlie to use and you can create some great pattern effects. It is, however, a bit of a shame that there aren’t additional options for pattern saving or for key-switching between multiple patterns.

That minor comment aside, as a source of very playable — and suitably thunderous — orchestral toms, Cinematic Thunder provides plenty of bang for a relatively modest buck. Well worth auditioning alongside some of the obvious competition. John Walden

www.timespace.com

Soundiron

Ambius Prime Kontakt Instrument

Soundiron’s original Ambius release received a glowing review from Martin Walker here in November 2010, since when the concept has gone from strength to strength. The entire series is now available as a single collection called Ambius Prime (9.15GB installed), which bundles Ambius 1 — Transmissions with Ambius 2 — Systematik and Ambius 3 — Expanse; the first two can be bought separately, but Ambius 3 is only available as part of the Ambius Prime bundle. The libraries run on both the free Kontakt Player and the full version of Kontakt (version 5.6.8 or later).

The libraries were created from organic field and live instrument recordings, which have been processed to create over 3000 original hybrid sounds. These have been whipped up into a huge array of pads, drones, ambiences, pulses, evolving textures and leads of a distinctly experimental nature. Not the kind of thing required by a keyboardist playing in a country rock band in deepest Virginia, but likely to be appreciated the subversive ear-shredding scream of ‘The Shakes’, like fingernails on a digital blackboard. By way of contrast, the library also features lovely, tranquil-sounding pads. Unleashing the arpeggiator on Ambius 2’s ‘Filter Slide’ preset yielded some great, bubbling percussive 1950s Radiophonic Workshop-style sequences, and I’ve made a mental note to dial up the library’s massive ‘Deeper Purple’ drone if they ever ask me to write the music for Blade Runner 3.

Moving on, I enjoyed the ethereal, slightly unsettling atmosphere of Ambius 3’s ‘Dominic 1’ sample. While many of that library’s presets are impressively complex, I found that disabling the X/Y pad motion and auditioning individual layers often revealed more generally usable material — for example, turning off the two cheesy Tron-style sequences built into the ‘Psychedelic Anxiety’ preset simplified and improved the overall sound. This points to the fact that while Ambius Prime has lot to offer at surface level, further rewards await those who take the time to investigate its deeper features.

Dave Stewart

$149

www.soundiron.com

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www.soundiron.com
We examine the production of some recent hits to help you brush up on your listening skills.

**SAM SMITH**

*’TOO GOOD AT GOODBYES’*

When shifting into their falsetto register, some male vocalists find it more comfortable physiologically to restrict (or even close off) the flow of air through their nasal passages. A side-effect is that ‘r’ and ‘m’ consonants begin to approach ‘d’ and ‘b’ instead, almost as if the performer has a cold.

Normally this isn’t a huge issue, because most singers only use falsetto pretty sparingly, but it’s very much Sam Smith’s stock in trade, and I honestly found the nasally congested lyrics in this song a bit distracting. In the chorus, for instance, we’re treated to “you hurt be” and “every tibe” twice, and (my personal favourite) “we don’t stad a chas”.

And before anyone tries to argue that a blocked nose simply adds tear-stained authenticity to his emotional message, bear in mind that there are plenty of other moments (most of them not falsetto) where he displays all the clarity of a menthol-lozenge endorsee.

So what to do? I’ve had a certain degree of success remedying exactly this kind of thing with extreme automation moves, boosting the underpowered consonants and the little vowel-consonant and consonant-vowel transitions either side of them by 6dB or more. However, I also know from experience that sometimes there’s simply not enough information to work with. So really the best cure is prevention: either try to get the singer to work on opening up their nasal passages more during falsetto singing (which, incidentally, tends to give the sound a bit more airiness and intelligibility in general), or just avoid writing those consonant sounds into falsetto lines.

I also can’t help feeling that the lead vocal’s tuning seems to be pulling a little bit flat, at least in relation to the piano and bass. This is something that I can’t imagine having been left to chance with such a high-profile artist, and the cynic in me wonders whether it might be the result of an inappropriate pitch reference in Melodyne. But if it’s intentional I can see a definite justification, as I reckon it helps make the song feel a little more plaintive.

Mike Senior

**TAYLOR SWIFT**

*’LOOK WHAT YOU MADE ME DO’*

One of the things I immediately liked about this production is that it inverts the long-term dynamics archetype that most pop songs are built around: namely, that the verses start relatively stripped back in arrangement terms, and then ramp up into a relatively full-sounding chorus, the idea being that the chorus payoff aligns with the arrangement payoff. Here, though, the chorus is about as minimal as you can get, with just the spoken vocal hook and a spartan drum beat. Its impact is built not on textural thickness, but shock value and ‘upfrontness’. But what makes this particular subversion special is that the arrangement gives all the signs of being more stereotypical, with a pared-down verse leading to a pre-chorus build that still seems to be leaving plenty of room for a big chorus payoff; when the chorus turns out to be so sparse, the emptiness is all the more startling.

Comparing the different choruses reveals another thought-provoking detail: choruses one and three have a very dry-sounding lead vocal, whereas choruses two and four give Taylor quite a strong, short room ambience. As the third and fourth choruses constitute the song’s final double-chorus, this makes a lot of sense in terms of opening out the sound for the finale, and likewise the idea of a second chorus offering a more expansive sound than the first is no big news. It’s just that most of the time producers pull off those kinds of section contrasts with added instrument or vocal layers, rather than with such a characteristic lead-vocal ambience, and I can’t recall having heard the clear odd/even connection as noticeably in any other recent production.

There’s a nice lyrical trick in the choruses too, adding the word “just” to the third and fourth iterations of “look what you made me do” to syncopate the
speech stress patterns — a great example of getting more repeats out of a hook without tedious, simply by subtly shifting the context. Reminds me of a previous hit of hers, in fact: ‘We Are Never Ever Getting Back Together’ pulls a similar stunt with its extra “ever”. Mike Senior

IMAGINE DRAGONS

‘BELIEVER’

If you want to wrong-foot the listener to highlight the onset of a song’s chorus, but your main hook/melody happens to start only on the chorus’s first backbeat, then one tried-and-trusted arrangement method is to create a kind of arrangement ‘suckout’ on the downbeat, allowing the full texture to explode into existence on beat two. This is most often used as a one-off device, perhaps to add that extra bit of spice to the final choruses after a middle-section build-up, but this Imagine Dragons number rides it till the wheels fall off!

The first specimen appears at the head of chorus one, the downbeat stripped down to its solo bass synth (a gloriously grubby-sounding one!) before reinflating into the backbeat’s mob-vocal “Pain!” But it substitutes a full-texture variant with distorted guitar two bars later, when I’d half expected it to repeat verbatim, as well as suddenly reappearing following new thematic material at 1:09. So the resulting A-A’-B-A structure effectively extracts three attention-grabbing moments from a single ‘sit up and take notice’ arrangement stunt.

It gets better. Since the second chorus is cast from the same mould, it creates a sense of inevitability about the approach to chorus three and the producers capitalise on that expectation to refresh both the suckout moments. The first replaces the bass drop with an extended taiko swell first heard during the build-up to chorus two (1:53), transplanting that from its original beat-three location to the downbeat, slily generating a flavour of bar-extension/contraction without actually deviating from the meter. But the second instance trumps them all by muting everything but the vocal reverb tail.

What’s also unusual about this production is the extensive use of transition effects to connect and add momentum between the basic groove’s ponderous main beats. Pitch-bend transitions are many and varied, for example: there’s not just that signature bass drop and its rising-pitch guitar substitution, but also the shift between downwards and upwards pitch inflections on “Pain!” during the different chorus sub-sections. There’s plenty of ‘velocity ramping’ too, most obviously in the Taiko-style fills that are baked into the basic groove, and which (as I already mentioned) so effectively flag the second and final choruses from different metric viewpoints. Extended reverse-envelope effects can be heard on the final backbeat before choruses one and two, while a shorter one returns us to the full texture following that final dramatic suckout at 3:12. Furthermore, these different transition effects are frequently chained for added impact. Check out the quintuple-whammy going into the second chorus from 1:53: two different taiko velocity ramps lead into both the third and four beats of the bar preceding the chorus; a reverse-envelope transition effect propels us into the downbeat; and then the pitch-drops of the bass-drop suckout and its following “Pain!” carry us right on to beat three. Whatever the band paid their production team, it wasn’t enough! Mike Senior

CLASSIC MIX

SPICE GIRLS

‘WANNABE’ (1996)

One of the great intangibles of mixing is how to develop the most successful ‘vision’ for your mix, especially when you’re working with new artists who haven’t yet nailed down a definitive musical identity. In this regard, what I find so interesting about the Spice Girls’ debut single is that it provides an opportunity to compare the visions of two of the world’s greatest mix engineers, by virtue of its two release versions. The first was created by Dave Way, who has mixed chart smashes like Christina Aguilera’s ‘Genie In A Bottle’ and Macy Gray’s ‘Try’. The band ending up rejecting it, and the official radio version that most people know was mixed by Mark ‘Spike’ Stent.

What I find most striking about Way’s mix is that the backing track is tons cooler than Stent’s, full of that gritty, urban R&B attitude that was tearing up the airwaves in the mid-’90s. It’s got low end for miles and the drums sound massive. Stent’s version, on the other hand, boasts a backing track that feels downwards and upwards pitch inflections

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by the vocals in the balance. But while Way’s sonics are much more to my own taste, I think the band chose the better version as the official single — Stent’s vision just makes a whole lot more sense in relation to the identity the band were trying to promote.

The section that demonstrates this most clearly is the rap (1:45-1:54), where Scary’s sing-song “Pain!” during the different chorus sub-sections. There’s plenty of ‘velocity ramping’ too, most obviously in the Taiko-style fills that are baked into the basic groove, and which (as I already mentioned) so effectively flag the second and final choruses from different metric viewpoints. Extended reverse-envelope effects can be heard on the final backbeat before choruses one and two, while a shorter one returns us to the full texture following that final dramatic suckout at 3:12. Furthermore, these different transition effects are frequently chained for added impact. Check out the quintuple-whammy going into the second chorus from 1:53: two different taiko velocity ramps lead into both the third and four beats of the bar preceding the chorus; a reverse-envelope transition effect propels us into the downbeat; and then the pitch-drops of the bass-drop suckout and its following “Pain!” carry us right on to beat three. Whatever the band paid their production team, it wasn’t enough! Mike Senior

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Notes From
The Deadline
TV Music From The Inside
Is it really still OK for our livelihoods to depend on a small piece of plastic?

PAUL FARRER

I n May 2004, I happened to find myself in New York the same weekend that Shrek 2 opened. Having grown up in rural England, where our Internet is in black and white and we still only have one type of coffee, I remember the buzz of excitement that went with seeing a movie in America on its opening night. Walking back to my hotel afterwards, I had to step over a man sitting on a blanket on the sidewalk selling counterfeit DVDs, handbags and sunglasses (I’ve never understood this most familiar, and yet peculiar combination of bootleg consumer goods). Pausing to have a look at the quality of his wares, I spotted he was selling Shrek 2. When I expressed my surprise at seeing a knock-off DVD of a movie that had only opened that same day, he swiftly produced a portable DVD player and screen, popped the disc in, and played it to me. It even had a director’s commentary on it. Not by the real film-makers, of course; it was a couple of Russian blokes chatting in broken English over the wobbly back-row-of-the-cinema footage, pretending they were involved in the production and swapping clearly made-up stories of what a cool guy Mike Myers was to work with.

The Keys To The Universe
I’m not sure about you, but one of the first things I’d grab if my house was on fire would be my iLok, my Mac, my cat and (time permitting) the majority of the members of my immediate family. In my world my studio computers never really die, they just get moved upstairs and replaced, meaning that at any one time I have as many as five different studio Macs I could use to hit the deadline and get the job done. But thanks to my friend on his blanket two blocks from Times Square and his Roy Bon sunglasses, I’d be seriously inconvenienced if I lost my Syncrosoft key, and I’d be royally screwed if I lost my iLok.

Copy protection is one of those odd things that massively inconveniences the genuine customer and simultaneously doesn’t bother the pirate in the least. And yet the reality is we live a world where irreplaceable, tiny, fragile, plastic USB sticks are the gatekeepers to your being employed or not. It’s a silly situation, no doubt — but there is, at least, a bit of light at the end of the tunnel. And it comes in the shape of a telephone that allows you to animate and match your facial expressions onto the emoji of a cartoon poo in real time. Think different indeed.

Face Off
The release of the iPhone X marks the world debut of industrial-quality 3D facial recognition, and whilst I couldn’t care less about using it to authorise the purchase of a flat white at Starbucks (whatever that is), I desperately want it to migrate to Apple’s desktop computers and be rapturously adopted by third-party audio plug-in developers as a matter of extreme urgency. It took more than five years for Siri to migrate from the headline-grabbing billion-dollar prestige of the iPhone 4s to the smelly disabled parking space of Apple’s OS 10.12. But I’m hoping Face ID’s journey will be faster and grabbed with both hands by anyone who makes software. I want my Mac to know that it really is me trying to make a living slumped in front of it, and I’d like all the bags-under-the-eyes-forgiving, stubble-ignoring facial recognition software to tell all my plug-ins that it’s really OK to let this pale, shattered, frustrated lump hear those viola samples he paid good money for without redress to an unspeakably precious, awkward piece of plastic sticking out the back of a USB hole somewhere.

I’ve spent more than enough hours of my life gawping at my computer. It seems only fair that it recognises my presence in the relationship.
The Funkhaus (‘broadcast house’) in the district of Oberschöneweide in the former east of Berlin is no longer a well-kept secret. Located on the banks of the river Spree, away from the city centre and nestled between an old power station and a cement plant, it’s one of the finest examples of recording studio architecture to be found anywhere in the world, and quite possibly the largest structure on the planet to have been purpose built for music production.

The reason why this sonic miracle exists is intimately connected to Berlin’s unique history. When Germany was divided after World War II, what was left of its former nationwide broadcasting services had to be split as well. For a brief period in post-war Berlin, broadcast networks for the Western and Eastern sectors were operating from the same location in West Berlin, but as political tensions rose, this became untenable.

Eventually the nationwide East German broadcast networks found a new headquarters in East Berlin in the first half of the 1950s. The Funkhaus was Block B of a much larger architectural ensemble created by Bauhaus architect Franz Ehrlich, and became the recording centrepiece of this ‘city within a city’, which once was home to 5000 employees. Today rated a protected architectural site, Block B
The remote preamp rack, a vital contributor to Nils Frahm’s piano sound, comprises an RCA OP-6, several WSW preamps and a pair of converted Ampex 601 preamps from tape recorders.

Several types of brushes are on hand for unusual piano sounds.
is home to an 800-square-metre recording hall as well as almost half a dozen other independent recording studios conceived for various purposes from chamber music to Foley recordings. Remarkably, all of these studios share the same outer shell, but within this structure they were designed like individual buildings, all erected on their own separate foundations.

**Troubled Times**

Shortly after German reunification the broadcast business at Funkhaus was phased out, and the whole site became subject to real estate speculations: The Funkhaus complex has changed hands numerous times since the early 1990s, sometimes under criminal circumstances. Music has always been made on the premises, but although a few remarkable recording projects have taken place at Funkhaus over the past 15 years, the incredible building hasn’t consistently been used to its full potential. It is not easy to build or operate large-scale recording venues in the current climate, and the internationally recognised strengths of Berlin’s music scene largely encompass underground genres which don’t require large halls. But things are developing, and for the first time in its modern history, the Funkhaus now houses a truly world-class studio where the technical setup is on par with the architectural and acoustic properties of the place.

In 2016, acclaimed pianist, composer and producer Nils Frahm took over operation of the former chamber music hall at Funkhaus, and after extensive remodelling, the studio is now ready to embark on the next phase of its journey. “We spent a lot of money to make it look like we didn’t do anything at all!” explains Frahm. Some acoustic properties of the control room were modernised, the studio was fitted with new mains wiring and signal cabling, and the exquisite woodwork received a makeover as well.

Nils Frahm insists that he is simply “hosting” the venue, which will still go by its original unpoetic name, Saal 3 (‘Hall 3’). As he puts it, “You can’t name something that already has a name!” The studio comprises the large 140-square-metre live area, as well as a smaller multi-purpose room of 25 square metres. The centrepiece of the 32-square-metre control room is a custom-built console based on Danner-format modules such as Neumann faders and Lawo EQs. Basically a 28-channel inline configuration, the desk sums on eight busses, offers eight aux sends and 16 microphone preamps, as well as 16 additional inputs for a sidecar made of Siemens modules like the W295a/b EQs. Together with his two main technicians, Nils Frahm designed a digitally controlled switching matrix based on high-quality relays integrated directly into the desk — a notable upgrade over the commonly used external switching solutions.
When a globally recognised brand asks for a piece of music that playfully evokes the 1960s spy thriller genre for their latest advertising campaign, where do you start? Brooklyn-based music house Hyperballad brought in a drummer, horn player and bassist and started tracking. We find out how the track took shape, explore the musical demands of delivering multiple versions for a range of media, and hear how Hyperballad communicated with their client throughout the process.

www.youtube.com/soundonsoundvideo

This month — Hyperballad: Music for Ads

When a globally recognised brand asks for a piece of music that playfully evokes the 1960s spy thriller genre for their latest advertising campaign, where do you start? Brooklyn-based music house Hyperballad brought in a drummer, horn player and bassist and started tracking. We find out how the track took shape, explore the musical demands of delivering multiple versions for a range of media, and hear how Hyperballad communicated with their client throughout the process.

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The Saal 3 control room is now based around a unique custom analogue console featuring a digitally controlled switching matrix.

Outboard EQs include vintage Pultec and Lang units.
patchbays mostly fed by longer, unbalanced leads. Based exclusively on Class-A, transformer-balanced modules and employing short signal paths, this one of a kind console meets mastering-grade specifications.

**Top Class**

The outboard collection reveals Frahm’s taste for certain signal-processing topologies. There are several Gates valve limiters such as a pair of SA-39s, a Level Devil and a Sta-Level, along with a Department of Commerce limiting amplifier and four 1176 clones made by his own technicians. Outboard EQs include a Pultec EQP-1 as well as four Lang PEQ-2s, and there also is a number of vintage effects units like two pairs of Ursa Major Space Stations and Roland RE-501 Chorus Echoes, a Binson Echorec 2, a Quantec QRS and Dynacord VRS 23 and EMT 245 reverb units. Saal 3 also offers EMT 140 and 240 units, and Nils Frahm has also revived one of the three original Funkhaus echo chambers.

Much of the amazing kit on offer at Saal 3 can be found in the live areas, including a remote preamp rack with an RCA OP6 and some WSW and converted Ampex 601 preamps, and an instrument collection which should satisfy any keyboarder’s wildest dreams. There is a Yamaha CFX grand piano and several upright pianos, as well as a Wurlitzer 200a and more special instruments like a Mannborg Harmonium, Deagans Celeste and numerous analogue synthesizers including a Rudi Linhard-modified Memorymoog, two Roland Juno 60s, a Moog Taurus, a Formant modular and many more. The microphone collection is built around a Decca Tree based on three Neumann M50s, an RFT bottle mic with M7 capsule, numerous Neumann KM54, KM56 and KM84s, an RCA BX-44 and 77-DX as well as Coles 4038 ribbons, and rare gems such as an Altec/Western Electric 639A.

The first project to be recorded at the new Saal 3 was Frahm’s upcoming album *All Melody*, which will be released in January 2018. But the studio is now ready for client bookings, too. In Frahm’s words: “The Funkhaus offers all the ingenuity of a classic Prada shoe: it’s durable, comfortable and it matches well with everything else. The room, the whole building, all details convey this idea that you are the artist, this is your place, readily awaiting whatever you’d wish to do. Nothing will stop us out here, in the nature, next to the water, in this friendly place which at times feels like an old holiday home.”

A holiday home capable of crafting recordings to the highest of standards, that is...
The ability to transpose parts after they've been recorded can be useful for a number of reasons, and Cubase offers a number of tools that make this possible, for both MIDI and audio. In the article that follows, I'll run through a few examples to help you get started.

All Change

Tucked away under the Add Track menu option alongside Cubase's other track types is the Transpose Track. As a broad-brush tool for transposing your entire project or just a specific section of it, the Transpose Track is easy to use and very useful. Once added to your project, the Pencil tool lets you add transpose events (they look a bit like an empty MIDI clip). These extend from the insertion point to the end of your project or, if there is one, the next transpose event. Located bottom-left of each event is the transpose setting; hover the standard cursor over this value, and then click and drag up/down, and you can set the transpose value up/down in semi-tone steps.

An obvious possibility — something of a pop cliché — would be our old friend the key change. To add the top line. A single transpose event at the start of the Transpose track can be a great help in finding the ideal key for the singer's range. If this requires more than a few semi-tone steps in either direction, some re-recording of the backing tracks (particularly audio tracks) is advisable, but at least you have an easy way to find your singer's comfort zone.

Another common application for the Transpose track is more corrective than creative; finding the right key for your singer. There are lots of producers who create musical beds and then hire in a singer to add the top line. A single transpose event at the start of the Transpose track can be a great help in finding the ideal key for the singer's range. If this requires more than a few semi-tone steps in either direction, some re-recording of the backing tracks (particularly audio tracks) is advisable, but at least you have an easy way to find your singer's comfort zone.

Another practical role for this technology is helping a vocalist hit the high notes: if the key is just a step or two too high for the singer to get the top-most notes, you can drop the whole arrangement down by the step(s) required while tracking the vocals, before simply returning the project to the original key: the vocals will be pitched up accordingly.

Let's Dance

Not all of us are blessed with great keyboard skills, and while Cubase offers all sorts of ways to assist you in creating MIDI-based chord sequences (for an example, see SOS May 2015: http://sosm.ag/cubase-0515), the transpose features can be a handy alternative — I’ll use the creation of an EDM-style chord sequence to illustrate this. The screenshot shows an eight-bar sequence ‘starting point’, where the rhythmic feel might be OK, but eight bars of Cmaj7 is perhaps not the most interesting dance chord sequence ever created!
pointed out to you, and you can let your ears judge whether the occasional ‘out of scale’ note is musically acceptable or requires further tweaking.

Note that while this ‘dance chords’ example has focused on MIDI tracks, the same Chord Track colour-coding is available when editing monophonic audio tracks in VariAudio. If you want to experiment with transposing/re-writing your vocal melodies, for example, Cubase can give you some useful guidance.

Any Other Business

Once you’ve used these transpose tricks to create a killer dance chord vamp, you may be interested to know what the actual chords being played are, and perhaps want other MIDI tracks to follow those chords. Simply right-click the MIDI clip containing your new chord sequence and choose Chord Track/Create Chord Symbols. Based on the settings you choose, this dialogue will place the relevant chord labels on the Chord Track.

The Transpose Dialogue can help you find new chords, keeping notes in the appropriate scale. If transposing notes by hand, Chord Track-based colour-coding can help you avoid errors.

Colour Me Bad

Of course, you don’t have to use the Transpose Setup approach. Having set the Chord Track key-scale and the Chord Track colour-coding in the piano-roll editor, you can move notes around in the piano-roll editor manually. As you shift all the notes of your chords up/down, if any red notes appear you can simply select these and experiment with shifting just those notes an additional step or two to see what new ‘chord creations’ manifest themselves. Yes, it’s still a case of trial and error, but at least some of the errors are being scale-sensitive assistance. For example, the settings shown in the screenshot will transpose the selected notes by a semitone, but because Scale Correction is selected and both the current and new scale are set to the project’s scale (C-major in this example), any note movements created by the dialogue will remain ‘in scale’ (so some may move by more than one semitone). To ensure that transposition doesn’t take any notes too far from their original register, the Use Range parameter can constrain the range of allowable notes.

The colour-coding of notes on the piano roll doesn’t have to relate to velocity — it can also display key/scale/chord information relating to the Chord Track.
Impact is one of the virtual instruments that has been bundled in Studio One since its earliest days. It might look like a standard drum machine, but Impact has some cute tricks up its sleeve (drum machines have sleeves, right?), a number of which are not documented very well (or at all). This month, I’ll share some of them.

Get Padded Up

A drum machine is a sample player optimised for playing percussion sounds and making beats. PreSonus bundle quite a few kits with Impact, and more are available from third-party developers like MVP Loops and The Loop Loft, but making your own kits is as easy as dragging individual sounds from Studio One’s browser and dropping them on Impact pads.

A sample can be trimmed to remove silence at the beginning and/or end using the Offset Start and End settings below the waveform display. Getting these trims right often necessitates zooming in and out on the data, and Impact hides the tools for this in plain sight. A long and a short bar run immediately below the waveform display, the shorter bar on the right sets the size of a window on the data, while the longer bar on the left moves that window through the full length of the sample.

Each Impact pad can trigger up to four different samples, which opens many possibilities. The most common and obvious use for this is velocity switching. Perhaps you want samples of a snare drum being hit very quietly (pp), fairly quietly (p), kind of loud (f), and really loud (ff). Adding samples to a pad is as easy as holding down the Shift key while you drag and drop the sample onto the pad. Each sample is added to the high end of the velocity range, so plan how you want the samples laid out in the velocity range and drag them starting from the quietest sample and working up to the loudest. A sample can be removed by clicking it in the velocity bar and then clicking the ‘minus’ button to the right of the Prev button below the waveform.

By default, the full range of velocity values is evenly divided between the samples dropped on the pad. The range assigned to each pad is visible in the bar above the waveform display. Clicking in the velocity range bar plays the sample assigned to the range you click. However, equally sized velocity ranges for each sample are rarely desirable in practice, at least, when any amount of realism is the goal. Generally, it is better for relatively few hits to sound the very loudest and softest samples, while most hits trigger the middle samples. The best range sizes will vary depending on the sounds, controller, musical genre and your playing style. Thus, you will usually want to customise the velocity range assigned to each sample. This is easily done by dragging the range boundaries in the velocity range bar.

The Long View

Although Impact is intended for playing drum and percussion sounds, you can load its sample editing includes a zoomable waveform display, velocity switching, sample trimming, and auditioning of a sample by selecting it in the velocity range bar.

The key to using long samples in Impact is the Decay slider, which must be all the way up to play a very long sample.
processor power, which opens the door to using multiple instances. One instance might be dedicated to snare drum: left and right hand strokes, rim shot, side-stick, brushes, rolls... the works. A second instance might be for cymbals. You probably won’t need a dedicated instance for kick drums, but, hey, I once saw the great Billy Cobham playing with 10 kick drums.

People often make beats by adding parts while a sequence loops. The Track / Transform / Transform to Audio Track command lets you capture what you have recorded this way. I sometimes do that to get an audio rough I can play with. When you are ready to perform the final mix, choose the Event / Explode to Pitches to Tracks command for easy processing and mixing of each element separately. Doing so also makes it easy to archive parts by transforming each track to audio when the project is done.

velocity modulations, they can produce some great sounds. For one project, I dragged a left/right pair of snare samples onto two pads, applied a band-pass filter with cutoff modulation to the left-hand stroke sample only, and panned them slightly left/right, just outside of the centre: a great sound, quickly arrived at. The high-pass filter, likewise, can produce great variations on a sample. Creating variation with filters is very useful when you have only a single sample of a sound.

The Stretch Factor field in the bottom right corner of Impact is another fun resource. It is actually a playback rate control, a speed multiplier. You can set this parameter within a range from 10 (plays 10 times normal speed) down to 0.1 (10 times slower than normal speed). Try applying a setting of three or four to a cymbal sample for a complex and interesting sound. Interestingly, Stretch Factor appears to be the only parameter besides sample start and end that can be set on a per-sample basis, rather than on a per-pad basis.

Impact demands surprisingly little any sample you want onto a pad. Loading longer samples makes Impact function as a triggered clip playback system. I haven’t had the patience to find out if there’s an upper limit on the length of the samples that can be loaded, but if you’re going down this road, there are a few things you should know about the block of controls in the lower right of the Impact window.

The Play Mode field has a drop-down menu with four choices. One Shot Poly plays the entire sample, with additional triggers starting additional copies playing; One Shot Mono plays the entire sample, with additional triggers cutting off and restarting playback. In Toggle mode, the first Note On message starts playback and the next stops it, while in Note On/Off mode, the Note On message starts playback and a Note Off message stops it. What all of these modes have in common is that they are supposed to allow playback of the entire sample — but the length of playback is actually determined by the AHD envelope controls in the amp section just above.

Given that, you might think that to play back a two-minute long sample you need only slide the Hold slider all the way up, where it displays an infinite hold time value. In fact, however, that only enables playback of about 10 seconds. Darn! Here’s the secret: slide the Decay slider all the way to the top. As I said, I can’t say just how long it will play, but the 20-minute sample I started in Toggle mode as I was writing this has been toodling along for well over 10 minutes so far.

Change It Up

Impact does not have all of the processing resources of Studio One’s Presence virtual instrument, let alone the editing and processing of a full-featured sampler like MOTU MachFive 3 or Native Instruments’ Kontakt. But it does have a few nice sample-shaping features. One such is the multimode filter on every pad, offering three different variations each of low-pass, high-pass and band-pass responses.

I like the sound of these filters on drum samples, and combined with the hard-wired envelope and processor power, which opens the door to using multiple instances. One instance might be dedicated to snare drum: left and right hand strokes, rim shot, side-stick, brushes, rolls... the works. A second instance might be for cymbals. You probably won’t need a dedicated instance for kick drums, but, hey, I once saw the great Billy Cobham playing with 10 kick drums.

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An Impact kit entirely devoted to cymbals and cymbal effects: ride ping and bell, crashes, chokes, stick slides, rolls and so on. Most pads have more than one sample. To the processor, Impact does not have a heavy impact, so several instances, such as this cymbal kit, can easily be used in a song document.
Surgical Solutions

We help you get to grips with Reaper’s new spectral-editing functions.

One of Reaper’s most recently added facilities is spectral editing, which can now be performed in the main project window. In this article, I’ll help you discover what this powerful tool can do. It won’t (yet) entirely replace the more advanced third-party spectral-editing tools such as iZotope RX or Magix SpectraLayers, but given time I’m sure the tool will evolve. In the meantime, it’s possible to achieve a great deal already...

Before you can start editing, you need to understand what the new Spectrogram view is telling you. We’re all familiar with the usual Peaks waveform, which plots level (‘Y’ axis) against time (‘X’ axis). It doesn’t tell us anything about frequency, though — for that most people fire up a frequency analyser plug-in. The Spectrogram view displays frequencies on the ‘Y’ axis and time on the ‘X’ axis, but with the level at different frequencies indicated by colour. With experience, it’s really easy to spot distinct sounds in the Spectrogram — the squeak of fingers on a fretboard, for example, or manuscript page turns.

Peaks Practice

There are two ways to access this new view: globally, for all tracks, by going to the Peaks Display Settings window (via the View menu); or for individual tracks right-clicking on a clip and selecting ‘Spectral edits/Show spectrogram’. Note that you can ‘protect’ a track from the global view settings by applying the action ‘Track: Prevent spectral peaks/spectrogram’. (We’ll discuss the Spectral Peaks view another time.)

Whichever approach you choose, you’ll want to use the Peaks Display Settings to customise the Spectrogram view. This window features a drop-down menu where you can choose the desired display: Peaks, Spectral Peaks, Spectrogram (and some combinations). The Display Gain is used to increase the size of the waveform — it’s useful when analysing quiet sources, but note that this global setting applies to the Peaks view as well as the Spectrogram, so it’s worth assigning a shortcut key to access this window. Often useful in Spectrogram view is the ‘freq log’ parameter, and increasing this from the default value of zero has the effect of zooming in on the lower parts of the frequency spectrum.

At the bottom of the window, you can drag the coloured bar to change the colours used on the Spectrogram display. This sets the broad colour scheme, and the ‘curve’ parameter governs the intensity of that scheme — reducing the default setting (maximum) makes louder and quieter frequencies appear less differentiated. Typically, I leave this at the default, but I’ve sometimes found it useful to juggle the settings of this and those of the contrast and brightness controls — while the resulting appearance is perhaps technically less precise, it can help to reveal certain sounds more clearly at lower zoom settings. To understand what I mean, try reducing the curve parameter and brightness considerably but increasing the contrast — notice both the change of the Spectrogram display itself and in the colour scale at the bottom of the Spectrogram Settings window. It’s perhaps not the most intuitive system, but it’s possible to fine-tune the colour scheme considerably.

Note that the Spectrogram displays the frequency/level information post any clip processing (such as reverse and pitch changes), but pre any effects processing (whether they be Clip FX or channel inserts).

Fret Not

OK, enough about appearances — let’s work through an example to help you get your bearings. The screens show a short acoustic guitar recording (from the excellent resource freesound.org — it has a Creative Commons license, and you can grab it here if you want to follow the example: https://freesound.org/s/389401). We’ll use Reaper’s spectral editing to make the finger/fretboard scrapes intrude a little less on the music. The first step, of course, is to play the part back and listen for problems. Locate an offending scrape and zoom in on the Spectrogram so you can see what’s going on.

As you can see from the screenshots, the onset of each note is visible as a vertical line, but there are also some areas where you can see a few vertical lines, with a bit more mid/high-frequency information following them — these are the finger scrapes, which you’ll soon learn to recognise.

Zoom in fairly close on one of those, and make a selection (I have Reaper set up to make a selection by right-click-dragging, and quieter frequencies appear less differentiated. Typically, I leave this at the default, but I’ve sometimes found it useful to juggle the settings of this and those of the contrast and brightness controls — while the resulting appearance is perhaps technically less precise, it can help to reveal certain sounds more clearly at lower zoom settings. To understand what I mean, try reducing the curve parameter and brightness considerably but increasing the contrast — notice both the change of the Spectrogram display itself and in the colour scale at the bottom of the Spectrogram Settings window. It’s perhaps not the most intuitive system, but it’s possible to fine-tune the colour scheme considerably.

Notice the vertical lines that show individual notes, and the ‘smearing’ that is a finger-scrape or fretboard squeak.
which makes this really easy. Right-click on your selection and from the 'Spectral edits' menu, select 'Add spectral edit to item'. A translucent box will appear, accompanied by a number of small knobs. The box is your 'Spectral edit'. It determines which frequencies and time selection will be affected by your spectral editing. The knobs govern the actions that can be performed. Despite these being somewhat limited in scope (you can’t apply effects, for example, or cut certain frequencies to paste them on another track) you can still achieve quite a lot.

First, we're going to simply turn down the offending frequencies. By default, the box occupies the full height of the Spectrogram so that all frequencies are affected. To adjust this, drag up from the bottom of the box (it’s possible to drag down the top but it’s all too easy to grab other parameters such as the clip volume!). Then grab the line across the middle of the box to move it. With the top and bottom lines of your selection box now visible, it’s easy to refine the selection. One frustration at the time of writing (Reaper 5.52; hopefully it will have been addressed by the time you read this) is that there seems to be a glitch in the ‘X’ axis selection. You need Snap disabled, of course, but even then I’ve found that the box can suddenly grow when trying to make horizontal adjustments to the selection, and it can’t be shrunk again. This seems to be related to the ‘Take processor FFT size’ setting, because to shrink it, my workaround is to change that setting, reduce the box size as desired, and then change the setting back.

Once you’ve made your selection, use the uppermost knob (‘Spectral region gain’) to attenuate the scrape sound. If it helps, go further than sounds good so that it’s more obvious that you’re removing the right frequencies. Then back off the attenuation until it sounds in the right ballpark. The boxes aren’t the most precise means of selection, and you’ll need to finesse the result using the pair of knobs immediately beneath the gain. Try the left one (Spectral Region Time Fade) first. As you increase this, you should see the Spectrogram information change — it’s crossfading into and out of your spectral edit, to make the results sound smoother. By adjusting this setting and the width of the edit box, you can target the offending sounds relatively unobtrusively. The right knob does the same thing but on the vertical axis, crossfading between the frequencies inside the box and those outside it. In this acoustic guitar example, I found this control essential to get my edits sounding reasonably transparent — I dragged and placed the box slightly lower than I had done when initially selecting the scrape, and then increased this setting to create a gentle fade that didn’t draw attention to itself.

It can be bit painstaking going through a part to refine every bang and scrape, but if you listen through the results a few times and refine your edits according to what you hear, it will soon become easier — and it can work very well. The trick in this instance was to allow enough of the lower frequencies through that the edit didn’t stick out like a sore thumb, and then to use the time/frequency fades to refine the result. Don’t be scared to experiment though — spectral editing is not a destructive process; if you disappear too far down the rabbit hole, you can delete any spectral edit via the right-click menu!

**Final Thoughts**

This is just one example to get you started. You’ll want to experiment with the gate and compressor knobs (I don’t have the space to cover them here). And note that different problems and different material may require a very different approach. For example, it’s not a problem to hear a little fretboard squeak in a guitar part, so you might just want to attenuate it, but if a dialogue part for a period drama includes the sound of an aeroplane captured via the rear lobe of a shotgun mic, you’ll want no trace of the plane left behind! In that particular case, I’d suggest using a more capable spectral editor like RX6, but you can add as many of Reaper’s spectral edits as you wish, and it’s possible to get quite surgical by using lots of small edits.

Having this facility in the project window really is welcome — and I can’t wait to see what new features are added to Reaper’s spectral editing in future updates. 

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Screen 2. The same part, but with adjustments to the Spectrogram made via the Peaks Display Settings window, to emphasise the problem area.

Screen 3. I added a Spectral Edit via the right-click menu, and then used the Spectral Region Gain to reduce the level of the scratch, before using the Spectral Region Time Fade, and Spectral Region Frequency Fade knobs to make the edit less intrusive.

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The mix is where all your musical efforts come together into a satisfying sonic experience — or at least, that's the ideal. Fortunately, Sonar has several tools that can help you with the mixing process, so let's put them to good use.

Thanks For The Memories
Mix Recall is very useful, but it is not 'clip edit recall' — it’s a function that comes into play only when the arrangement is complete and you're ready to mix. While editing, I’ve usually done a rough mix so, when it's time to do the 'real' mix, the first step is taking a Mix Recall snapshot so I can always reel myself in if the song strays too far from its roots.

As you build up a collection of mixes, you’ll likely want to compare them. However if the mixes are complex, it can take several seconds to recall them. The solution is simple: call up each mix, bounce it to its own track, then choose Exclusive Solo. Now you can compare tracks instantly, simply by hitting the Solo button for the track you want to hear.

Another advantage of this approach is if, for example, you nailed the verse on one mix but not the others, you can split just that section, and insert it in a mix where everything else is right. Also, it's possible to recall only certain elements of a mix. So if you achieved perfect drum levels in one mix but you much prefer another mix except for the drums, you can bring in only the automation for the drums.

Mix Recall was introduced some time ago, but has undergone numerous enhancements over the past few years. I highly recommend looking over the documentation so that you don't miss out on taking advantage of its talents; the February 2016 Sonar column covered Mix Recall in quite a bit of detail.

Console Yourself
Because Sonar's ProChannel (Professional and Platinum versions) lets you choose a mixer architecture, it helps to have a strategy before you start mixing. For example, the three different Console Emulators have different characteristics, and you don’t necessarily want to use the same emulation on all channels. I find the S-type emulation the least ‘aggressive’ of the three, and tend to favour it for drums — whereas the A-type can get more ‘sizzly’. The N-type usually gets the call for vocals.

I’ve run tests on the Console Emulator, and found that it’s not only about creating the non-linearities inherent in the electronics of analogue gear. There also seems to be input transformer emulation; like tubes, audio transformers are processors that generate distortion, in particular at lower frequencies.

If you turn the Console Emulator’s Drive all the way up, it’s like saturating an inductor. Although people generally don’t use the CE as an effect, when you want the bass to ‘speak’ during a mix, insert the S-type in the bass track, turn up the Drive control all the way, and hit the input hard. You’ll thank me.

The Console Emulator isn’t the only ProChannel module with hidden talents, but you need to know how to take full advantage of what the ProChannel modules offer. For example, the four QuadCurve EQ curves affect the sound in very different ways, and some curves are more suitable for certain instruments and contexts than others. To learn what these curves do, I highly recommend running white noise through the curves (there’s a free white-noise generator FX Chain you can download in the 2017.10 update) and varying the controls — this makes it easy to hear, for example, how Bandwidth affects the sound when you’re boosting or cutting.

The ProChannel is also where you can choose whether or not to use tape emulation, either on the entire mix or on individual channels — or all channels, if that’s what you prefer. However, when mixing it’s important to remember that both Tape Emulation and Console Emulation are cumulative. I recommend not adjusting each one so it’s obvious on any individual track, but rather, when all of

The fade-out consists of two fade curves, one after another. The second one is a ‘Fast Curve’.
them are enabled they should make a difference compared to having all of them bypassed.

You can bypass all of a certain type of ProChannel effect at once; while holding Ctrl, click on the bypass (power) button for one of the modules in a track that's not selected, and the same operation will happen to all modules. However, the modules have to be the same type — for example, doing this with the modules in a track will not affect those in busses, and vice versa.

**Chain Keep Us Together**

Of course this is simply personal preference, but I generally use a limited number of plug-ins while mixing. This seems to give a more natural sound, and puts fewer 'layers' between the performance and the listener. However, there are certain plug-in combinations that keep ending up in my projects; most are ProChannel, except as noted:

**Drums:** Concrete Limiter / QuadCurve EQ / FX Rack (drum-room mic FX Chain) / Sizzle Bus FX Chain (on drum bus).

**Bass:** Lunar Limiter / Tube / QuadCurve EQ.

**Electric Guitar:** FX Rack (CA-X Amp) / ProChannel QuadCurve EQ (post-FX Rack).

**Voice:** Concrete Limiter / CA-2A / QuadCurve EQ / Breverb

**Acoustic Guitar:** FX Rack (CA-X Acoustic Piezo)

**Percussion:** QuadCurve EQ / FX Rack (Sonitus Delay)

**Keyboards:** Channel Panner FX Chain

**The Going Rate**

This tip is not a panacea for improving sound, but it’s underrated as a way to clean up your sound significantly with certain types of audio. In brief: recording at 96kHz doesn’t improve sound quality much, if at all, when recording acoustic or electric instruments. However, sounds generated ‘inside the box’ with significant harmonics can push audio above the clock frequency, resulting in foldover distortion or aliasing that adds a type of ‘woolliness’ to the sound.

Using upsampling on harmonic-rich soft-synths and high-gain amp sims can make a significant improvement in sound quality — in some cases, the quality may actually be better than oversampling inside the plug-in. However, note that oversampling stresses out the CPU more with real-time playback, so don’t use it if you don’t need it. (To hear just how much oversampling affects a synth’s sound, check out this video: [https://youtu.be/ wq6l0YAK9k].)

**Love Me Render**

This is another personal preference, but I prefer rendering all instrument tracks to audio prior to mixing, either by bouncing, recording a software instrument into its own audio track, or recording into an aux track. I do this anyway as a ‘fail safe’ backup for instrument tracks — who knows if the soft-synth manufacturer will be around a year from now, or if it will be compatible with the latest Windows OS? Then, during mixdown, you can archive the instrument and MIDI track to free up CPU for plug-ins that require considerable CPU power (like the L-Phase EQ and L-Phase Multiband in linear-phase mode).

**Level The Playing Field**

There is much controversy about whether you should master within a project, or regard it as an external process. Suffice it to say that as an award-winning (ahem) mastering engineer, I’ve struck a balance of concentration on the mix until it’s finished, bouncing it to stereo, and then often — but not always — mastering the final stereo mix within Sonar.

Usually this involves a little EQ, and some loudness maximising from the Waves L3 Multimaximizer. So, at this point, I want the levels hitting the master fader to be relatively close to 0dBFS, because now we’re interfacing with the outside world where we don’t have the luxury of 64-bit floating-point calculations.

If you experience ‘level creep’ during mixing, Quick Grouping is the perfect antidote. In Console View, monitor the master bus meter. Hold Ctrl, then click on an audio track’s fader that is not selected. This essentially becomes your ‘group fader’ and all audio channel levels will follow the fader. Bring the fader either up or down until the master channel’s meter is hitting -2dBFS, or whatever is your preferred amount of headroom. Now you can introduce dynamics control in the master bus, with a predictable input level.

**The Perfect Fade**

Sometimes it seems like none of the stock curves work for a subjectively pleasing fade-out in the master bus’s fader automation lane, but you can ‘construct’ the ideal fade-out easily.

Add a node where you want the song to start fading, then add another node where you want the song to have faded out. Drag the second node down to zero volume, as well as any nodes past this node. Next, add a third node equidistant between the fade-out start and end nodes.

Right-click on the envelope between the fade-out start and mid-point, and then choose Slow Curve. Finally, right-click on the envelope between the mid-point and fade-out end, and then choose Fast Curve. This results in a ‘composite’ curve that I’ve been using in all my recent songs, and it works very well to give a smooth, even fade-out. 🎵

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**Mix Recall has undergone numerous enhancements, including its handling of virtual instruments. You can specify what you want recalled in the Settings menu.**
Turning The Tables

We explore Reason’s new flagship synth: Europa.

Simon Sherbourne

In the early sealed world of the Reason Rack new versions generally centred around new devices. Rack Extensions changed all that, and we’ve had a few years of updates that have developed the app’s core DAW functionality. So it’s really exciting to get an old-school device drop that raises the baseline instrument list for all users.

By Jove

Reason 10 has two new A-grade synths: Europa and Grain. Grain is an absolute beauty: a granular scanning synth which uses any sample as its sonic starting point. We’ll return to it soon. Europa is a versatile, general-purpose synth that takes over the top-dog spot from Thor. Despite the possible name connotations, it’s neither a Jupiter clone nor modern modular, it’s at heart a wavetable synth with multi-stage sonic shaping and modulation powers.

Europa has three identical voice engines, although you rarely need all of them given the rich sonic range available to each. The signal path is fixed and fairly simple: there’s no user-definable audio routing as with Thor. Instead each source — which can already be harmonically rich and dynamic — can be mangled and shaped repeatedly through various stages before being spat out into the shared filter, amp envelope, and effects.

The modulation matrix is the same as the one found in Propellerhead’s Parsec synth, which was itself a refinement of Thor’s patching grid. Grain has this also, and they both share the same four completely flexible multi-stage envelopes and five master effects modules (which can be re-ordered by drag and drop).

Give Us A Wave

Europa’s sound sources are selected and controlled in the top-left panel titled ‘Wave’. This section has a real-time waveform display, below which is a pop-up menu for selecting the wave type. In screen 1 you can see that this has a main list and some sub-folders, one of which is ‘Wavetable’. In fact they’re all wavetables, but the ones in the folder are of the type more commonly associated with wavetable synth: collections of mostly complex waveforms providing rich timbral starting points.

The first four options in the list provide all the usual sonic fodder for analogue subtractive synths: sweeping through the ‘Basic Analog’ wavetable (using the Shape knob) morphs you from Sine to Saw, passing through Triangle and Square on the way. Pulse Width, unsurprisingly, gives you a square wave with sweepable width. The Saw-Triangle mode is really useful: visually a triangle wave that can ‘lean over’ in either direction until it becomes a saw or ramp, providing anything from a soft PWM-style swirl to a progressive bite.

We then leave more familiar territory for some specific and characterful starting points. ‘Game’ is a series of stepped waveforms just waiting to be modulated for chip-tune bleeps. Sync’ed Sine and Formant Sweep are fairly self-explanatory, as is Electro Mechanic, which provides shaped sine timbres for electric piano sounds.

Then things get more interesting with Vocal Cord and Karplus-Strong. These waves, as well as all those in the Noise folder, have built-in movement even before you get started with other modulations.

1. As well as traditional analogue waveforms and complex tables, there are some interesting highlights like Karplus-Strong: a source of animated plucks and percussion.

Vocal Cord is a particularly pleasing soft saw with noise and the merest hint of an ‘ah’ formant. It can form the basis of really pretty poly patches. Karplus-Strong is a toolbox for physical-modelled pluck and percussion sounds and could almost have been the basis of a synth in its own right.

If nothing on the menu appeals, the Envelope 3-4 option lets you draw your own waveforms using the Envelope designer. Whatever shapes are currently set in these two envelope slots will appear at either end of the Shape control’s travel, and you can morph between the two.

Shape Up

On board each of the three wave generators you have two switchable Modifiers that can get in and meddle with your waveforms right at the source. Screen 2 shows the range of options, again with some sub-foldered families. There are various ways that you can simulate transformations and cross-modulations of the oscillator with itself, such as Sync, Multiply, Invert and Mirror. You can also down-sample, bit-crunch, phase-distort and modulate with noise.
It’s an over-simplification to say that the Shaping options are varieties of drive and distortion, but that’s part of it. The glorious thing is that you can see exactly how the modifiers are operating on the waveform in the display. It’s informative and fascinating to just start with a sine wave then flick through the modes and see exactly what they’re doing.

Harmonic partials of various intervals can be added with the Harmonic Modifiers. Alternatively there are five ratios of FM you can apply for anharmonic glassy tones and bells. Finally, the Detune folder supplies unison timbres. And if you can’t choose just one, both Modifiers offer all the same modes, so you can have your cake and eat it.

Partial Solution
As you can imagine, all these options can quickly lead to a very harmonically complex starting point, which Europa tempers with some serious filtering power. The Spectral Filter has a host of modes, all of which can sculpt your source sound partial by partial by re-generating the wavetable. All the usual filter flavours are represented as well as Coast and Resonance filters. The Dual Peak and Formant filters are particularly useful, both re-purposing the Resonance control to adjust the shape. As with the oscillators, you can build your own filter type via the Envelope designer.

Paired with the Spectral Filter is a Harmonics section which further moulds the spectral fingerprint of the signal. This is shown superimposed on the filter shape display, but operate whether or not the filter is enabled. The Position and Amount controls let you shift this harmonic template around and adjust its intensity. The Ensemble and HP Noise modes have built-in movement. As with the Parsec synth, there’s a Stretch mode that moves the intervals between the harmonics with really interesting results.

Modulate
Often with wavetable synths, modulation is key to transforming bright, angular starting points into big organic sounds. Europa is brimming with modulation capabilities. There are three fairly standard LFOs, all of which can be beat- and key-sync’ed. The four envelopes themselves can also be pressed into service as custom LFOs by putting them into Loop mode, again with sync options. All these modulators have global modes, allowing you to keep them in sync across multiple voices.

Envelopes can be created in any shape you like, or there are a number of presets. By double-clicking on an envelope graph you can add breakpoints, which you can then drag around freely. The Interconnecting gradients can be curved by clicking and dragging. If you want a Sustain stage, you enable it from the button, then place the marker on the point where you wish your envelope to pause. If you prefer to sketch your own Envelope freehand, hold Command (Mac) or Ctrl (Windows) and draw straight into the graph display.

With both Europa and Grain, custom envelopes are incredibly handy for making sounds move in exactly the way you want. You can manually play with controls to find a nice dynamic effect, then automate the same move by making an envelope that does the same thing. It’s surprising how limiting a traditional ADSR feels afterwards.

Europa has a mod matrix, but adding movement is much easier thanks to the addition of local modulation assignment controls on many key parameters. The Wave Shape position, and both filters, are pre-assigned with modulation sources, and you can add velocity scaling via dedicated knobs. These assignments are not fixed: you can click on any of them and choose from the list of all modulators and performance inputs (velocity, key scaling, mod wheel, etc). The mod amount knobs on these local controls aren’t bi-polar, but you can choose inverted versions of the mod sources from the list.

Complete control of modulation (and routing of CV inputs) is available in the matrix: it should be simple to understand if you’ve mastered Thor or Parsec. Each ‘slot’ enables you to assign sources to one or two parameters, set the depths, and then add another mod or data source to scale the mod amount. There are some interesting options to explore such as combinations of the modulators and a random source. The Polyphony source is also useful for trimming the gain of patches that distort when you hold lots of keys.

With just the tiniest touch of modulation to the Wave Shape you can create lovely, interesting patches that move and swirl, or you can slam rhythmic envelopes into the Modifiers for intense dubstep nastiness. Reason’s new synths should give us plenty to get our teeth into for the months to come. 

www.soundonsound.com / December 2017
We show you how you can treat Live as one big modular synth.

Len Sasso

This is our second look at emulating modular synthesis techniques in Live. In the May 2015 Live column we explored using software modular synths in Live. Here I’ll concentrate on modular synth-building techniques using Live Device Racks. The key advantages of modular synthesis are being able to select which modules to use and how to patch them together. Live excels at the first; any Live device or third-party plug-in as well as any Rack you build from them can be used as a module. Patching modules together is a bit more limited, although with the aid of various Max For Live MIDI devices, it is still quite powerful.

Getting Started

Let’s start by building an instrument from two devices from Live’s Core Library Pack: ‘Duetta’ from the Instruments/Simpler/ Ambient & Evolving menu and ‘Dual Osc1 Wow Bass’ from the Instruments/Analog/ Bass menu. Duetta is a one-voice sustained pad with a stacked-fifth sound, and Wow Bass sounds a bit like bass vocoded with the word ‘Wow’. To combine them, create an empty MIDI track and drag Duetta to the track’s Instrument drop area. Select Duetta and enclose it in an Instrument Rack (Command-G/Control-G). Reveal the Rack’s Chain List and drag Wow Bass, also a one-voice sound, to the Chain List’s drop area to create a new chain. Next insert Live’s Auto Filter effect after the Instrument Rack and use its Side-chain input to make the filter envelope follow Wow Bass’s level. To reveal the side-chain, click the arrow at the top-left of Auto Filter and then click the Side-chain button. From the Audio From drop-down menus select your track and the Pre FX output of the Wow Bass chain. To hear the effect, you’ll need to adjust Auto Filter’s Gain and Envelope settings as well as the filter frequency and resonance. The side-chain’s Dry/Wet setting should be 100 percent. Once you hear the envelope, you can fine-tune it with Auto Filter’s Attack and Release settings. The elephant in the room is that, although instruments may be polyphonic, effects process all voices together. To get a sense of the problem, change Duetta’s Voices setting to two or more, play and hold one note and after the Auto Filter envelope reaches its sustain level, play and hold another note. Both notes will follow the second filter envelope, whereas with true polyphony, using Simpler’s filter for example, each note gets its own filter envelope.

Screen 1 shows a couple of alternatives to using the Wow Bass envelope. The first (labelled C) uses a dedicated instrument chain for Auto Filter’s side-chain input. This leaves you free to fiddle with the Wow Bass envelope without affecting Auto Filter. To create ADSR side-chain envelopes, an instance of Analog with only Osc 1 and Amp 1 active is the easiest (and most CPU-efficient) option. Mute the output of the Analog chain (speaker icon) so that it feeds only the side-chain and then edit the Amp1 envelope to suit. Analog’s Amp envelopes can be linear or exponential and include velocity scaling for attack time and envelope amount as well as envelope looping. The second alternative (labelled D) uses Simpler in place of Analog to extract the volume envelope of an audio clip. In screen 1, Simpler is in Classic mode and loops a section of a drum clip. Warping in Transients mode syncs the rhythm to Live’s tempo. The timbre of the loop will change as it is triggered by different MIDI notes, but that will not affect its use as a side-chain input.

Gain Control

Using an effect’s side-chain is a great way to go when the effect provides that option, but
2. At the top, an envelope follower is mapped to the Instrument Rack’s chain selector to control volume. At the bottom, Max For Live MIDI effects control the Instrument Rack’s chain selector, Duetta’s pan position, and the LFO depth.

most don’t. Fortunately, the free Live Pack ‘Max For Live Essentials’ (www.ableton.com/en/packs/max-live-essentials) contains an envelope follower that you can map to most Live parameters. At the top of screen 2, the envelope follower is placed after the Analog ‘Amp Env’ chain to capture its amplitude envelope. The envelope follower’s output is then mapped to the Instrument Rack’s Chain Selector, which is set up to control the amplitude of both Duetta and Wow Bass. Alternatively, you could map the envelope follower to a Macro knob mapped to the Instrument Rack’s Chain Volume controls or insert Live’s Utility effect after the Instrument Rack and map the envelope follower to its Gain knob.

Max For Live Essentials also contains a variety of MIDI utilities including an LFO, an ADSR envelope, an X-Y controller and a mapper for common MIDI parameters such as Velocity and Modwheel. Devices requiring MIDI input, like the ADSR envelope and MIDI mapper, are provided as MIDI effects, whereas those requiring audio input, such as the envelope follower, are provided as audio effects. Devices that require no input, like the LFO and X-Y controller, are provided in both forms so you can insert them where it’s most convenient.

3. NoteModulator, with help from Live’s Pitch and Scale MIDI Effects, turns kit-piece loops into tonal sequences.

The setup at the bottom of screen 2 uses three devices from the Pack’s ‘Control Device MIDI’ section — Envelope, Expression Control and LFO MIDI — to manage volume, panning and LFO amount for the Duetta & Wow Bass Instrument Rack. I’ve inserted the three MIDI devices in a separate Instrument Rack chain, but you could also place them before the Instrument Rack. The Envelope effect is mapped to the Instrument Rack’s chain selector and the LFO effect is mapped to the Rack’s pan control for the Duetta chain. The Expression Control effect maps the MIDI Modwheel to the LFO’s Depth control.

Drum Step

Modular synths typically include step-sequencers as well as various kinds of note processors. Max For Live’s MIDI note generators and processors and Live’s MIDI effects offer similar options. In screen 3 I’ve used one of my favourite free Max For Live MIDI devices, Robert Henke’s NoteModulator (www.roberthenke.com/technology/m4l.html), to generate instrument loops from individual kit-piece lanes of MIDI drum clips. NoteModulator modifies the pitch, velocity and length of incoming MIDI notes according to its 16 step settings. For example, feed it the hi-hat lane of your drum loop and send its output to a trumpet preset, and you’ve got a trumpet loop following the hi-hat rhythm.

To get the most out of NoteModulator it helps to make some pitch adjustments on the way in and out as shown in screen 3. On the way in, Live’s Pitch effect selects a single note — the kit-piece note — and transposes it to C3. For that, the Pitch effect’s Range is set to zero and a Macro knob adjusts its Pitch and Lowest settings by equal amounts but in opposite directions. You can then feed in the whole drum loop, use the Macro knob to choose the kit piece being processed and always get the same pitch sequence on the way out. NoteModulator transposes the incoming string of notes by the amounts shown in the row labelled ‘P’. The row labelled ‘V’ represents positive or negative velocity offsets, and the row labelled ‘T’, together with the Fix buttons below it, sets the note length when the Fix button is on and has no effect otherwise. On the way out, the Scale effect transposes and scale-corrections that sequence. NoteModulator lets you specify the number of steps, mute individual steps and link step timing to song position, clock or note count. If you use note count and match the number of steps to the number of notes in the kit-piece row, the two will stay in sync; otherwise the kit piece and NoteModulator sequences may shift against each other — not necessarily a bad thing. You can reset NoteModulator manually or use MIDI Note 0 (C-2) or have it reset every so many bars.

The simplest way to save your favourite modular setups is to select all the devices, create a Device Rack (Command-G/Control-G) and save that. Creating a separate Rack for the MIDI Effect devices at the beginning and for the Audio Effect devices at the end lets you separately save those as well as the Instrument Rack. Better still, create a shell Rack consisting of empty MIDI Effect, Instrument and Audio Effect Racks and use that as the starting point for your modular creations.
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Q & A

YOUR TECHNICAL QUESTIONS ANSWERED

Q Will my audio interface reduce the sound quality of a good external ADAT converter?

I have a fairly cheap and cheerful audio interface that has ADAT ports to allow me to add more inputs and outputs. If I were to use these to hook up a more expensive interface with better mic preamps, will I see an increase in quality, or can the cheap interface’s converters still compromise the sound quality anyway?

Via SOS web site

SOS Technical Editor Hugh Robjohns replies: Assuming you connect a higher-quality device you should see an improvement because you’re not at the mercy of any A-D/D-A conversion! Inside your cheap (or any other) interface, there are a bunch of A-D and D-A converters that convert between digital and analogue signals for the unit’s onboard analogue inputs and outputs. These have nothing to do with the interface’s digital (typically AES, S/PDIF and/or ADAT) inputs and outputs.

The digital signals to and from these converters and all of the digital ports are routed through some form of DSP, which handles the task of data reformatting needed to translate between the computer’s USB/Firewire/Thunderbolt ‘language’ and the ‘language’ required by the interface’s converters and digital ports. Think of it not as ‘conversion’, but as ‘reformatting’ or ‘repackaging’. The digital audio data from the ADAT socket is completely unmolested; it’s just repackaged differently as it is passed on to the computer. It is most definitely not converted to analogue and then re-encoded back to digital.

So, in the scenario you describe, the cheaper interface is providing the interface between the incoming ADAT data from the higher-quality device and the cheaper device’s USB or Firewire (or whatever other protocol) connection to your computer. The digital audio from the source remains as digital audio throughout that ‘translation’ process, and the only ‘conversion’ that could potentially be incurred is a sample-rate conversion, if the interface provides that facility — and even that remains entirely within the digital domain.

Few interfaces offer that facility anyway, which is why it is important to set the clocking options correctly on the interface and external ADAT source!

Ideally, assuming you’re using the preamps on both your cheap interface and the ADAT-equipped mic preamp box, the cheap audio interface would be set as the clock master, the expensive (external ADAT source) one as the slave, and a synchronising word clock signal taken from the interface’s clock-out port to the expensive interface clock-in. That assumes that the A-Ds and D-A’s in the cheap interface perform best when running on their internal crystal clock, while the external ADAT source has a decent clock-recovery system. If you’re not using any of the analogue inputs on your cheaper interface and your interface allows, you should set the higher-quality unit as the clock master.

Q How do I remove the noise from my lavalier mic recordings?

I need to do some recording of spoken word, using a cheap and cheerful lavalier mic. The actual tonal character of the thing is perfectly good but it’s noisy — there’s a steady ‘hiss’. It’s nothing too offensive, but still, I don’t want that in my recordings!

If I EQ the hiss out then the sound of what I want to hear suffers. Is there some plug-in that can reduce said noise, perhaps by taking a sample of it and then adding it out of phase? I’m using Cubase 7 on a PC, and looking to spend little or no money!

Via SOS web site

SOS Reviews Editor Matt Houghton replies: You can find noise-removal software that can tackle hiss for little or no money. But it always leaves you with a compromise: remove all the noise, and you’ll leave far too many artifacts; remove too little and you can avoid the artifacts but the noise will still sound annoying.

So if you’re yet to do this recording, I’d invest in (or borrow) a better lavalier mic. About £100-150 should get you something perfectly usable by Sennheiser or Rode, for example. If you’re planning to do this sort of work a lot, then such a mic will easily earn its keep. And if you don’t do this a lot, then you can sell the mic after the session and claw back most of your investment. If you still feel you can’t afford a better lavalier, though, consider using another higher-quality mic you already own — a directional (cardioid or super-cardioid) mic pointing at the source should work; and if the source is on-camera, just keep it out of shot, either above or below.

If you or someone else has already done the recording, and it’s suffering from such noise, then you could try Cockos’s ReaFIR plug-in. It’s free for Windows users as part of the ReaPlugs pack, and you can use it within Cubase. For both Mac and Windows, it’s included in the modest price of admission for their DAW, Reaper. The only other free noise-removal facility I can recall at the time of writing is the one that’s built into Audacity, though I’ve not used it myself. If spending money, you could invest in any number of paid noise-removal plug-ins, most of which operate in a similar way to ReaFIR is typical of the FIR-based noise reduction software that captures and analyses a ‘fingerprint’ of the offending noise and then attempts to remove it from the wanted signal.

But no such processor will give you perfect results — it’s much better to eliminate noise at source where possible.

Many interfaces have ADAT connections that allow you to connect external mic preamps and converters — and as ADAT is a digital format, a cheap interface will not compromise the quality of a more expensive unit connected to it via ADAT.

December 2017 / www.soundonsound.com
AWARDS

Chosen by the readers of Sound On Sound

Starting this year, the SOS Awards nominations are now taken exclusively* from products that came on sale or were tested and reviewed by Sound On Sound in the previous year (for this year that means the 12 months prior to September 2017).

This year’s nominations were unveiled for voting on the dedicated section of the Sound On Sound website on Thursday 21st September 2017 and will remain open until midnight on 30th November 2017.

THE CATEGORIES:

Audio Interfaces
DAW
Effects & Processing Hardware
Guitar Technology
Plug-ins
Music Software
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Mic Preamps
Software Instruments
Headphones
Live Sound Products

We very much look forward to seeing your choices for all the best new products of the last year in music technology and recording.

www.sosawards.com
(voting closes on 30th November 2017)

*The DAW nominations retain the same criteria as before, with all the major players listed. With some DAWs now being available on a subscription model and most having at least a ‘point’ update during the year, we felt this was the only fair solution that would give all voters the chance to express a preference.
Q & A

Why does one of my speakers appear to play back louder?

I have a set of Yamaha HS8 monitors and I set them up on my new desk the other day and noticed that the left speaker was way louder than the right. They were both on the same volume setting from the back panel so I decided to calibrate them using pink noise (at -20dBFS) with my interface master volume at its centre position, aiming at 80dB SPL in the listening position from each speaker using a dB meter app. I ended up turning the right speaker’s volume knob up about half way to get 80dB SPL, while the left only needed less than a quarter.

So my question is, is it normal to have this amount of volume discrepancy between two speakers?

Darryl Portelli via email

Hugh Robjohns replies: The short answer is that small differences in the sensitivity settings are not uncommon, but what you’re describing does sound excessive. So the first thing I’d check is that the speakers themselves, and their connecting cables, are all OK. A failing bass driver or one-legged balanced cable (where one of the two signal wires has broken) will reduce the overall volume dramatically but, bizarrely, may not be immediately obvious when listening in stereo. The easiest way to make these checks is to swap the speakers over (having set their rear-panel controls the same), and then swap their cables. If the louder side moves you’ll have found the cause!

And talking of rear-panel switches, having one set for quarter space and the other for free space can make a big difference to the amount of low end they produce, so take care over choosing and implementing any configuration settings.

If you’re confident that the speakers are set up and working identically, and the cables are good, then the issue will either be upstream (in the monitor controller, audio interface or DAW software — and again, channel swapping will usually identify the culprit) or in the room’s own acoustics. If the speakers aren’t set up symmetrically in the room, one speaker might end up with more boundary-effect bass boost than the other, for example, so some experimentation with placement, and trying to achieve a more symmetrical layout, would be worthwhile.

It’s also possible that the measurement microphone is capturing different levels from each speaker because it’s in a node from one and an antinode from the other! In your case, though, you can hear the problem so it’s obviously more than just a measurement issue — but this kind of thing can be a real problem, especially when aligning subwoofers. For that reason, when aligning satellite speakers it’s always best to use band-limited pink noise, rather than full-range noise. The benefit is that the absence of low end in the test signal avoids most issues with LF room nodes, and the constrained high-end minimises issues connected with strong local HF reflections. The Bluesky speaker company have some very useful audio test files available for free download here: http://abluesky.com/support/blue-sky-calibration-test-files/. Use the band-limited (500Hz-2.5kHz) pink noise file when calibrating your main speakers. And finally, when working with a normal stereo speaker array, I prefer to only measure and calibrate one speaker rather than both. The reason is that the inherent meter tolerance when reading a noise signal means the two speakers will always end up being slightly different — enough to pull the phantom centre off slightly to one side. So, I align one with the meter, and then adjust the second by ear while listening to a (dual) mono signal until the phantom centre image is actually mid-way between the two speakers.

Must my cables be PAT tested?

I’m looking to get some kit PAT tested at the moment and am getting some quotes, but everyone wants to know how many items there are to be tested? It’s not clear to me if a non-captive cable (e.g. a standard IEC lead) and a mains-powered unit are tested separately or together? So if I have a power amp with an IEC mains lead, would that count as two tests or one?

Hugh Robjohns replies: As far as the PAT (Portable Appliance Testing) regulations are concerned, anything that can be connected to the mains, either directly or through a removable cable, must be tested (and the results logged), and that includes each separate mains lead and mains plug-boards. If you think about it, it must be done that way because there’s no point in having a nice safe PAT-tested device powered from an untested and potentially broken and unsafe IEC mains lead! So if your electrician charges per item then you may well need to double your budget as testing will be needed for each electrical piece of gear and each individual removable mains lead.

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Orchestral performance and the role of modern technology.

HUGH ROBJOHNS

I was rather disappointed recently after an orchestral concert in one of the UK’s best concert halls, and the occasion highlighted the incongruity between traditional classical music composed specifically to be performed in front of an audience, and many ‘contemporary classical works’, which still use orchestral elements but are fundamentally composed to be recorded. This latter approach features heavily in soundtracks for feature films as well, of course, and the key difference is that it permits the balance between instruments or orchestral sections to be altered radically from anything that can be achieved naturally on stage.

The concert was a performance of ‘The Armed Man: A Mass for Peace’ conducted by the composer Karl Jenkins, and performed in the Symphony Hall in Birmingham. I was seated centrally about three-quarters of the way back in the stalls — a position which, from much past experience, sounds fabulous in traditional orchestral concerts. However, of particular interest to me was the programme’s inclusion of ‘Symphonic Adiemus’ — a brand-new reworking of the immensely popular original from 20 years ago. Since the original version relied heavily on the recording process (because of the inherent acoustic imbalance of its various elements), I was intrigued as to how Jenkins would rearrange things in this new symphonic form.

I feared the worst when I spied foldback monitors for the conductor, seven mics around the choir seats at the back of the stage, at least two mics above some hand percussion on stage (congas, etc) and one more for a solo vocalist. Clearly, someone decided electronic assistance was required…

The original ‘Adiemus’ comprises string orchestra, a three-part female-only choir, and a lot of orchestral and ethnic percussion. Jenkins’ programme notes explained the main impetus for the revision was to satisfy calls for an SATB choir arrangement, but that he also wanted to explore a more ‘Romantic’ style of orchestration. Consequently, he scored this revised work for 60 strings plus 12 woodwind, 12 brass and six percussionists (and a full SATB choir).

However, six percussionists, with most of them hitting things most of the time, are very loud indeed. For the performance I attended the 12 woodwind and 13 brass were also present, but only 34 strings and a 64-strong choir. Sadly, it probably required a choir of 120 and at least 60 strings to stand a chance of achieving the intended acoustic balance — and that notion was confirmed in the car on the way home when I listened to the new Decca recording of ‘Symphonic Adiemus’ purchased in the lobby after the concert. (It is a brilliant revision of the original, by the way, and well worth a listen if you enjoy Jenkins’ work.)

Sadly, even with all the choir mics and a mighty PA system, the choir was comprehensively drowned out just by the percussionists’ rhythmic hand clapping, let alone when they were let loose on all their drums! And the small string section fared no better even though they were 50 feet closer to the audience. Of course, percussion is an important element in Jenkins’ work and I could see that it needed all six percussionists to play all the parts…

According to the CD’s sleeve notes, ‘The Adiemus Symphony Orchestra of Europe’ was recorded in Budapest on July 10th this year, the London Philharmonic Choir was recorded in London on July 18th, and the percussion was recorded separately in another London studio over unspecified July dates. So these three elements were only ever heard together in the mix room, where their relative balance could be controlled artificially to achieve the desired blend — a blend which places the choir in equal prominence with the strings, with the brass and woodwind supporting and punctuating, and the complex percussion driving the music from the back.

It is a conductor’s role to balance the orchestra to achieve the desired sound, but entirely predictably, the sound — at least where I was sitting — was overwhelmingly dominated by the percussion, with strong brass contributions, but severely drowned choir and strings. Perhaps the funding wasn’t available for chorals and orchestral forces of the required magnitude, but relying on amplification really didn’t work either!

So, what does this mean for the performance of contemporary orchestral music? Does it mean Perspex screens around the percussion sections, close mics on individual instruments or sections, and PAs to allow everything to be re-balanced far beyond their normal capabilities? I’m not sure I like that idea, but I’m sure no promoter wants to overhear conversations like I did on the way out along the lines of, “The CD sounds so much better!”

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