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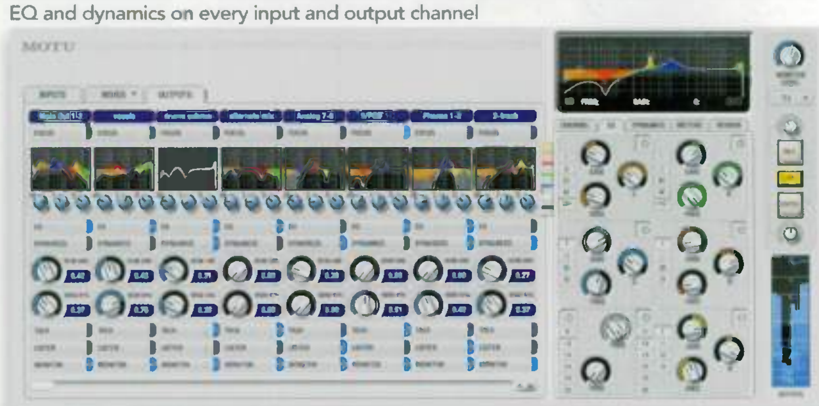

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Pick & Mix

One of the recurring themes of conversation amongst exhibitors at the recent NAMM show, especially those selling music software, was the low selling price of Apple's Logic Studio audio production software and the implications of that for their businesses. There's no denying that Logic now represents a great deal for the end user, but at the same time it might also be responsible for distorting the perception of what other software, especially plug-ins, ought to cost. With Logic 8 you get a top-shelf sequencer with over 100 integrated plug-ins and a number of other bundled software applications too, yet for the same money you may only be able to buy one or two third-party plug-ins.

The commercial reality, however, is that anyone producing a mainstream sequencer can actually afford to give you far more for your money than any individual plug-in designer, simply because of economies of scale. After all, there's only a handful of mainstream DAW packages out there but there are dozens of plug-in companies selling hundreds of different plug-ins to the same customer base. It stands to reason, therefore, that there will be far more copies of any of the popular sequencer packages sold than there are individual plug-ins, and as sales have to pay for development, speciality plug-ins that sell in smaller quantities clearly have to cost more.

Leading on from this was an underlying concern that the low cost of Logic 8 would inevitably undermine the sales of other sequencers running on the Apple platform, unless manufacturers adjusted their pricing to compete (as some have done). Of course, this is not an issue for existing users, as people tend not to 'change horses in midstream' unless they have a really good reason for doing so — there's simply too much mental investment that has to go into learning any of the major sequencer packages. Logic users also need to factor in that Apple now charge for all tech support calls, whereas many of their rivals continue to offer free support. A lower price for an equivalent package is clearly going to be a factor in attracting new users to the platform, though, and that obviously makes business sense from Apple's standpoint. It is a logical extension to giving away their entry-level Garage Band audio application with all current Macs, especially as Garage Band songs can be seamlessly imported into Logic if the user develops a serious interest in music recording.

I guess everyone will have their own view, but my take on all this is that all of the mainstream sequencers, along with their bundled plug-ins, represent a great basic construction set for producing music, but you have to accept that the same construction set will be in use by thousands of other musicians. You can make your own productions more 'individual' by adding tools that complement your personal tastes and requirements, and by driving down the price of DAW software, it could be argued that Apple have actually freed up more of the end-user's cash to be spent on adding high-quality third-party plug-in processors and instruments, which has got to be good news for both software designers and users. By the same token it places more pressure on the purveyors of 'me too' plug-ins, but then the Darwinian imperative to evolve or die applies just as firmly in the commercial world as it does in the natural one.

Paul White *Editor In Chief*

SOUND ON SOUND

Media House,
Trafalgar Way, Bar Hill,
Cambridge CB23 8SQ, UK

T +44 (0)1954 789888.
F +44 (0)1954 789895.
I ISDN +44 (0)1954 781023.
E sos@soundonsound.com
W www.soundonsound.com

editorial

E sos feedback@soundonsound.com

Editor In Chief Paul White
Technical Editor Hugh Robjohns
Features Editor Sam Inglis
Reviews Editor David Glasper
Reviews Editor Matt Houghton
Production Editor Debbie Poyser
News Editor Chris Mayes-Wright
Editorial Assistant Chris Korff

Publisher Dave Lockwood

advertising

E adsales@soundonsound.com

Group Sales Manager Robert Cottee
Classified Sales Manager Patrick Shelley
Classified Sales Executive Luci Smith

production

E graphics@soundonsound.com

Production Manager Andy Baldwin
Designers Alan Edwards, Andy Baldwin & George Hart
Classified Production Michael Groves
New Media Manager Paul Gilby

marketing

E marketing@soundonsound.com

Marketing Manager Andy Brookes

administration

E admin@soundonsound.com

Managing Director Ian Gilby
Publisher Dave Lockwood
Accounts Manager Keith Werthmann
Administration Assistant Mandy Holmes

subscriptions

E subscribe@soundonsound.com
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Circulation Administrator Laura Hinks

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in this issue

april 2008
issue 6
volume 23

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Euphonix MC Mix 94

techniques

20 **Q&A**
Your studio problems solved by *SOS* staff and contributors.

60 **Mix Rescue**
We help a home recordist on a budget to add that elusive polish to his jazz-trio recordings.

112 **Phase Demystified**
Phase cancellation can destructively interfere with recorded signals, but an understanding of the process can turn it into one of the most powerful creative tools available to you.

156 **Studio SOS**
Reader Stu Evans has a smart studio but a lacklustre drum sound. Can the team breathe life into it?

174 **Advanced Arpeggiation In Reason**
Delving beneath the surface of Reason 4's new RPG8 arpeggiator reveals a treasure trove of rhythmic modulations and variations.

180 **Making Bass Patches With Cubase's Prologue**
There's more to the Prologue synth than its presets — and it doesn't take long to create some excellent and unique sounds.

184 **Customising Sonar SFZ Files**
Understanding how to program SFZ files opens up lots of ways to better exploit what Sonar's bundled soft synths have to offer. Don't panic: it's easier than you might think...

188 **Logic's Best-kept Secret: The Hyper Editor**
Logic's Hyper Editor is a powerful way to create and manipulate MIDI data, but is often overlooked. We give you the low-down, and suggest some ways you can use it in your music.

196 **Using Templates & Real-time Backups In Pro Tools**
The focus is on better project management in this month's Pro Tools workshop, as we explain the advantages of using template Sessions, and look at some tools that can help avoid data loss.

features

38 **Classic Tracks: Frankie Goes To Hollywood 'Relax'**
The debut single from Liverpool's Frankie Goes To Hollywood was the result of adventurous production and enjoyed massive chart success — as well as creating a great deal of controversy.

76 **Guitar Technology**
We take a look at two new gizmos that are new to market: Radial's Re-amping Kit and the Waves iGTR.

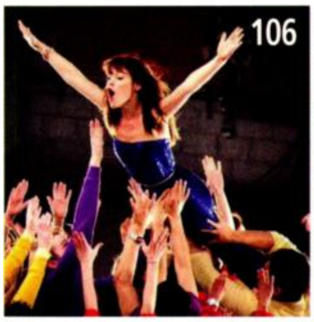
78 **The Story Of The BBC Radiophonic Workshop**
Fifty years ago this month, the most celebrated electronic music studio in the world was established. We trace the history of the Radiophonic Workshop, talking to the composers and technical staff who helped to create its unique body of work.

106 **Inside Track: Renaud Letang**
Aided by its memorable video and starring role in an iPod ad, Feist's '1234' has been a refreshingly different worldwide hit. Renaud Letang was behind the desk during recording and mixing.

142 **Chris Denman: Recording XFM Radio Sessions**
Working straight to DAT, with a minimum of equipment, Chris Denman records more bands in a year than most engineers manage in a lifetime.

170 **Playback: Readers' Music Reviewed**
The *SOS* team take time out of their hectic schedules to check out some of the hundreds of demos we get through the door.

192 **Digital Performer 6: Preview**
The days are getting longer and Spring is definitely springing, but with the announcement of Digital Performer 6, users might well feel like it's Christmas all over again.

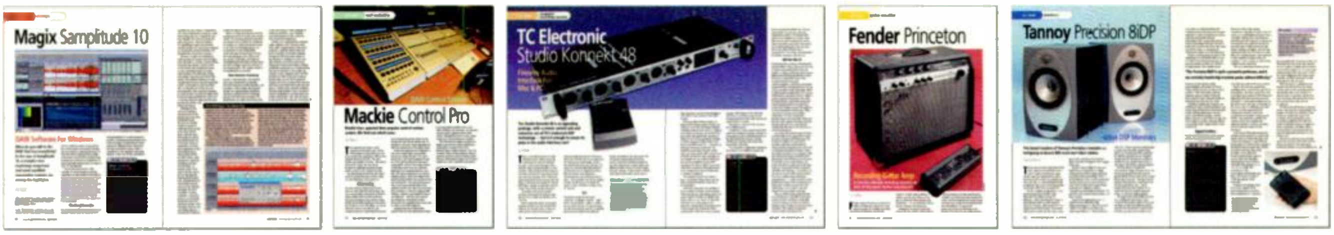


202 **Apple Notes**
Have you ever wondered why Power PC plug-ins can't run in Intel applications, or why your 32-bit plug-ins won't work in 64-bit applications of the future? Apple Notes explains all.

204 **PC Notes**
Microsoft's Vista Operating System is now a year old — but is it loved or loathed by musicians?

regulars

6 **What's New**
206 **Classified Ads**
168 **Crosstalk**
224 **Sounding Off**



product tests

54 **Aphex 207D** Dual-channel Mic Preamp

34 **Audio Ease Speakerphone** Speaker Simulation Plug-in

173 **Big Fish Audio LA Bass Sessions** Sample Library

26 **Clavia Nord C1** Digital Combo & Tonewheel Organ

132 **Eiosis E² De-esser** Sibilant Processing Plug-in

94 **Euphonix MC Mix** DAW Control Surface

124 **EWQL Gypsy, Voices Of Passion & Stormdrum 2** Software Instruments

56 **Fender Princeton** Recording Guitar Amp

172 **Loopmasters Trafik** Sample Library

164 **Mackie Control Pro** DAW Control Surface

148 **Magix Samplitude Professional 10** DAW Software

68 **MOTU MachFive 2** Software Sampler

76 **Radial X-Amp & J48** Re-amping Kit

160 **Samplebase Satellite Pro** Software Sampler

146 **SSL XLogic VHD Pre** Four-channel Mic Preamp

102 **Tannoy Precision 8iDP** Active DSP Monitors

136 **TC Electronic Studio Konnekt 48** Firewire Audio Interface

32 **Telefunken USA AK47** Multi-pattern Condenser Microphone

172 **Ueberschall Urbanic** Sample Library

77 **Waves iGTR** Portable Effects Processor

173 **XLN Audio AD Retro** Sample Library

146

54

160

68

124


32

26

MOTU unveil DP6 And update long-serving 828 audio interface

US-based hardware and software manufacturers Mark Of The Unicorn made some major announcements at the NAMM show in January. For starters, they've released Digital Performer 6, the latest version of their popular DAW software package.

As is often the case with whole-number incremental software updates, the overall look of Digital Performer has been revamped, although not so drastically that existing users will be deterred. Usefully, users can now resize the height of their DAW channels, and the left-hand inspection palettes can be customised to show important information.

But it's under the hood that the most significant changes have been made. MOTU have obviously had post-production and audio-for-film users in mind with this update, as there are new XML file-interchange facilities for use with Apple's Final Cut Pro video-editing software, as well as long-awaited support for interleaved Broadcast WAV (BWAV) files. On the pro-audio front, they've added better support for third-party Audio Units plug-ins, as well as beefing up the facilities for those who use DP6 as a front end for Pro Tools HD systems. Also, users now have the ability to 'bounce and burn' directly to an audio CD.

A neat new track-comping feature allows users to easily select and edit audio to build a new track out of a number of takes, and you can even 'comp a comp', for still more flexibility.

For the first time in Digital Performer, there's a bundled convolution reverb, called Proverb. It comes with dozens of impulse responses, and should you wish to use your own audio file as an IR, you can simply drag and drop from the Mac OS Finder window. All parameters can be adjusted in real time, so you can change the length of the IR on the fly, for example, without the plug-in 'powering down' to re-calculate. What's more, there's a Dynamic Mix feature that automatically rides the relative levels of the wet and dry outputs, to retain intelligibility when using large amounts of reverb.

DP6 also features a new compressor/limiter plug-in called Masterworks Leveler that's accurately modelled on the Teletronix LA2A levelling amplifier. Interestingly, when designing the plug-in, MOTU measured a number of LA2As and found that their LA4 opto-couplers reacted differently depending on age. So they decided to include models of a number of different LA2As, allowing you to select your favourite model from history.

For a full preview of DP6, turn to page 192 of this issue. While at the NAMM show, we were also given a sneak peek at the latest revision of the MOTU 828 Firewire audio interface, which should be available at a cost of £595 by the time you read this. New features of the 828 Mk3 include additional digital I/O, of which more in a moment, and an on-board 32-bit floating-point DSP. This allows the device to mix signals internally and apply EQ and effects. In practice, this means you can, for example, set up a zero-latency monitor mix with reverb, to help the vocalist or performer get the best possible foldback mix, which we all know can be tricky to achieve. Mixes can be set up using an updated version of MOTU's Cuemix utility, but the unit itself can be used as a stand-alone mixer, and programmed from its front panel if necessary. Using Cuemix is far easier, however, as you can draw in EQ curves (the EQ is modelled on that of the Sony Oxford console, by the way) and view and edit mixer settings in a GUI that's pleasing to the eye.

The most significant addition to the hardware complement of the 828 in the latest revision is the inclusion of a second bank of optical connections, which can operate using the standard eight-channel ADAT protocol or the 'double-rate' SMux format. This brings the total number of simultaneous inputs and outputs on the 828 Mk3 to 28 and 30 respectively at 48kHz, and 18-in, 18-out at 96kHz. However, the 828 Mk3 is capable of recording at sample rates of up to 192kHz with lower channel counts.

Also new from MOTU is Electric Keys, a sample-based virtual instrument dedicated to the faithful recreation of 50 of history's best-loved electric keyboard instruments. Electric Keys comes with a massive 40GB library of sample content, with over 20,000 24-bit/96kHz samples, and includes multisampled patches of numerous Fender Rhodes electric pianos, the Yamaha CP80, a Hammond organ, the Farfisa combo organ, the Mellotron, various Moogs and many, many more classics.

The library runs on the increasingly popular UVI engine, and is separated into 12 categories of instrument. Each category has its own 'skin', making it look like the hardware on which it's modelled. Should you need to program the sound of someone repeatedly falling onto your keyboard of choice with the sustain pedal down (which, I must say, I have often felt the urge to do), Electric Keys has a massive 256-note polyphony which, according to MOTU, ensures that "notes are never cut off due to voice-stealing".

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www.musictrack.co.uk
www.motu.com

Sontronics swing both ways

Chimera hybrid preamplifier

Sontronics have announced a new product in the form of the Chimera, an interesting single-channel mic, line and instrument preamplifier that has both tube and solid-state signal paths. Simply flick a switch on the Chimera's face, and you can transform the character of your preamp from sparkling solid-state cleanliness to silky tube warmth.

Built into a chunky box, as opposed to the long, thin rackmount form factor that we're more familiar with, the Chimera has input connections on both the front and rear panels for convenience, plus an insert point on the back for adding external dynamics processing to your signal. Front-panel controls include an input gain knob and an output level attenuator, so you can overdrive the gain stage while keeping grips on the master volume. Also, there are switches for engaging a pad and flipping the phase, as well as applying a high-pass filter to the input signal.

Sontronics say the Chimera is "100 percent made in Europe", with design and manufacturing teams in England and Portugal. It will cost £799, and will be shipping very soon. Note that this picture is of a prototype; production versions will be finished to Sontronics' typical high standard.

Also new is the STC6 handheld condenser mic. Building on the reputation of the STC5, which Sontronics launched back in 2005, the mic has a newly designed internal capsule mount and a 'multi-point electrode post', which apparently increases its signal integrity and reliability. Other features include a pad and a 75Hz high-pass filter, which are unusual for a handheld vocal mic but probably quite useful. It costs £112.

Sonic 8 +44 (0)8701 657456 www.sonic8.com
www.sontronics.com

Abbey Road launch on-line mastering venture...

...While Chandler announce new Abbey Road-inspired plug-ins

Speculation still surrounds the future of Abbey Road, Olympic Studios and LA's Capitol Studios following the announcements by EMI's new management of a massive restructuring programme within the company. At the time of writing, though, it appears that the axe has fallen mainly within the group's 40 record labels, leaving their three flagship studios relatively unscathed.

At any rate, the partnership between Abbey Road and Chandler continues to bear fruit, and on display at the NAMM show in January was the new Brilliance Pack. The three plug-ins included emulate three passive EQ designs that were widely used at Abbey Road in the '60s, helping to shape the sound of the Beatles, among many others. Two versions of the RS127 equaliser offer a choice of three corner frequencies and a boost/cut dial, while the '8k box' (which was never officially named — left) simply allows the user to apply up to 10dB of gain at 8kHz.

Mastering is an increasingly important area of Abbey Road's work, and they were previewing their new on-line service at NAMM. In operation, it seems similar to the services offered by the likes of Metropolis and eMasters: clients upload their music by FTP, pay a flat fee of £75 per track, and receive the results as a downloadable Red Book or DDP master. Visit their web site for further details.

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www.ari-mastering.com

Audix at NAMM

Gizmos and more

In the 'why has nobody thought of that before?' department, Audix's new Cab Grabber looks like a small work of genius. If you're fed up with musicians stumbling into your carefully placed mic stands — and who isn't? — this little gizmo could change your life. The Cab Grabber can be clamped on to the side or top of almost any guitar or bass amp, and sprouts a standard threaded mic connector on a curved neck that can be rotated to achieve the best position. It can support mics of up to 1lb in weight and has virtually no footprint on the stage or studio floor.

Meanwhile, the company also have a new flagship studio condenser mic. The CX212 has been developed from the existing 112, but unlike its predecessor it is a multi-pattern design offering omni, cardioid and figure-of-eight polar patterns, and can cope with very high SPLs. Shipping with a shockmount and carrying case, it will retail in the US for $599.

SCV London 020 8418 1470 www.scvlondon.co.uk
www.audixusa.com

The legend continues Akai launch MPC5000

Sampling experts Akai have announced the latest model in the MPC range of sampling workstations: the MPC5000, their new flagship product.

As expected with any MPC, the 5000 features 16 of Akai's trademark trigger pads, but unlike any other MPC it's capable of recording eight tracks of audio to hard disk. Other new features include a revamped sequencing engine, a 20-voice, three-oscillator synth with built-in arpeggiator, and a new effects engine with four buses, each of which can run two effects patches. Its 64-voice drum sampler has 64MB of sampling RAM as standard, although this can be expanded to 192MB into which samples from a Compact Flash card can be loaded. The MPC5000's screen is an improvement over that of the MPC2500, being twice the size and hinged, so audio waveforms and status information can be displayed in greater detail at an angle that suits the user.

The MPC5000 has 10 analogue outputs, as well as an ADAT optical port that can be hooked up to a D-A converter for further connections. There's also an S/PDIF input and output, alongside two MIDI inputs and four MIDI outputs, an RCA turntable input (with built-in phono preamp) and combi XLR mic/line inputs.

What's more, in addition to the ability to record onto the machine, users can chose to have a CD-R/DVD drive fitted, so they can burn their tracks directly to audio CD.

As you've probably noticed by now, it's a feature-packed machine, and one that we'll want to take a look at as soon as it's available. Keep your eyes peeled for an *SOS* review in forthcoming issues.

Also new from Akai is the XR20, a device that the manufacturers call a 'beat production station'. It's essentially a desktop sampler that's geared towards hip-hop and R&B makers, which comes with over

700 pre-loaded sounds, from drums and percussion to instrument samples and sound effects. Usefully, it can be powered by batteries and used on the move, and users can plug a microphone into it and sample 'on the fly'. The XR20 is due to become available later this year, costing £220 including VAT.

Numark +44 (0)1252 341400

www.akaipro.com

Hot Shu Is this the best way to mount a kick-drum mic?

So you need to close-mic a kick drum, but what approach do you take? Do you remove the resonant head and stick a mic inside, or mount the mic outside the drum and hope for the best? Whatever way you do it, chances are that it'll involve a microphone stand somewhere along the line, and what's to stop the guitarist (or the tea boy) from kicking the mic stand mid-session, thus losing your drum sound? This was obviously what American company Kelly Concepts were thinking when they dreamed up the design for the Shu, a handy system that could prove to be very popular.

It's a kick-drum mic-mounting system that enables any microphone to be located almost anywhere inside or outside the drum, suspended by flexible cords in a horseshoe-like cradle (hence the name). The cords clip directly on to either the external lugs, or to loops that can be fitted by the user to the inside of virtually any bass drum. As the cords are made from rubber, the actual mic-mounting section, which is constructed from sturdy but lightweight aluminium, is acoustically decoupled from the drum. Cleverly, once set up, the Shu and its paraphernalia can be left in place so you can ensure the mic ends up in the same position every time you set up. Also, the rubber cords are adjustable, and the metal mount section has a variety of points where the cords can be connected, so you can mount the Shu anywhere you like.

Currently, Kelly Concepts have no UK or European distribution, but customers can buy direct from their web site. Shipping costs will vary. The Shu has a retail price of $154, which was just under £80 at the time of writing.

Kelly Concepts +1 402 421 1169

www.kellyshu.com

Hands off my knob!

Analogue hardware legends API (**www.apiaudio.com**) have successfully Trademarked their distinctive dual-concentric knob design, which is found on every piece of their gear. After almost 40 years of operation in the pro audio market, the company decided that they should prevent other manufacturers copying the API knob, so as not to confuse the market or API's loyal customers. API President Larry Droppa commented, "we felt it was especially important to protect API now, given the many studios and engineers investing in API equipment based on its distinct sound and reputation for excellence".

Good news for Logic users

Frontier Design, the company who manufacture compact DAW-control devices, have announced that their Alpha Track single-channel control surface is now compatible with Apple's Logic DAW software. Alpha Track was launched in 2007, but wasn't initially compatible with Apple's DAW software upon release. Now, however, users of Logic Pro and Express v.7.2 and v8 can tweak, edit and automate using the Alpha Track's high-resolution fader, rotary encoders and ribbon controller. For Logic to recognise the Alpha Track, a plug-in, available for free from Frontier Design's web site (**www.frontierdesign.com**), is required.

Moog Music ditch the digital

Synth players looking for a reliable, roadworthy replacement for their old Minimoogs might want to check out the new Voyager Old School from Moog Music (**www.moogmusic.com**). It offers similar sound-generating and modulation features to the existing Voyager, with three VCOs, two envelopes, a filter and an LFO, but without the digital control and patch-recall features of its big brother.

Also new is the MP201 Multi-Pedal, a floorboard sporting four on/off switches and a rocker pedal. These can be used in a variety of configurations to control other devices through four CV/Gate outputs, or via MIDI over either USB or conventional five-pin DIN connectors.


10am already?
crap, you're late for work!
6th cup of coffee... in the last two hours.
MACKIE
dust bunnies collected from the last time your girlfriend cleaned your desk. it's been a while.
the mix you'll actually give to your buddy after he hears your new stereo. he'll never figure out why your system sounds better than his.
MIX #3
MACKIE. 402-VLZ3
bass heavy mix you'll play for your buddy to show off your new car stereo system.
MIX #4 BASSY
the marker you swiped from the office. we won't tell.
ever since your buddy borrowed them, they haven't smelled the same. be afraid. be very afraid.
dust footprint left over from your last mixer. it's really nasty, dude. you should clean it.
RECLAIM YOUR DESK SPACE.


402-VLZ3
802-VLZ3
1202-VLZ3
1402-VLZ3
1642-VLZ3
1604-VLZ3

That's the way to do it!

Earthworks PianoMic

Mic manufacturers Earthworks have developed a new system for recording grand pianos. The PianoMic, as it's called, is a unique lightweight bar that houses two miniature gooseneck microphones. The bar is adjustable in length and extends across the width of the piano, positioning the two mics directly above the strings. The goosenecks allow the mics to be aimed in almost any direction, so users can achieve the optimum mic placement.

Because the PianoMic system resides completely inside the piano, it requires no stands, but also enables the lid of the piano to be closed whilst the mics are still operating. Earthworks say that this is extremely useful in live situations, where it's often difficult to get enough gain from stand-mounted mics before feedback, but also in small studios where a band is recording live, as the amount of spill entering the piano body is minimised when the lid is closed.

Interestingly, the omnidirectional capsules used in the PianoMic system are of the 'random incidence' type, and as such are balanced for use in a diffuse soundfield, where sound comes from everywhere, rather than in a 'free field', where sound is directional. Typical applications for mics in a diffuse field include measurement and ambience recording, not close-miking, so Earthworks must have decided that the 'random incidence' mics sound better. This is quite feasible when the piano lid is shut. Watch out for a forthcoming *SOS* review on the PianoMic system to see how it measures up.

In unrelated Earthworks news, the company have announced that their mics were used on the recent and much-hyped Led Zeppelin reunion gig at the O2 Arena in London. Legendary front-of-house engineer Mick Hughes chose to use several Earthworks DK25/L drum mic sets and some Periscope P30s on Jason Bonham's drum kit, mounting the mics above and around the kit, and using a Kickpad (a passive XLR in-line equaliser/pad) on the bass drum. Talking about the mics, Hughes commented that "their extended high-frequency response gives a fantastic openness and clarity, which was perfect for Led Zeppelin's dynamic open drum sound". To read the full story, check out UK distributor Unity Audio's web site.

Unity Audio +44 (0)1440 785843
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www.earthworksaudio.com

Universal attraction

Universal Audio have released a number of new products in recent months. In collaboration with German hardware manufacturers SPL, they've created a new plug-in for the UAD platform, the aptly named SPL Transient Designer For UAD. It's a software version of the reputable hardware device of the same name, and as such has the same simple Attack and Release controls. For years, the Transient Designer has been used to shape the envelope of recorded audio and is often used on drums, where it has the ability to completely change the 'feel' of the sound. It's available now for the UAD platform, costing $199 (around £100). In other UAD news, rumours have it that Universal Audio are set to release a UAD version of Empirical Labs' Fatso in coming months. We'll keep you posted on any developments.

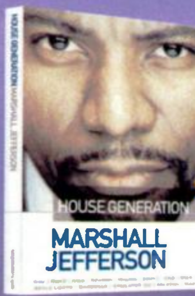
Also new from Universal Audio is a signature edition of the LA610 channel strip, of which there will only be 500 manufactured. The front panel of the special model is all black, setting it apart from the silver-faced standard model, and it sports the signature of Bill Putnam Jr, the son of Bill Putnam, who founded Universal Audio in the '60s. But inside, there are some more significant customisations. All of the tubes in each LA610 Signature Edition are hand-picked from New Old Stock (NOS). The special edition also features custom-wound Cinemag input and output transformers. No doubt it'll be a collectors' item in years to come.

Source Distribution +44 (0)20 8962 5080
www.sourcedistribution.co.uk
www.uaudio.com

Kam turn the tables

A new line of studio furniture has been announced by Kam (**www.kam.co.uk**), who specialise in equipment for DJs and musicians. The Studiodesk series comprises four products, all aimed at home- and project-studio owners. They feature height-adjustable monitor shelves, large table-tops suitable for mixing desks or synths, and platforms on which to situate computer monitors. Sliding keyboard shelves also feature, and some have space for mounting 19-inch rack equipment. Prices range from £180 for the Studiodesk 1, a basic two-shelf design with speaker shelves, to £290 for the Studiodesk 4, a single-platform desk with integrated 14U rack. For further details, check out Kam's web site.

Marshall Jefferson's private sample collection



Sample library gurus Loopmasters have launched a new product in their Artist series that features sounds from house music 'super-producer' Marshall Jefferson's personal collection.

House Generation Marshall Jefferson, as the collection is called, comprises over 1.2GB of WAV and REX 2 files, as well as patches for many popular software samplers, including Kontakt, EXS24 and NNXT. A full Reason Refill also comes with the library.

Commenting on the collection, Marshall Jefferson said "I honestly think every single loop on this CD is a hit component", adding that there are "no throwaway loops" in the library. It comes on a single DVD at a cost of £35, or it can be downloaded from **www.loopmasters.com** for £30. It's available now.

EastWest sample forbidden fruit

Sample library developers EastWest, who have bought LA's Cello Studios to use as the base for their sampling exploits, were showing a number of promising new products at NAMM. Foremost among them was Forbidden Planet, which will have obvious appeal for anyone whose tastes run to the dark and industrial side of things. It introduces a convolution technology EastWest are calling Q-Fusion, which allows the user to cross-pollinate the source material in previously unheard-of fashion, and a new take on wavetable synthesis called Riptide. Effects provision is equally innovative, with tuned feedback and ring modulation on offer among numerous other intriguing processes.

The company's Quantum Leap series has been augmented by Goliath, the sequel to their existing Colossus. Goliath repackages all the existing Colossus content within EastWest's new 64-bit Play engine and adds some 8GB of new material, making it an even more comprehensive tool for songwriters, composers and producers. The Play engine should make it possible to use more of the 600-plus patches simultaneously, since it can access more than 3GB RAM on a 64-bit computer. This will be especially welcome to users of EastWest's Quantum Leap Symphonic Orchestra, which has also been ported to the new Play system.

New too is Quantum Leap Stormdrum 2 (SD2), which builds on the success of the Stormdrum acoustic percussion instrument to produce a library more than 12GB in size, adding exotica such as bowed gongs and bowls, Indonesian hand drums, Anklungs and Udus. These are paired with more than 100 MIDI performances created with the film composer in mind. Check out the full review of Stormdrum 2 on page 126 of this very issue.

Soundsonline Europe +31 20 404 1687
www.soundsonline-europe.com

Lauten clear A trio of new mics

Lauten Audio, a relatively new US-based hardware company, have announced three new microphones. First up is the FC357, a large-diaphragm, multi-pattern mic that uses solid-state circuitry to achieve what Lauten call "very clean and natural" results. The FC357 is quite fat, in terms of physical shape, but not as chubby as the LT381, which is also new. This uses transformerless tube circuitry, employing a pair of New Old Stock (NOS) tubes in a pentode-in, triode-out configuration which, according to Lauten, "increases the clarity and transient response of the signal and reduces noise, RFI and signal loss". It's designed as a vocal microphone and, like the FC357, uses a large-diaphragm capsule and can operate with cardioid, figure-of-eight and omnidirectional polar patterns.

The final new mic from Lauten Audio is the ST221, which comes as part of a matched pair. Unusually, the ST221 is a small-diaphragm tube microphone, and as such is vented to prevent the internal components from overheating. Its capsules are interchangeable and, as standard, the mic will ship with both cardioid and omnidirectional capsules.

All three new mics will be available in the second quarter of 2008. At the time of writing, UK prices had not been announced.

Analog Audio +49 (0)8142 53980
www.analogonline.de
www.lautenaudio.com

Are 23 tubes really enough?

There are niche products, and then there's Metasonix's G1000 guitar amplifier, which also goes under a different moniker, as displayed below (**www.metasonix.com**). Described by its inventor as "not intended for middle-aged 'tone questers' who believe that they will be able to play like Eric Clapton simply by spending a lot on equipment", it features no fewer than 23 vacuum tubes, mostly of varieties never previously used in guitar amps. These are used to form two independent channels called Happy and Angry, the latter designed for "instability and raw, berzerk distortion effects", with a massive range of different sounds on offer. The amp will be available as a custom order costing around $5000 in the US.

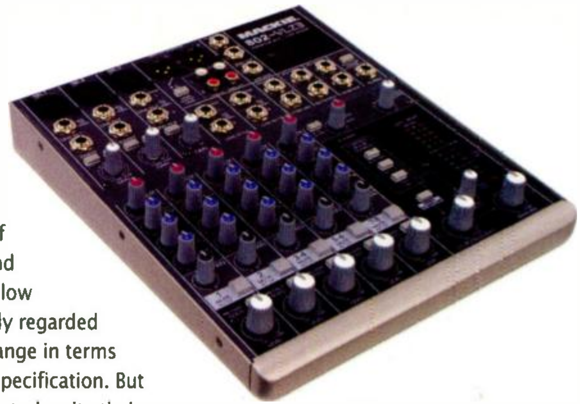
Rycote's new range

Mic-mounting aficionados Rycote have launched a new Invision range of shockmounts. There are eight models in the range, INV1 to INV8, each using Rycote's new, patented Lyre mounting system. The Lyre design comprises a pair of what Rycote call "virtually unbreakable" W-shaped mic holders, which clip onto the mic and hold it in place. Each model has different sized clips and different distances between the mic holders, and the range has been designed with popular mic choices in mind (for example, the INV3, above, is suitable for use with the Sennheiser MKH8000). To find out which Invision shockmount best suits your mic, check out Rycote's web site, **www.rycote.com**.


UF Series Version 2
UF70
CME
Professional keyboard actions, tough build-quality, comprehensive connectivity and features like aftertouch and a breath-controller input have already made CME the no.1 choice for gigging and recording musicians.
UF Series V2 keyboards add WIDI (wireless MIDI interface), increased velocity and aftertouch curves and registration banks for storing and recalling panel settings - all accessed from a sleek, new-design control panel.
There are four models available; UF50 (49 key), UF60 (61 key), and UF70 (76 key, pictured above), all with semi-weighted keyboards, and the 88-key UF80, with piano hammer-action.
For full specifications, UK prices and details of your nearest dealer, please, visit www.arbiter.co.uk.
CME - go make music.
CME
www.arbiter.co.uk

Mackie's budget beauties

New compact mixers and studio monitors



New to Mackie's VLZ3 series of compact mixers are the 402 and the 802, the smallest two products in the range. The 402 VLZ3 is a four-channel mixer with two of Mackie's acclaimed XDR2 phantom-powered preamps, which can accept mic, line and instrument signals. Each channel features a two-band EQ and rotary level control. Elsewhere on the 402's diminutive top panel, a stereo line input can take balanced or unbalanced signals on quarter-inch jacks, while tape inputs and outputs on RCA sockets allow for external recording and playback devices to be connected.

The 802 VLZ3 (above right) is larger in size than the 402, and has more features, including an Auxiliary bus, a three-band EQ, control room outputs, and XLR connectors for the master outputs rather than jack sockets. The 802 also has mute and pre-fader solo buttons, so you can monitor your signals using the headphones or the control-room outputs.

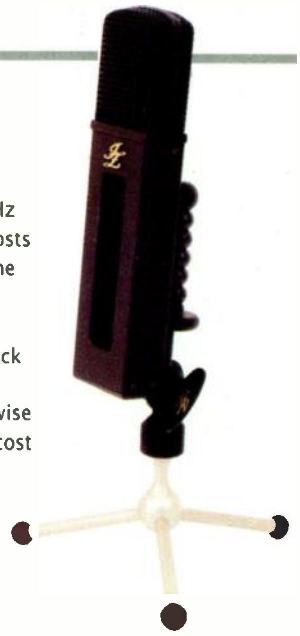
Mackie say that these are the best-sounding mixers in their class, thanks to their "high headroom/low noise" design. They should be shipping later this year, at a cost of £120 and £230 for the 402 VLZ3 and 802 VLZ3 respectively.

Also new from Mackie is the MR line of active studio monitors, which comprises an eight-inch model and a five-inch model. Both are aimed at the budget end of the market and as such sit below Mackie's highly regarded HR monitor range in terms of price and specification. But Mackie say that, despite their affordable price tag, the new range still pack a punch.

They use bi-amped designs and can be hooked up using XLR, quarter-inch jack or RCA connectors. They're also both magnetically shielded, so they can be situated close to CRT screens, and they feature sculpted baffles that Mackie say minimise diffraction and therefore improve imaging. They're priced at £360 and £560 per pair for the MR5 and MR8 respectively.

Mackie UK +44 (0)1494 557398
www.mackie.com

Latvia: Event horizon JZ Microphones' Black Hole



JZ Microphones are a newly formed mic manufacturer from Latvia whose first product is the Black Hole, a multi-pattern condenser. Originally announced under the Violet Mics brand (but not released or substantially marketed as such), the Black Hole offers a seemingly new concept in mic design. As its name suggests, the Black Hole is black in colour and has a whopping great hole through the middle. The interesting rubbery shockmount clips onto the mic using two contact points located at the top and bottom of the hole.

There is also innovation at work at the business end of the new microphone: the Black Hole has an unusual dual-capsule design and can operate with omnidirectional, cardioid and figure-of-eight patterns. The manufacturers say that it has a frequency range of 20Hz to 20kHz and can cope with a maximum SPL of 134dB. It costs 1690 Euros, which was just over £1250 at the time of writing.

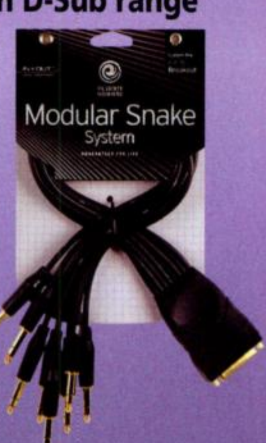
Just before we went to press, a single-pattern version of the Black Hole was announced. The Black Hole SE, as it's called, only operates in cardioid mode and has only a single capsule, but is otherwise the same as the standard version of the mic. It'll cost 1299 Euros (just under £970).

Currently, JZ Microphones do not have distribution in the UK, but you can buy directly from them in Latvia.

JZ Microphones +37 167 246648
www.jzmic.com

Planet Waves lend a hand with D-Sub range

Cable innovators Planet Waves (**www.planetwaves.com**) have unveiled a new line of cabling products based on the D-Sub (DB25) connector. The Modular Snake System, as its called, is based on what the manufacturers call the Core Cable: a five-, 10- or 25-foot eight-way D-Sub multicore. Breakout cables (such as the one pictured) can connected to each end of the Core Cable, providing a choice of connectors, including TRS jacks, male XLRs, female XLRs, and male/female XLR combinations, which allow users to mix and match parts of the system to build the perfect cable.



Cue light systems for your DAW

UK pro audio distributors Out Post Sound (**www.outpostsound.co.uk**) have announced that they will be distributing Punchlights Cue Light systems, which provide visual cue information for Digital Audio Workstations (DAWs).

Products available include the Universal Studio Display, a large timecode readout with record indicator; the Recording Display, which illuminates differently depending on whether the DAW is in record or record-ready mode; and the Recording Lamp, a classy-looking desk light, which glows different colours depending on the status of the DAW.

The products can be connected to DAWs such as Pro Tools (HD, LE and M-Powered systems), Logic, Sonar, Cubase and Nuendo, via special trigger boxes that connect to the MIDI and outputs of the connected interface.

For more details on the range, head to Out Post Sound's web site.

Olympus LS10

No, it's not a camera!

The Olympus brand has been a household name in the realms of consumer electronics, particularly with regard to photographic equipment, for years, but the Japanese company have branched out into pro audio with a neat product that may turn some heads.

The LS10 is a portable stereo audio recorder that can capture uncompressed (linear PCM) WAVs at up to 24bit/96kHz. It's also capable of recording and playing back MP3s and WMA files, and it can even play back recordings using its built-in stereo speakers. As you can see from the picture, the LS10 has a pair of microphones located at the top of its body, but an external mic can be plugged in via a mini-jack socket on the side of the device. Data can be recorded to the LS10's 2GB of on-board memory, although the device also offers a slot for inserting an SD or SDHC card of up to 32GB in capacity.

Front-panel controls include the expected play, record, stop and skip features, as well as buttons for entering and navigating menus. An A-B Repeat button enables you to loop a section of audio that's being played back, making transcribing slightly less painful and perfecting that guitar lick a breeze! The generous backlit LCD displays a reasonably large bar-graph meter, as well as information on the current status of the machine.

Down the sides of the slim aluminium body are thumb wheels for controlling the input and output volume of the device, and switches for adjusting the on-board mics' sensitivity and engaging an Automatic Gain Control (AGC) function. There's also a stereo line-level input on mini-jack.

The Olympus LS10 should be available by the time you read this, costing around £250. Keep your eyes on the pages of *SOS* for a forthcoming review.

Olympus UK +44 (0)1923 831100
www.olympus.co.uk

Get Futzed up! Down and dirty with McDSP

New from McDSP is Futz Box, a Pro Tools plug-in that presents new and exciting ways to make your audio sound worse. Recreations of everything from broken speakers to megaphones are on offer, thanks to a technology McDSP are calling Simulated Impulse Modelling. This, they say, allows them to create extremely accurate emulations of the devices in question without the CPU overhead associated with convolution processing.

Also new is the NF575 Noise Filter, designed to remove hums and buzzes from audio. Up to five notch filters can be linked to remove hums both at their fundamental frequencies and their associated harmonics. It's joined by the equally new DE555 (geddit?) de-esser. This, they say, uses intelligent analysis of the incoming signal to de-ess effectively without the need for a manual threshold setting, and a 'focus' control to target the processor's actions as accurately as possible. The plug-ins will be available as part of the McDSP Emerald Pack as soon as testing is complete.

Unity Audio +44 (0)1440 785843
www.unityaudio.co.uk
www.mcdsp.com

Fighting fire with tubes

Presonus add to their computer recording line

Three new products were in evidence on Presonus's NAMM booth. The comprehensive FireStudio Tube offers no fewer than 16 analogue inputs and 10 outputs in a 1U rackmounting Firewire interface. Two tube-based 'super channel' preamps are complemented by eight further mic preamps on the rear panel, plus six balanced line inputs. There's also MIDI I/O, a headphone output, and a matrix mixer allowing inputs to be monitored at very low latency.

The more modest AudioBox USB is a simple stereo USB interface with two mic preamps, balanced line outputs and MIDI I/O, while the Digimax 8D is a stand-alone eight-channel preamp that uses Presonus's high-quality XMAX preamp circuitry. The first two channels also feature high-impedance instrument inputs, and the output complement includes jack and ADAT optical connections.

Source Distribution +44 (0)208 962 5080 www.sourcedistribution.co.uk
www.presonus.com

VirSyn embrace VST3

It's been announced that German soft synth experts VirSyn are the first third-party company to release a product that uses Steinberg's new VST3 plug-in protocol (**www.steinberg.net**). VirSyn's Matrix Vocoder v1.1 has support for the protocol, which brings better integration with VST3-ready hosts such as Cubase and Nuendo. It uses the new native side-chaining facilities, making routing the Vocoder's carrier and modulator signals a breeze, and can be automated with single-sample accuracy, making it possible to perform the finest of tweaks. For audio examples of Matrix Vocoder, and to download the latest update, visit VirSyn's web site, **www.virsyn.de**.

Zero-G release Phaedra

The latest release from software experts Zero-G (**www.zero-g.com**) is Phaedra, a sample-based virtual analogue synth that runs on the Kontakt Player 2 sample engine. It comes with 4GB of content and over 20,000 samples, with 720 patches to help stir your imagination. Phaedra's sounds are split into 10 different categories, which each contain various patches, from conventional synths, basses and leads, to evolving sequences and complex multi-instrument setups. For more information, and to listen to a demo, check out Zero-G's web site. Phaedra is available through Time + Space in the UK. **www.timespace.com**.

SAE's boat race from Byron Bay

SAE, the world's largest audio engineering school, are to open a new institute in Oxford, UK. The Oxford school will be the fourth in the UK, but it will also serve as the school's global HQ, which will move from Byron Bay in Australia once the Oxford facility is up and running. The company have invested £12 million in the project, which will include the building of state-of-the-art facilities and provision for a number of fully residential places on SAE courses. In other SAE relocation news, the London facility is moving premises to a new £8 million building. According to SAE, it'll have a studio large enough to record a full 88-piece orchestra. Check out **www.saeuk.com** for further details.

Sony ECM957 Pro

Portable Mid/Side microphone

Sony's latest tool for the mobile sound recordist is the ECM957 Pro, a single-point stereo electret condenser microphone that operates in a Mid/Side configuration. Inside its body, which is just slightly bulkier than a typical handheld mic, is a Mid/Side matrix that converts the signals from the middle and sides capsules into conventional left/right stereo. This signal is output on a five-pin XLR socket, and a special cable that terminates in a stereo mini-jack is supplied, allowing the user to plug the mic directly into a portable recorder, such as Sony's PCM D1 or D50. Furthermore, the mic is powered by a single AA battery, meaning that you don't need a device with phantom power to use it.

Usefully, the middle capsule can be rotated through 90 degrees, enabling the mic to be used in both front- and side-address arrangements. What's more, the directional characteristics of the mic can be manipulated to give coverage of either a 90- or 120-degree sound stage.

But why, you may ask, are Sony making this product, when their acclaimed PCM range of digital recorders have high-quality condenser mics built into them? Well, the Mid/Side configuration of the ECM957 Pro provides a distinctly different response to the near-coincidental X-Y pairs found on the D1 and D50, and has a slightly more 'synthetic' sound that will suit certain situations better than the existing configurations. Also, because the mic is attached to the recorder by a cable, the recorder can be hidden from view or protected from the elements, while the mic serves its purpose.

Sony Europe +44 (0)1256 483795
www.sonybiz.net

Best Buddies SM Pro Audio launch affordable personal monitor controller

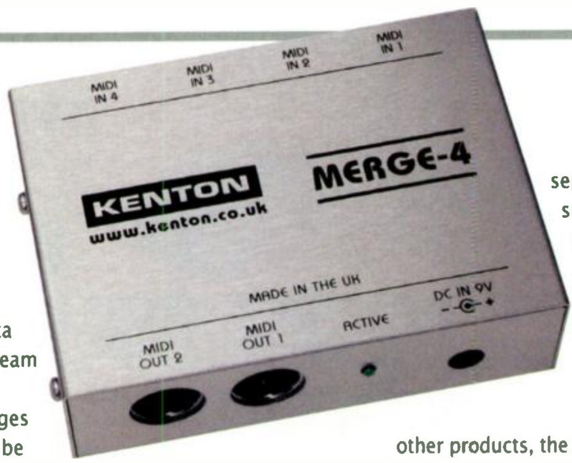
Whether you're distributing a monitor mix on stage or to a band in the studio, the ultimate in flexibility is to give each band member control over his or her own monitor level. Until now, this has been the preserve of relatively expensive pro audio gear, but SM Pro Audio's new Stage Buddy system could change that, and eliminate the need for separate mic preamps or DI boxes into the bargain.

There are two components: the Stage Buddy Remote (pictured) provides a headphone amp, two mic/line preamps with phantom power, and a simple reverb. The musician uses the 'More Me' control to balance the input signal against a monitor mix arriving at the Remote via Cat 5 cable. The Remote splits the two mic or line signals to four XLR outputs, so you can use it to feed separate front-of-house and monitor mixers. The other component of the system is the Master unit, which distributes a stereo input signal to eight Cat 5 outputs. Multiple Remotes can be daisy-chained from each Cat 5 output, so in theory you could use a single Master unit to feed no fewer than 64 Remotes. Stage Buddy Remotes are available individually, and the core product packages four of them with a single Master.

Also new are the iNano, a simple passive volume attenuator in a smart box, the PR8DS eight-channel mic preamp, and the HP6E six-way headphone amp. SM Pro Audio's sister company Violet Audio, meanwhile, were showing a new signal splitting and distribution system. Like the Stage Buddy, the DX16I and DX16IT use Cat 5 cable, allowing mic-level signals to be sent over long distances without the mess and expense of connecting multiple cables, and are significantly cheaper than existing rivals. Each provides a three-way split for 16 mic inputs, and operates entirely passively, with the IT version adding transformer isolation.

Electro Vision +44 (0)1744 745000
www.electrovision.co.uk www.smproaudio.com

Kenton MIDI Merge 4



Kenton Electronics, whose MIDI gadgets act as the plumbing in many a synth lover's studio, have released the MIDI Merge 4, a device that's designed to remove the need for complex computer-based merging of MIDI data. The Merge 4 has four MIDI inputs and two MIDI outputs, the signal from the latter of which is identical. Standard MIDI data from the four inputs is simply combined to form the data stream from the MIDI output so, for example, two MIDI controllers can be used to control the same parameter, but some messages are handled slightly differently. Messages that can't sensibly be merged, such as those for Active Sensing, MIDI Time Code and MIDI Clock, are locked to a 'master port' at whichever input first receives such a message after the unit is switched on. This ensures that important data is transmitted in full, albeit from only one MIDI port. Cleverly, System Exclusive (SysEx) messages from each input are separately buffered, so that signals from all connected equipment are sent from the MIDI outputs, but their data packets remain intact.

As with Kenton's other products, the Merge 4's metal casing is robust enough to be used on stage. It's powered by an included 9V AC-DC converter, and it's got a power indicator LED. It is available now, at a cost of just under £75 including VAT.

Kenton Electronics +44 (0)20 8544 9200
www.kenton.co.uk

Q&A

Your technical questions and queries answered

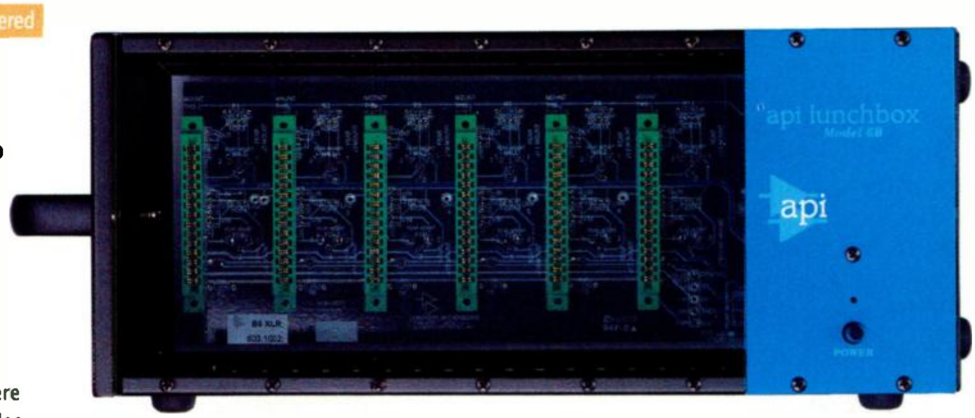
Q What's so good about a Lunchbox?

I'm seeing more and more new units for the API Lunchbox. Why is this, and what's so special about the format?
Gary Freeman

Technical Editor Hugh Robjohns replies: API's console channel strips were traditionally built from individual modules for mic preamps, equalisers, compressors and so on. The success of the consoles lead to demand to use these '500 series' modules independently, so a rackmounting chassis was designed to house a number of units, the current version housing 10 modules. The idea was that a rack of just preamps, perhaps, could be used to supplement a console's inputs, or to make a convenient location-recording rig, for example. However, not everyone wanted a large 19-inch rackmountable solution: some only wanted a handful of modules, so a more compact and self-contained unit was conceived. That was the original Lunchbox, which had space for just four modules. Its name comes from the size and shape of the case, and the provision of a carrying handle.

The current version of the Lunchbox (the 500-6B) is a little larger and can can hold up to six standard-width 500-series modules (partly to make it easier to combine single and double-width modules). It contains an integrated power supply, and provides a full set of XLR connectors to handle all the inputs and outputs. The convenience and inherent modularity has made the API Lunchbox system extremely popular, although there is nothing particularly 'special' about it — it is just well thought-out and easy to interface with, so much so that many third-party manufacturers now make compatible modules, as you suggest, to take advantage of this very flexible and well-designed system, whose backplane connectors provide all the necessary standardised power rails, grounds, I/O buses, and so on, in a well-known format.

In fact, there is a 'sticky thread' on the *SOS* Music Recording Technology forum where forum members have gathered information about the various compatible Lunchbox modules — and the list of manufacturers making suitable units is surprisingly expansive. A-Designs, API, Audient, Avedis, Brent Averill, Buzz Audio, Chandler, Daking, Dane, Eisen Audio, Great River, JLM Audio, Lachapell, Old School Audio, Purple Audio, Roll Music, Shadow Hills, Shiny Box, Speck, Vintage designs... and I'm sure many more besides. Several of these companies also make their own alternative module chassis units. For example, A-Designs make a unit that will take two 500-series modules on their sides, to fit a 1U rack space, as well as a little 'mini-console' that holds 16 modules. There are also several different bespoke modular systems around, which are similar in concept but incompatible.

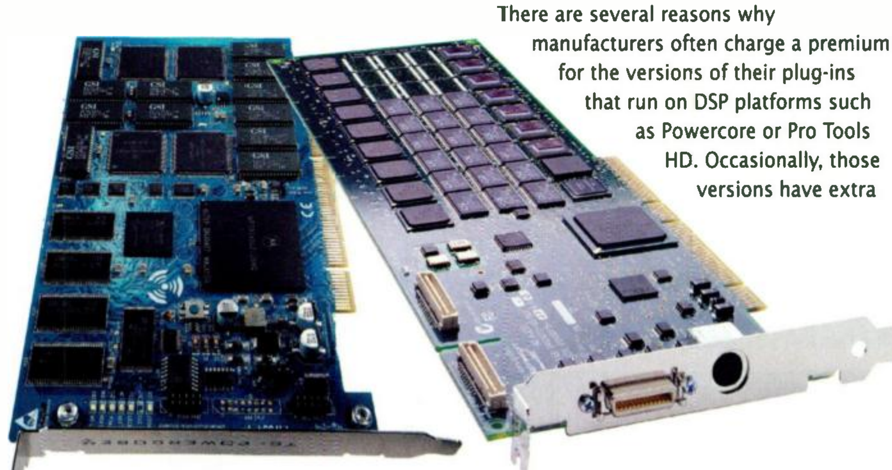


API's 500-6B (above) is the current revision of the Lunchbox, a modular hardware processor system. Third-party modules are available for it, including the A-Designs EM-PEQ, pictured on the right.

Q Is there a difference between native and platform-based plug-ins?

I've noticed a price disparity between the Native and Powercore plug-in offerings from Sonnox. Native versions (Audio Units in my case) are much cheaper than the Powercore versions, but do they differ in quality? If so, are these differences typical of the plug-in market at large? I don't use Pro Tools but have noticed similar price differences between RTAS and TDM plug-ins. Are customers here paying for the privilege of offloading processor burdens to DSP cards, or are the plug-ins better somehow? Is there, perhaps, different algorithmic encoding involved)? I've been running native plug-ins for a while with this on my mind, and I'm hoping you might clear it up for me.
Will Wright

Features Editor Sam Inglis replies: There are several reasons why manufacturers often charge a premium for the versions of their plug-ins that run on DSP platforms such as Powercore or Pro Tools HD. Occasionally, those versions have extra ▶



TC Electronic's Powercore (left) and the Digidesign HD system (right) both use fixed-point processing to run plug-in algorithms, whereas so-called 'native' plug-in platforms, such as Audio Units, VST and RTAS, operate using floating-point mathematics. Plug-ins written for native systems therefore have to be re-coded to be able to operate on specific platforms.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functionality or superior sound quality compared with the native versions, but this is increasingly rare. However, even when the functionality of native and DSP versions is identical, there is a considerable amount of extra work involved in creating the DSP version.

With a native plug-in, the core code (usually in C++) can be re-used across different plug-in formats and even ported from Mac to PC without having to be radically rewritten. Coding for hardware DSP platforms, by contrast, is completely different, and requires that the "same" plug-in be rewritten from a very low level. For instance, native plug-in formats all use floating-point mathematics, but the Motorola DSP chips used in Powercore and Pro Tools HD are fixed-point units. Another major difference is that computer operating systems can allocate RAM and CPU cycles dynamically, whereas DSP-based plug-ins have to declare the amount of processing power and memory they need at load time, and ensure that they never exceed these limits. It is this that, as you suggest, which increases the price of platform-specific plug-ins.

Q How do you connect a mixer to a PC?

Referring to your article in Q&A from *SOS* January 2007, regarding a basic home recording setup, can you connect the Yamaha MG102C, which you suggest, to a PC directly? If so, how?
Via email

News Editor Chris Mayes-Wright replies: To get the audio from a mixer (or any source) directly into a computer, you'll need some kind of audio interface — also known as a soundcard. It's likely that your computer has an 'on-board' soundcard, and yours may have line and/or microphone inputs on mini-jack connections. If this is the case, you can simply plug the outputs of the mixer (in this case either the Stereo Out, Monitor, Rec Out or Aux Send) into the input of your on-board soundcard. You'll obviously need a cable with the relevant terminations. If connecting from the quarter-inch jacks of the MG102C to a mini-jack line input of your PC soundcard, you'll need two mono quarter-inch jacks on one end of the cable and a stereo mini-jack on the other. From here, almost any audio recording package should be able to 'see' the incoming audio.

An alternative would be to purchase an external audio interface, and you can get these from around £50. Not knowing

The output section of a mixer may have a number of different connectors (this Yamaha MG102 only uses jack sockets), so you'll need to have the correct cable to interface with the PC input.

your specific requirements, I can't really recommend one that will best suit your needs, so I suggest visiting our on-line Forum, www.soundonsound.com/forum, where you can post questions.

Q Should I mix on high-end headphones or low-end monitors?

I don't really want to get into a discussion about whether or not you should mix on headphones, but I'm wondering whether I'd be better off with a pair of Sennheiser HD600 (or similar) headphones instead of some low-end monitors? At the moment my 'studio' is just the corner of my bedroom and there's a huge peak around 150Hz, so it's not ideal for mixing. My current monitors are Blueroom Minipods, which are not the best thing for monitoring. So most of the time I mix on Sennheiser HD497s and check later on the monitors.

The problem is that my mixes are not really as good as I would like them to be, so I'm just wondering what the next step should be. (I don't think the missus wants bass traps on top of her bed, by the way...) So, basically, the question is: what monitors should I be looking for in order to have something better than mixing on, say, a pair of HD600s and checking on the Minipods?
***SOS* Forum Post**

Sennheiser HD600s are well respected, and are often used for critical listening. They'll yield better results on a mix than a pair of poorly placed or indifferent-sounding speakers, particularly if your listening environment is compromised. There are certain considerations to bear in mind when mixing on headphones, however.

***SOS* contributor Mike Senior replies:** I'd rather mix on a good pair of headphones than a similar-priced pair of (especially ported) speakers any day of the week, especially in a dodgy-sounding room. The Sennheiser HD600s are a very well-respected choice in this regard, and should seriously outclass your HD497s, in my opinion. However, the way that headphones place phantom central images 'inside' your head, and the lack of crosstalk between left and right channels (as in loudspeaker listening), can make it a bit tricky to judge panning and balance between the instruments. Gauging the low-end response will also be tough, as headphones simply can't generate the same physical sensation as the low frequencies emerging from loudspeakers.

A little temporary acoustic treatment should allow you to use your existing monitors to judge panning and stereo imaging, so working around that aspect of headphone mixing shouldn't be too much of a concern. However, in your position I'd probably invest in an additional mono 'grotbox' (such as the Avantone Mixcube or Pyramid Triple P, perhaps), which should give a more reliable impression of the balance of the instruments than either the headphones or Minipods are likely to provide — it won't sound nice, but that's not the point of it — and will also confirm mono compatibility.

The bass problems are more difficult to work around, however. The HD600s are very good in this respect (especially if you make sure to reference your mix against commercial tracks), and could be supplemented with a software spectrum

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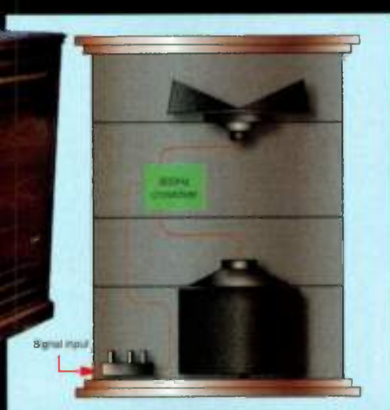
Tel: 020 7692 7628 www.carillondirect.com analyser (such as Roger Nichols' Inspector), but I'd still advise checking the low-end response somewhere else if possible. Even spending lots of money on new monitors won't help you here, if your room sounds rubbish at the low end. If you can't check mixes elsewhere, sources with mainly high-frequency and mid-range content, such as guitars, vocals and drum overheads, can be high-pass filtered. Try, also, to keep a tight grip on the dynamics of remaining low frequencies. For more headphone mixing tips, see the 'Mixing On Headphones' feature in *SOS* January 2007, which includes links to plug-ins that simulate loudspeaker listening on headphones.

A single 'grotbox', such as this Avantone Mixcube, will give a better idea of musical balance than headphones or unsuitable monitors. 'Grotboxes' are useful in all studios, large and small.

SoundAdvice

#10 How do you record a Leslie speaker?

The crack team of Paul White and Hugh Robjohns have travelled the world solving readers' problems. Here, they put aside the Hob Nobs and answer some of your recording queries in our Q&A mini-series, Sound Advice.



Hugh: The Leslie speaker is one of those delightfully old-fashioned mechanical/acoustic devices that works brilliantly, but is actually extremely complex technically! There are several different designs, varying from a single-rotor model, through the 'classic' arrangement of separate bass rotor and treble horn, to more elaborate systems with built-in reverb channels, static channels, and so on. Also, while the 'Leslie' was the original and best, several other manufacturers have had a go at making their own versions, including Karma, Elka, Roland and others.

Paul: In the classic Leslie, the cabinet is divided into three. The middle section houses the two drivers, with the bass speaker facing down into a rotating drum. This has an angled baffle that 'throws' the sound around the room. The high-frequency driver sends the treble up into a rotating horn in the top section of the box. The rotor has two horns, but sound only comes from one — the other is a dummy just to balance the working one. The crossover between the two drivers is at around 800Hz. The horns rotate in the opposite direction to the bass rotor, and they each have a pair of motors, which provide slow 'chorale' and high-speed 'tremolo' modes.

Hugh: The horn and bass rotors are different weights, so they take a different period of time to speed up or slow down, which is an important part of the Leslie effect, and Hammond players often modify the system to change the sound. The baffles at the end of the treble horns are often removed to give a more pronounced effect, and weight is sometimes added to the bass rotor to make it take longer to change speeds!

Paul: The standard miking approach is a single mic in front of the bass rotor and a pair for the treble horns, and usually dynamic mics are used, such as Shure SM57s or Sennheiser MD421s. You can use two mics on the bass rotor for a bigger effect if required, but often the swirling stereo bass can be overpowering and makes cutting vinyl records more of a challenge!

Hugh: The relative positioning of the two horn mics is the most critical aspect, since this determines the stereo width and the depth of the rotary effect — but it is very much a matter of personal taste and what works in the track. The most extreme effect is with the mics on opposite sides of the cabinet, facing each other. A more familiar kind of sound is obtained with the mics angled towards the pivot point from each side of the open rear panel — but, as ever, experimentation is the key.

Paul: If you're in a large room and you need to minimise the amount of reflections, it's best to close mic. Be careful to keep the Leslie away from the wall, at the same time avoiding miking anything at the exact centre of the room, as that seems to be a trouble spot, especially at low frequencies, and more so in square rooms. In a small room where proximity to the walls is unavoidable, hanging acoustic foam or other absorbers on the wall at the level of the tweeters may reduce unwanted coloration, but try both with and without absorbers, as in some rooms the reflected sound may improve the effect.

Hugh: Things to be aware of include the turbulence that the horns create; so be careful not to put the mics too close or you'll get wind noise. The bass rotor is normally covered in a fabric 'skirt' which helps reduce turbulence, but sometimes this has been removed. Simple foam windshields can help if you have to mic very close.

Paul: A relay in the amplifier chassis is used to change the power feeds to the drive motors, and this can produce a surprisingly loud click. So it is sensible to place the bass rotor mic on the opposite side of the cabinet and angle it to minimise picking up the clicks.

A Leslie cabinet (centre) being close-miked at Tori Amos' Martian Engineering studios.

Clavia Nord C1

Digital Combo & Tonewheel Organ

With the demand for (and prices of) old combo and tonewheel organs rising all the time, a keyboard that emulates both at a reasonable cost seems like an excellent idea — but can the Nord C1 cut it when compared to its ancestors?

Gordon Reid

Swedish synth maker Clavia's first stab at a Hammond emulation was featured on the Nord Electro 61 keyboard, which I reviewed in the December 2001 issue of *SOS*. I was very complimentary about its organ mode but, due to some serious errors in the implementation of its electromechanical pianos, I described my time with it as "an emotional rollercoaster". So when I read that Clavia were designing a new keyboard based on the Electro, throwing away the pianos but further refining the organ model, I was rather excited. When I heard that the resulting instrument was to have dual 61-key manuals and include emulations of the Vox Continental and Farfisa Compact Duo (which, perhaps for copyright reasons, Clavia refer to as the Electric-V and Electric-F models) I was *very* excited.

Sound On Sound

Clavia Nord C1 £1595

pros

- Three classic organs in one.
- Light and transportable.
- Its emulations are all but indistinguishable from the originals except in direct A/B comparison.
- The Leslie rotary speaker and Hammond chorus/vibrato simulations are excellent.

cons

- If manipulating physical drawbars is an important part of your performance style... well, you can't.
- The key-click in Hammond mode is far too loud, and needs to be controllable without using the EQ.
- The effects deserve more parameters.
- The manual contains errors.

summary

'Clonewheels' — digital organs that emulate vintage Hammond organs and Leslie speakers — have come of age, and the C1 is an excellent example of the genre. Furthermore, its imitations of the Vox Continental II and Farfisa Compact Duo are far from mere add-ons; they could possibly justify the purchase of a C1 even if the Hammond mode did not exist. If you're in the market for this type of instrument, you cannot ignore the C1.

Physically

The C1 arrived in a padded and wheeled soft case. This is a chargeable option, but it's nice nonetheless. Removing the organ gave me a pleasant shock: it's lighter than many modern workstations. I had prepared a heavy-duty double-X stand for it, but it was apparent that this would not be necessary. The C1 will be a doddle to transport and set up for live use.

The back of the C1 is not complex by modern standards, but you have to watch what you're doing. For example, there are three MIDI sockets, but these are not In/Out/Thru, they're In, Out and a dedicated omni-mode MIDI In for bass pedals, the input from which is channelised and directed to the MIDI Out if required.

Alongside the MIDI sockets, a USB connector is available for upgrading the OS, should it be necessary. Since the review unit had an early version (v1.02), I upgraded it to the latest (v1.12) by downloading the application, connecting my Mac to the C1, running the program, and waiting about 20 seconds for it to do its thing. The procedure was faultless, and PC owners should also have no difficulty, although they will require a driver (available on the company's web site) to enable Windows to 'find new hardware'.

"Would I use one? Certainly, and I believe that few, if any, listeners would be able to tell that I was not using an original Hammond, Vox or Farfisa."

There are three control pedal inputs: a TRS (stereo quarter-inch) socket for a swell pedal, a switch input for controlling the speed of the rotary speaker effect, and a sustain pedal input (which is not as odd as it seems, given that you can use the C1 as a MIDI controller).

Finally, we come to the outputs. These include a headphone output and the main stereo outputs, which can also be used as independent outputs for the Hammond emulation (left) and the Vox and Farfisa emulations (right). When you consider how you might amplify and/or mix these organs, this makes a great deal of sense. Next comes

excellent, particularly for the deeper effects V3 and C3". The same is true on the C1.

The Vox & Farfisa Models

I suspect that the 'V' model in the C1 seeks to emulate the Vox Continental II but, due to its physical configuration, this isn't quite possible; in particular, the Vox was a spinet organ (dual four-octave keyboards) with a different arrangement of controls. But if we forget the ergonomics and concentrate on the sound, we can make useful comparisons.

The Continental II had differing sets of drawbars for the upper and lower manuals: 16', 8', 4', II and III on the upper, and 8', 4', 2' and IV on the lower, with additional drawbars that controlled the contributions of the sine-wave generator and the triangle-wave generator for each. (The II, III and IV settings were different combinations of the higher harmonics.) The C1 gets things almost right, but with the addition of a 16' drawbar on the lower manual. I view this as a bonus, not an error, especially since the octave relationship between the manuals is correctly retained.

Sonically, the sound of the C1 is close to that of the original, but my Vox Continental II is simultaneously brasher and deeper. Hmm... there's something strange going on here. Despite showing the correct legends on the control panel, the C1's manual suggested that its Electric-V waveforms are triangle and square waves rather than the sine and triangle waves of the original, but my ears told me that the waveforms are neither those of the Vox nor those described in the manual.

The Hammond A100

The A100 used for comparisons in this review is identical to the B3 and C3 except that it contains a spring reverb plus internal amplification and speakers. If you defeat the chorus/vibrato, switch off the percussion and turn the reverb to zero, you can listen to the un-affected sound of the tonewheel generator through the static speakers, which makes it ideal for a comparative review, to evaluate the un-affected sounds generated by – say – a Nord C1 versus a real Hammond.

When I checked the waveforms on the oscilloscope, I discovered that the Vox's sine wave isn't a pure sine; it's mildly distorted, and there's a slow modulation that adds movement to the sound. In contrast, the C1's sine wave (which, dear manual writer, is not a triangle!) is close to an ideal sine, so it's no surprise that the Vox has more of an edge. Moving on, the Vox's triangle wave looks a bit like a sine wave with the back edge cut off each half-cycle. The C1's is a more precise version of the same general shape, and certainly not a square wave! Happily, we're dealing with small differences here; while the Vox has a little more character in ▶

a direct comparison, the Electric-V model still sounds very good.

A more significant difference lies in the nature of the distinctive Vox vibrato. The emulation on the C1 is much slower than on my Continental II. For the percussive playing of, say, Two-tone or post-punk New Wave, the Vox vibrato is more desirable, but for sustained chords I prefer the Nord's.

The C1 differs from the Continental II in other regards, too. For example, some Continental IIs (including mine) had percussion; Electric-V does not. Likewise, the bass controls on the Continentals were a switch to select 16' or 16' + 8', two drawbars controlling the contributions from the sine- and triangle-wave generators, and a sustain knob. In contrast, the 'synth bass' model on the C1 (which is automatically selected when you choose the 'V' or 'F' models) has drawbars for the 16' and 8', and produces yet another wave shape. There are also controls for the amount of 'pluck' as well as sustain. To be honest, I prefer the C1's bass sound but — for the purist — it's not the same.

Unfortunately, I was unable to make a comparison between the Electric-F model and an original Farfisa Compact Duo, because I didn't have access to one during the review period. All I can say is that the C1's emulation sounded much as I remember, as did its four modes of vibrato. Subjectively, I liked the F-mode very much, but I can't guarantee its accuracy.

The Effects

In addition to the effects contained within the organ models, there are seven 'outboard' effect processes available in the C1 — delay, overdrive, EQ, amplifier model, chorus, reverb and rotary speaker — but it strikes me that these are in the wrong order. For example, the overdrive would normally lie immediately before the speaker emulation, and the reverb should lie after the rotary speaker. It would be interesting to discuss this with Clavia's engineers, and to find out why they placed them in the order that they did.

Testing the effects themselves, I have to say that I'm not a fan of the delay; it's too pure and too limited in scope. The overdrive is much more satisfying and, although it has just one control, it is superior to many equivalent effects found elsewhere, imparting anything from a warm growl to a full-on howl.

The three-band EQ is basic, with no sweepable frequencies or Q controls to allow you to refine the effect, but it's useable. More so are the amplifier models: L-type (Leslie), F-type (Fender Twin), R-type (which might mean Roland Jazz Chorus, but I wouldn't swear to it), and bypass. These are particularly useful in Electric-V and Electric-F modes, but don't discount them when you're playing in Hammond mode; torturing the output of a B3 through a valve head driven to the point of self-destruction remains a respectable way to get the attention of your audience.

The C1's back panel: note the unusual selection of MIDI ports.

Unison is a stereo chorus with just three settings: medium intensity, high intensity, and off. It's a gentle effect, but I found it much more useful than I had expected. Then there's the reverb, which offers five types but just a single unnamed knob for controlling the level of the effect. Despite the continuing paucity of controls, it's capable of creating some fabulously spacey sounds reminiscent of Pink Floyd and other psychedelic bands of the late 1960s.

Finally, we come to the rotary speaker simulation. This is very pleasing but I am not impressed by the degree of control that you can exert over it. There are just four parameters. The first adjusts the fast and slow speeds of the horn, with just high, normal and low settings that act simultaneously on both. The second controls the horn's acceleration, while the third adjusts the rotor's acceleration, again with high, normal and low settings. The fourth adjusts the fast and slow speeds of the rotor, again with just high, normal and low settings that act simultaneously on both. While the presets are well chosen, it would be nice to have more flexibility and have access to the range of parameters (such as the distance and separation angle of the virtual microphones) that you would find on most modern workstations or multi-effects units.

Put It All Together And...

Make no mistake: despite the lack of real drawbars, I like the C1 very much. It's superbly playable, and the sound is excellent. Choosing the Hammond mode, I experimented with jazz, blues and pop styles, and it responded and sounded just as I wanted. I strayed deep into prog-rock with Deep Purple riffs, Emerson Lake & Palmer and Focus, and then — at the touch of a button — transported myself into soul and gospel territory, and the C1 always delivered.

Having said that, there are problems and, as on the original Electro, it's not the big issues that let the C1 down, it's the details. Indeed, Clavia drive me completely bonkers! ... they get things so *nearly* right, and then let themselves down with avoidable errors. Sure, the problems in the Vintage 2 model are irrelevant because — in my view — Vintage 1 is as 'leaky' as you need for convincing imitations. But the key-click drives me insane. It's almost as if Clavia's engineers had said to themselves, "let's make sure nobody misses the key-click!" If not for this, I would give the C1's Hammond mode an almost unreserved thumbs-up.

My only other area of concern lies with the effects processors, partly because of the order in which they're implemented, and partly because control over them is so limited. If I were a manufacturer and had decided to place the equivalent of a Space Echo, a simple chorus unit, an EQ, a valve amp and a Leslie in the signal path, I would have given players the opportunity to adjust them at least as fully as they could on low-cost outboard effects units.

OK, that's enough moaning, because I want to finish on a high: I think that you could justify the cost of the C1 for its Vox and Farfisa modes alone! The sound of Electric-V is very close to the original, and I'm sure that I wouldn't be able to tell the difference without the opportunity to compare them directly. Similarly, Electric-F makes me believe that I am sitting at and playing a real Farfisa. Given the cost of a tatty Vox Continental II in today's market (perhaps £750) and that of a Compact Duo (at least as much again) and a Leslie 122RV or similar (as much, yet again) the C1 is a much cheaper and more flexible option. That it weighs just 15Kg and can be carried around in something that looks like a big red suitcase is another huge bonus. I like it!

Conclusion

The C1 is a first-class instrument. Its emulations are excellent, and the chorus/vibrato and Leslie effects are as good as I have heard. Would I use one? Certainly, and I believe that few, if any, listeners would be able to tell that I was not using an original Hammond, Vox or Farfisa. But please, Clavia, please let me turn down the key-click. It's driving me mad. SOS

information

£ £1595 including VAT.
T Hand In Hand +44 (0)1752 696633.
E info@handinhand.uk.net
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W www.clavia.se

Photos: Mike Cameron

an 11-pin Leslie connector. I suspect that the C1 would sound stunning through a real rotary speaker, but I have six-pin Leslies in my studio, so this will have to remain speculation. Finally, there's a high-level (14V) output for a Leslie preamplifier. This also uses a quarter-inch socket, so you must be careful not to connect it to standard line-level inputs.

Operationally

Like the Electro, the C1 has no physical drawbars, and uses pairs of buttons to increase or decrease the contribution from each footage. It's a more useable system than it sounds, but if your Hammond performances rely on 'playing' the drawbars, you may find that the C1 is not for you, and that one of the other, more conventional 'clonewheels' will fit the bill better. Nonetheless, the use of buttons and LED readouts has at least two significant advantages. Firstly, the system is flexible enough to mimic the disparate controls of the Hammond, Vox and Farfisa organs that the C1 imitates. Secondly, the LEDs give you immediate feedback about the nature of a Program (or 'preset') when you recall it from memory, which physical drawbars cannot do.

There is memory for 126 Programs that can be saved and recalled in the normal fashion, and dumped and loaded using SysEx. There are also two buttons called Live 1 and Live 2 associated with each manual. These act as non-volatile memories and allow you to recall your favourite registrations without scrolling up or down the patch list.

Inevitably, not all of the C1's parameters and functions are available on dedicated buttons or knobs, so there are three System Settings menus: System, MIDI and Sound. These contain the detailed settings that you are unlikely to want to change during ▶

The C1 uses pairs of buttons to emulate the drawbars found on more traditional organs. This does, of course, make it a good deal easier to save and recall preset sounds.

a performance. Navigation is a little clunky, but simple once you get the hang of it. Happily, the manual is wrong when it says that "System settings are global; they are not stored within a program, but apply to all programs all of the time". This would greatly reduce the flexibility of the C1, and I suspect that the writer intended to refer to the System menu, not the System settings as a whole.

The dual 61-note keyboards are interesting, having two possible responses. 'Fast trigger' mode is not velocity sensitive, but responds as soon as the key is depressed. This is ideal for organ playing, and is always used internally. In addition to this, a velocity-sensitive 'normal trigger' mode is directed only to the MIDI Out. Neither mode generates or receives aftertouch but — depending upon the MIDI mode chosen in the menus — the control panel can send and/or receive MIDI CC data, thus allowing you to record and automate changes in drawbar settings and so on. Unfortunately, the C1's knobs are potentiometers rather than rotary encoders, so if the physical position is different from the value recalled within a memory, the parameter will jump to the physical value when you touch the knob. This can lead to some unpleasant results, especially where the EQ and overdrive knob are concerned.

Before starting the review in earnest, I connected a set of Korg MPK180 MIDI bass pedals, and everything worked swimmingly. This is a big advantage over my favourite clonewheel, the Korg BX3, which has no ability to host a set of pedals. However, had I not had the pedals to hand, I could have used the C1's 'Split' function, which allows you to divide the lower manual into two sections, the upper three octaves responding to the sound chosen for the lower manual and the lower two octaves responding to the sound chosen for the bass. I tested this, and it worked perfectly.

Half-mooning

The review unit came with a 'half-moon' switch for controlling the speed of the rotary speaker effect. This is a chargeable extra, but I imagine that most players will want it, because it is what one expects when playing a real Hammond. The C1's switch can be mounted in a number of positions to the front and left of the lower manual, which is where it should be. Strangely, the manual tells me that, with the switch connected, the rotary speaker controls on the top panel are disabled. This is not correct. If you tell the C1 that it has a 'closed polarity' pedal connected to the rotary control input, the slow and fast options of the half-moon switch work, as do the top panel controls, so you are free to use either.

The Hammond Mode

You select the organ model — Hammond, Electric-V or Electric-F — using a dedicated button that cycles through the three. If you're programming a Hammond sound, your next stop is the Tonewheel model, which resides in the Sound menu. There are three models, called Clean, Vintage 1 and Vintage 2, and these offer increasing levels of artifacts such as leakage and crosstalk between drawbar settings. Of these, my favourite was Vintage 1, which sounded very realistic. Clean was also nice — the 'ideal' Hammond, maybe — but I found Vintage 2 to be unusable. If I ever met a Hammond that sounded like this, I would send it to the doctor. For example, there is a sub-octave leak on D-flat 4 and E-flat 4 that was clearly an electronic fault on the organ used to develop the model. Discovering this was not a question of microscopic investigation: I was playing Pink Floyd's 'Echoes' and there was a low-frequency rumble ruining the delicate registration used for this piece. I double-checked it against my Hammond A100 (see box opposite) and no such artifacts should exist.

Returning to Vintage 1, I then carried out a careful comparison against my A100 and was impressed. The raw sound and pitching across all footages and across the keyboard was excellent, the subtle inconsistencies of a real Hammond were reproduced accurately, the wrap-around at both extremes was correctly recreated, and the gentle compression of the original organ was much as it should be. However, as on the Electro, the key-click and key-bounce artifacts on the C1 are far too loud, and the only way to attenuate them is to turn down the treble EQ. But I don't want a dull sound; I just want less key click! I would like to give Clavia the benefit of the doubt and assume that the C1 conforms to the sound obtained from the Hammond(s) that they analysed to create its models, but somebody within the company should have realised that this level of click is not typical. In my view, correcting it is vital.

Moving on, the C1's percussion sound correctly recreates the tone and response of a Hammond, and there's a bonus, too... The 1' drawbar remains active when percussion is on, which is not the case on vintage Hammonds. This seems trivial, but it isn't, because it allows you to create interesting registrations that are unobtainable on the original.

The 'scanner' chorus/vibrato is the final element in a Hammond emulation, and Clavia's is perhaps the best that I have heard. In 2001, I wrote, "The [*Electro*] vibrato section offers all six options found on the classic Hammonds, and is worthy of a compliment or three. Accurate imitations of the Hammond scanner chorus/vibrato are notoriously difficult to realise, but this one is

Telefunken USA AK47

Multi-pattern Condenser Microphone

Paul White

Will this young gun fire the imagination like the Telefunkens of old?

The Telefunken name that we associate with rare and valuable microphones and preamplifiers dates back to well before World War 1. Telefunken were set up in Berlin as a collaboration between Siemens & Halske and the General Electricity Company and by 1939 the parent company had a staff of over 23,000 workers. In 1941 AEG bought Siemens & Halske, and in 1967 Telefunken were merged with AEG and the company renamed AEG-Telefunken. AEG were bought by Daimler in 1985, and the Telefunken name dropped from the name, though Daimler-Chrysler still use it as a brand within Europe.

Connecticut-based Telefunken USA were incorporated in 2001, after acquiring the rights to the Telefunken name, though founder Toni Fishman had been working on his ideas for a couple of years before that. He apparently started out by looking at the possibility of providing replacement parts for some very rare and valuable vintage mics, including some Telefunken models, but ended up wondering if his company could just build the whole thing from scratch to replicate the original — and clearly he could!

Initially, it was the Telefunken Ela M251 model that took Toni's attention. The original was developed during the 1950s in collaboration with Vienna-based AKG, as a competitor to the Neumann U47, and it used the famous AKG CK12 capsule. This capsule, the 6072 tube and the model T14 transformer were the same key parts as used in the AKG C12, so the models are inevitably similar in some respects, and the Ela M251 actually replaced the C12. Before Telefunken USA could attempt to replicate an Ela M251, they had to reverse-engineer and document the original, because there was no blueprint available. As you might expect, building an exact replica of the original capsule was the greatest challenge, as this particular model has a somewhat complicated design and sells at a premium price.

SOUND ON SOUND

Telefunken USA AK47 £999

pros

- Classy, vintage look.
- Good sound quality that is slightly flattering without being over-coloured.
- Multiple patterns.
- Sensibly priced.

cons

- No obvious cons.

summary

This is a very nice multi-pattern vocal mic, with vintage styling and a high-quality audio transformer that gives a definite boost to the mic's performance.

Outside In

Exact replicas of microphones that are impossibly rare and hugely valuable don't come cheap! For the rest of us, Telefunken USA have developed the RFT AK47 tube mic, which draws on the design ethos of microphones like the U47 but uses newly designed circuitry, based around more readily available parts, including a purpose-built AMI/TAB Funkenwerke B47 audio transformer, wound for them in the US. The one-inch-diameter dual capsule is sourced 'overseas' (which probably means the Far East), and uses a pair of six-micron sputtered gold-on-Mylar diaphragms. The tube is a NOS (New Old Stock) EF732 miniature wire-ended device (also used in some Soundelux microphones) that is soldered directly to the circuit board.

The AK47 has a wonderfully retro look, with black crackle paint on the body and bright plating to the basket and surrounding metalwork. The output transformer is a substantial-looking piece of engineering, and the main circuitry is built from good-quality components hand-soldered to a single-sided, glass-fibre PCB. A small metal U-clamp secures the miniature tube, and the body shell is held in place by means of a bright-plated locking ring at the mic's base.

A seven-pin XLR cable connects the microphone to the power supply, which is

Photo: Mike Cameron

a simple steel box with a three-pin balanced XLR output connector, a nine-position pattern switch (omni, through various widths of cardioid, to figure-of-eight) and a switched IEC mains inlet. The PSU has grey, hammer-finish paint and both mic and PSU bear the distinctive Telefunken logo. Overall, the AK47 measures 46 x 240mm and weighs a substantial 30.5 ounces.

This rather impressive-looking side-address microphone comes in a wooden box with a shockmount, a remote power supply and a Gotham GAC7 cable. A five-year limited warranty comes with the mic, which is aimed at serious project studios, smaller pro studios and the broadcast market.

Like most large-diaphragm studio mics, the AK's frequency response isn't ruler-flat, and as well as a very slight presence peak there's a noticeable mid-range dip at around 300 to 400Hz. At the low end the response rolls off gently below 200Hz, presumably to counter proximity effect in cardioid mode, and at the high end the roll-off starts at 15-16kHz. A frequency response of 20Hz to 20kHz is quoted, but from the included frequency-response graph made for this specific microphone, I'd say the two extremes are around 10dB down relative to the output of the mic between 200Hz and 10kHz. Not that this matters a jot, of course, because what makes mics of this type so attractive is the way they sound, rather than how they look on paper.

One paper specification that does matter is noise, and this mic manages a 76dB S/N ratio and a sensitivity of 14mV/Pa. At 125dB, the maximum SPL is a little lower than for most studio mics, but that's still adequate for all but the loudest sources, and I'd expect this model to be used mainly for vocals or acoustic instruments, rather than kick drums!

Studio Test

I used the AK47 on a long vocal session and it performed faultlessly, capturing a well-balanced sound that managed to be both solid at the low end and smooth at the top. The tube 'character' isn't overdone, and the investment in a decent audio output transformer seems to have paid off, as the high-end detail remains intact without getting gritty. While tube mics tend to be slightly more noisy than their solid-state counterparts, noise isn't an issue in typical studio applications and, used with a standard pop screen, the mic had no tendency to over-react to plosive 'B' and 'P' sounds. I also tried this mic on acoustic guitar, where it put in a good performance (though I wouldn't buy one specifically for that application).

Alternatives

The market for large-diaphragm condensers for vocals is very crowded and there are plenty of alternatives available, such as the Rode Classic MkII and NTK2, the SE Electronics Gemini and M-Audio's Sputnik. As always, there's never a guarantee that a mic, however good, will suit a particular vocalist and if you're working predominantly with one vocalist it's a good idea to audition a wide range of mics to find the one that works best.

As with all large-diaphragm mics, the AK47 isn't going to perform as accurately as a small-diaphragm model in omni mode, but provided the sound being recorded is positioned in front of it, it delivers. I used it mainly in cardioid mode but, even then, having a choice of cardioid widths was very useful.

Whether this mic is worth the asking price is a more subjective question. I've tried less costly tube mics that sound every bit as smooth, but I think you have to work with a mic for a long time to discover its true strengths. Also, you often find that cheaper mics sound great on one voice and quite disappointing on another. The Telefunken name has kudos, and although this model isn't one of their high-end vintage recreations it certainly looks the part. More importantly, it delivers a very credible sound.

Conclusion

Though not exactly bargain basement, the AK47 isn't overpriced for a multi-pattern tube mic, and you get a decent shockmount, so although there are cheaper options this model has its attractions — especially when you consider how smoothly it captures vocals. I certainly enjoyed using the AK47 and very quickly took it for granted, which I'd suggest is the sign of a good microphone: if you have to keep thinking about a mic during a session to coax a good sound out of it, arguably it isn't such a top performer. With the AK47 I got a great sound both really close to the singer and 12-14 inches away. In fact, it seemed that wherever I pointed it I got a positive result, and I needed little or no EQ to arrive at the sound I wanted... which is, of course, what we're all looking for. SOS

information

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Audio Ease Speakerphone

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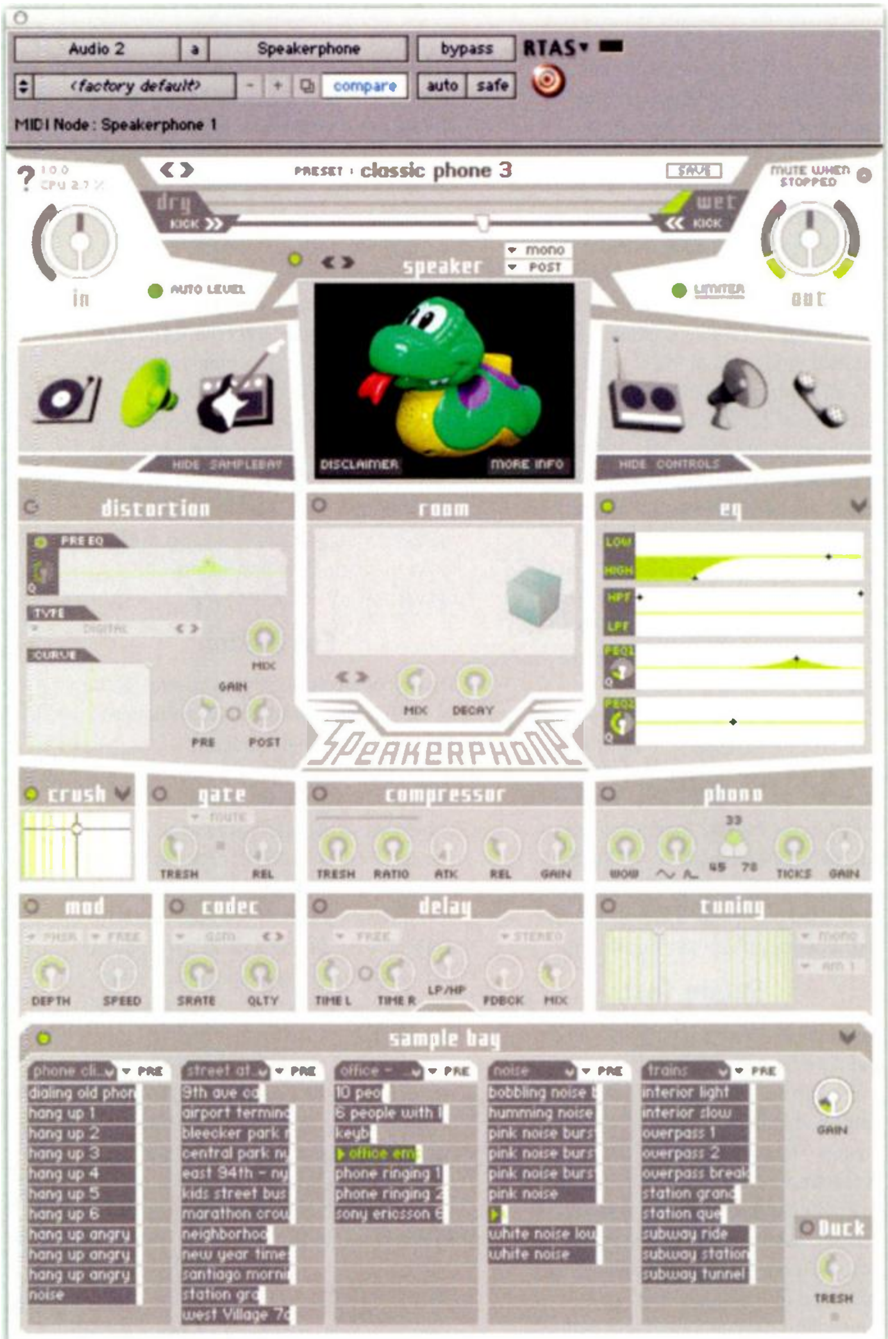
Jem Godfrey

I've always been taken with Audio Ease's rather insanely brilliant Altiverb. Not so much for its cathedrals and emulations of expensive reverb units, although they are stunning, but more because somebody also decided to chuck in the reverb from the back of a Ford Transit van as well, and the inside of a washing machine and a vacuum cleaner pipe. The last two are particularly poignant for me, as I spent a good portion of my childhood years delightedly amplifying my yells and Metal Mickey impersonations through items such as these, much to the irritation of my parents. So it was already with a smile and a warm heart that I took delivery of the latest addition to the Audio Ease family, Speakerphone. It's available in all the major native plug-in formats on both Mac and PC, and can be authorised either via iLok or challenge and response.

For those unfamiliar with the



Audio Ease Speakerphone £347

pros

- Brilliant fun.
- Incredibly versatile and useful for a variety of applications.
- Easy to use.
- Did I mention brilliant fun?

cons

- You have to stop mucking about with it and do some real work every now and then!

summary

Speakerphone is a must-have for post-production professionals, but has plenty of uses in music production as well.

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Classic Tracks

Frankie Goes To Hollywood 'Relax'

The debut single from Liverpool's Frankie Goes To Hollywood was the result of adventurous production and enjoyed massive chart success — as well as creating a great deal of controversy.

Richard Buskin

Artist: Frankie Goes To Hollywood
Track: Relax
Label: ZTT
Released: 1983
Producer: Trevor Horn
Engineers: Steve Lipson, Julian Mendelsohn
Studio: Sarm West

"Relax, don't do it, when you want to suck it to it. Relax, don't do it, when you want to come..." While these words provided ample excuse for BBC Radio and TV to impose a ban on the joyously hypnotic 1983 debut single by Frankie Goes To Hollywood, they also served as a mid-'80s anthem during an era when homo-eroticism became an intrinsic component of the Britpop scene. Thanks to a suitably lewd S&M promo video that, predictably, was also barred from the airwaves, along with a massive marketing campaign that saw kids all over the UK wearing T-shirts with the slogan 'Frankie Says Relax', the band rode a short-lived wave of high-profile controversy. Yet of far longer-lasting impact was the music behind all the hype — a hi-NRG brand of dance-synth-pop that, as crafted by production supremo Trevor Horn, broke new sonic ground, while epitomising '80s excess in all its garish, overblown glory.

Introducing The Band

Having honed his studio skills with Geoff Downes, when they wrote, performed and produced as synth-based band the Buggles (of 'Video Killed The Radio Star' fame), before also replacing Jon Anderson and Rick Wakeman in prog-rock band Yes, Trevor Horn became a full-time producer in 1981 and enjoyed considerable chart success with pop outfits Dollar and ABC. During the next two years he also co-composed several hits with Malcolm McLaren and Anne Dudley, at around the same time that he and wife Jill Sinclair acquired Chris Blackwell's Basing Street Studio complex. Renamed Sarm West, this also housed their new publishing company, Perfect Songs, and the ZTT record label that they founded with NME journalist Paul Morley and producer/engineer Gary Langan.

In May 1983, having seen Frankie Goes to Hollywood perform 'Relax' on Channel Four's *The Tube* music programme, Horn signed the band to ZTT. Fronted by singer Holly Johnson, with Paul Rutherford on vocals and keyboards, Brian Nash on guitar, Mark O'Toole on bass and Peter Gill on drums, Frankie had gone through various line-up changes between their formation in 1980 and the John Peel session they recorded for BBC Radio One in October '82. After accepting an invitation from *The Tube* to perform 'Relax' at the Liverpool State Ballroom in February of the following year, Frankie then included this song in a new BBC Radio session, along with 'Welcome To The Pleasuredome' and 'The Only Star In Heaven', and it was these broadcasts that caught Trevor Horn's attention.

From the outset, Horn focused on 'Relax' as the first single (as well as, interestingly, a cover of Gerry & The Pacemakers' 'Ferry 'Cross The Mersey' which would end up on the B-side of the 12-inch mix). However, initial attempts to record the chant-like 'Relax' with the band members and Ian Dury's backing group, the Blockheads, proved unsatisfactory.

"When I first heard the track, it was a lot funkier than the finished version," says Steve Lipson, whose engineering of the song represented his first collaboration with Trevor Horn. "Trevor's brilliance was to then take it in a different direction and to a whole other level. He really went at it. Frankie were his first signing to ZTT, and he wasn't going to give up until he had a hit."

Learning The Ropes

The producer and/or engineer of artists ranging from Annie Lennox, Grace Jones and Cher to Paul McCartney, Simple Minds, the Pet Shop Boys and Boyzone, Lipson started out as a guitarist and songwriter in a number of different bands around his native London, "nearly getting deals, always blowing it."

It was in 1975 that, fed up with his "terrible" guitar sound, he told a friend named Duncan Bruce that he'd like to become an engineer and learn how to record it himself. Bruce, who recorded jingles, had just purchased a building, and he asked Lipson if he was interested in constructing a studio. Lipson was, and over the course of the next year he put together the Regents Park Recording Company.

"I didn't have a clue what I was doing," Lipson now admits. "I read a few books, talked with different people, bought £15,000 worth of gear, did a bit of building work — with no acoustic treatment — and we opened for business. I obtained a 16-track Unicol tape machine from Command Studios, and the very first session featured a band that had to record a jingle in three hours. I had never recorded anything in my entire life, so I set up however I could, basically imagining what to do, and encountered a problem with the tape machine. When the pinch wheel went in, the tape rode up and down, and half the time it went over the top of the heads. I didn't have a remote, the machine was about 15 feet away from the console, and so I'd go in to 'record', stand by the machine to see if it settled and, if it did, I'd run over to the console, hit the talkback and say, 'Go'. It was a baptism by fire.

"For some bizarre reason, the studio started doing really well, and within six months a band named Sniff 'n' the Tears came in to record an album, *Fickle Heart*, and I ended up co-producing it just because I discovered this amazing thing called the

mute switch. They were really good musicians and they'd play everything right through each song. Well, when I was mixing a track called 'Driver's Seat', I realised that if I muted the guitars while the guy was singing, and then opened the channels, it sounded great. Stuff like that was revelatory to me.

"Another time, when Dave Robinson of Stiff Records asked me to engineer an album by a German singer named Inga Rumpf, he told me, 'I'm getting this guy to come down to start you off.' I was disgusted but there was nothing I could do, and the guy turned out to be Phill Brown, who had been a house engineer at Island Records' Basing Street Studios. He was unbelievable. Phill was only at Regents Park for two days, but his way of doing things floored me completely. The band sounded absolutely great, and he told me this had everything to do with the musicians and nothing to do with him. He'd get them to do things like change the inversion on the piano, or try a different guitar amp, or tune the snare differently, and in the process he showed me that, in a way, the less you do as an engineer, the better it is. You have to be invisible. You should make it look as if you're not doing anything.

"That idea absolutely fascinated me, and from then on I was up and running. I'd got the plot. Because I was a musician, I understood what he was saying, which was don't hold anyone up. Just get them going. I learned that from Paul McCartney as well. A few years later, when he was producing Ringo Starr at a French studio called Superbear, near Nice, Peter Henderson was the engineer and I was assisting him. Well, Ringo saw one of those old Shure mics that Elvis Presley had used early in his career and he told Paul that he'd love to sing into it. Macca said, 'Great, sing into it.' So that's what happened, and ▶

A Banned Band

Ironically the eventual success of the 'Relax' single would not only earn a fortune in royalties for 'Ferry 'Cross The Mersey' composer Gerry Marsden, but in topping the UK charts with their first three singles Frankie Goes To Hollywood would also equal the record set by Marsden's Gerry & the Pacemakers in 1963-64. Not that this looked likely upon the release of 'Relax' in October 1983, which saw the single hover around the lower reaches of the UK Top 50 for a couple of months, before climbing to number 35 at the start of 1984. It was a January 5th performance of the song on *Top Of The Pops* that made all the difference. Within a week, 'Relax' hit number six on the BBC chart, and its success was then sealed by that sure-fire guarantee of a spike in sales: a BBC ban.

On January 11th, expressing his on-air "disgust" at the record's sleeve artwork — depicting a man and woman pressing their bare bums against one another — as well as its printed lyrics, Radio One DJ Mike Read removed the disc from his turntable, and the BBC quickly followed suit. As it happens, the sleeve did contain an "accidental misinterpretation" of Holly Johnson's vocal, converting "Relax, don't do it, when you want to sock it to it," to "when you want to suck it to it." Yet the rest of the lyrics still left little to the imagination, and the instant result was that 'Relax' immediately shot to the top of the UK chart, where it remained for five weeks while *Top Of The Pops* resolutely displayed just a still photo of the group for the climactic 'Number One Spot' along with performances by non-number one artists.

The ban turned out to be an embarrassment for the BBC, not least because the UK's commercial radio stations continued to play the song, and it was deemed fit for broadcast in time for the Christmas Day 1984 edition of *Top Of The Pops*.

Trevor Horn in the control room of Sarm West, 1984. He was apparently on his way to a tennis game. To the right of the picture is the Synclavier that played such a big part in the Frankie Goes To Hollywood sound.

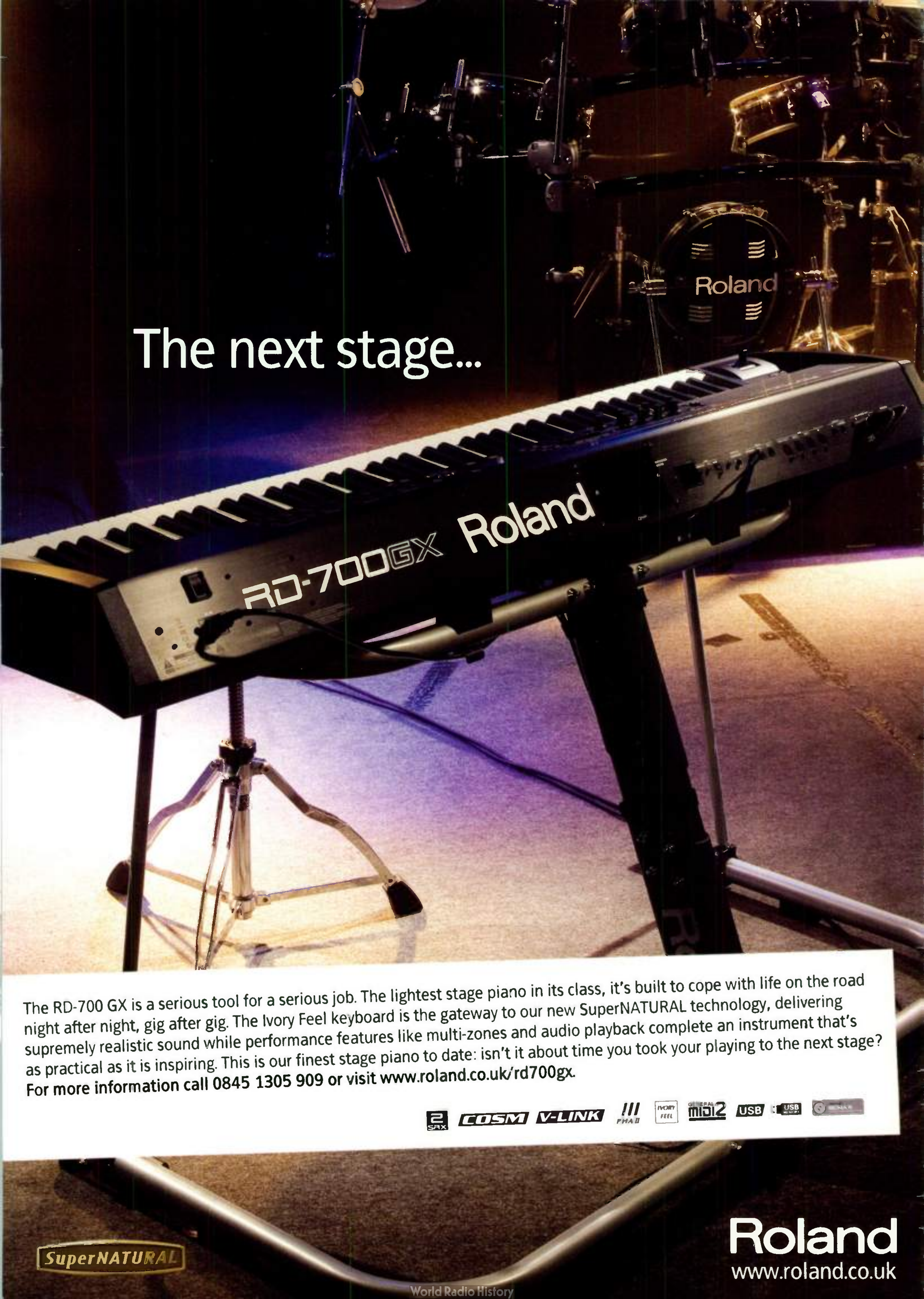
afterwards I asked Macca, 'How come you let him use that? It's a terrible mic.' His answer was that if Ringo wanted to sing into it, he wasn't bothered. So long as he got Ringo into the place where he was happy to be singing, that's all that mattered.

"The job of the engineer, surely, is to facilitate whatever's going on, to make it as easy as possible for whoever you're recording. We're not talking about computers here, we're talking about people, and taking three days to get a drum sound is not conducive to great results. Anyway, what Phill Brown told me was a huge lesson. And by the way, I also discovered that my guitar playing still sounded like crap. Having gone into this whole thing, I realised that the problem lay with me, not anyone else."

Regents Park To Basing Street

Although the Regents Park Recording Company was thriving and Lipson was gaining invaluable on-the-job experience, when Sniff 'n' the Tears asked him to produce their second album in Paris and Duncan Bruce demanded an 80 percent cut of the payment, Lipson walked. It was 1978 and, now freelance, he took engineering work wherever he could get it. This included producing and engineering Lindisfarne's *Sleepless Nights* album at Chipping Norton in Oxford, as well as numerous assignments at both Ridge Farm in Surrey and the aforementioned French studio, Superbear.

"One time, I was at Ridge Farm recording Sally Oldfield, and Herbie Flowers was playing the double bass," Lipson recalls. "I had the microphone where I thought it should go, and as I was walking back towards the control room he pulled it up, nearer his mouth. I therefore went back and repositioned the mic to where I had put it, smiling at him as if I knew what I was doing, and he smiled back. But then as soon as I walked away he moved it back nearer his mouth. I thought, 'You know what, he's Herbie Flowers. I'll just go with it.'



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Steve Lipson in the studio in the mid-'80s.

"When we did the take, he was humming with the bass, and he was humming a bit out of tune. Then he came up to the control room and said, 'Before you play it, put a harmoniser on me.' When I did that, it sounded unbelievable. Again, another huge lesson. And so was a session for Gerry Rafferty's *Snakes & Ladders* album at Air Montserrat in 1980. The band was recording a song called 'Welcome To Hollywood', and Richard Brunton was the guitar player. He and I got along pretty well, and after a take, when everyone was in the control room, he asked me, 'What do you think?'

"Remembering what Phill Brown had told me, I said, 'It's good, but I think you're playing the wrong inversion. You should be up an octave.' Richard said, 'That's a great idea. I'll do that on the next take.' Meanwhile, Gerry Rafferty saw us talking, and so he turned to Richard and asked what this was about. Richard said, 'Oh, Steve just suggested that I try a different inversion.' Gerry Rafferty looked at me, and in a really horrible way he said, 'You just get on with the engineering.' I never forgot that. It was a huge moment. Then and there I realised I should never ignore anyone. Everyone's got an opinion, and when I became a producer I should check them out."

It was in 1983, while producing at a small studio named the Producer's Workshop on London's Fulham Road, that Steve Lipson received a call asking him to spend a couple of days engineering for Trevor Horn. This would be at Sarm West, where an SSL E-Series console was supplemented by a couple of Studer A80 tape machines.

"I knew who he was," Lipson remarks, "but I just didn't want to do it. By then I wanted to be a record producer, and so engineering, for me, seemed like a black hole. At the same time, I also wanted to achieve success on my own terms, not through anybody else, and I was therefore anti the whole idea. Still, the job was for just two days, so I took it even though I had no interest in being there. I basically

The Synclavier Situation

"In those days, the gear we had was pretty limited, and although the Synclavier had just arrived, it was still sitting in the corner of the room and nobody was using it," says Lipson. "JJ had the Fairlight, Andy wasn't interested — he had his equipment — so one day I said to Trevor, 'You bought this thing, do you want me to look at it?' He said, 'Yeah,' and I therefore became the Synclavier operator. At that point, I was the engineer, the Synclavier operator and the guitar player, and it was quite interesting how, reverting back to what Phill Brown had taught me, the engineering was invisible. It was like osmosis. That didn't mean it was easy, but it just happened in a very fluid sort of way. We plugged things in and set them up as we needed to. It was about the concepts and the freedom, about using the available resources, and the result was that all those things sort of engineered themselves. There was no studied engineering going on.

"At the end of what could be considered the first chorus on 'Two Tribes', there's a sort of jazz drum fill that leads back into the second intro, and I programmed that on the Synclavier. Of course, it took an age, and Trevor thought it was the weirdest fill he'd ever heard — a jazz fill in the middle of a pop song. However, I told him I thought it was great, so he went along with it, although begrudgingly, and that was indicative of how we worked together."

"Steve was the first guy I ever saw running a computer all of the time in the control room," Horn told me in 1994. "The Synclavier... incorporated the first timecode-based sequencer, and the significance of that was being able to continually run it like a slave machine. It didn't run on MIDI Song Pointers or anything dumb like that, and so if an idea came to us or we needed an overdub, we could instantly sequence it. I never saw anybody do that before Steve. He was a great guy to work with, because he'd be constantly coming up with ideas."

adopted a couldn't-care-less attitude, but what transpired was that Trevor is the kind of producer who loves the people around him to get on with their jobs. That meant I was inadvertently doing exactly what he wanted me to do — it was weird. And after two days, without either of us saying a thing, we just kept going."

Having already tried to record 'Relax' with Frankie and the Blockheads, as well as with Frankie alone, Horn was now attempting to give the song some fresh impetus, and to that end he'd recruited the engineering talents of Lipson, along with keyboard player Andy Richards and Fairlight programmer JJ Jeczalik. Initially, three weeks were spent on trying to re-fashion the track, while also working on 'Ferry 'Cross the Mersey' and editing The Art Of Noise. Yet it wasn't until he took a dinner break during a 'Ferry' session that Horn came to appreciate Lipson's musical abilities.

"That song didn't have a guitar part yet," Lipson recalls, "so I plugged in my guitar and began playing something for the middle eight. All of a sudden, Trevor ran into the control room and asked whose sound he was hearing. I said, 'Oh, it's me.' He said, 'You never told me you could play the guitar,' and I said, 'Sure, I did, but you didn't appear interested.' Now he was. And this brings us back to 'Relax', which at that time bore no comparison to the finished record, even though the song itself was similar. Trevor had obviously gleaned its essence from the band, but he'd also incorporated some ideas from the Blockheads, along with a few sounds that were in the Fairlight."

These included bass hooks recorded by the Blockheads' Norman Watt-Roy, and a bass pulse sampled on a Fairlight CMI at Battery Studios a couple of years earlier by session musician Mark Cunningham. But Trevor Horn still had an ace up his sleeve.

Rip It Up...

"About three weeks into working on 'Relax', Trevor came in after dinner one night and said, 'We're going to start again.' I couldn't believe it. This 'start again' thing was new to me. I'd always thought that you started and then you finished, and that was the record. Starting again was alien. However, Trevor looked at me and said, 'I've got a rhythm in my Linn 2 that I've had for ages, and I want to work around it."

As Horn himself described it to me in a 1994 interview for *SOS*, "It was like my pet drum pattern which I fiddled about with. ▶

I thought it was more like an English square dance than anything else, but when I saw the effect that this had on the guys in the band, I realised that it was probably going to be a very good dance record."

"He had three patterns and he wanted to do it live," Lipson continues. "He told me, 'The bass needs to be programmed, and I want you to play guitar, Andy to play keyboards, and JJ to get a whole load of shit going on the Fairlight.' That's precisely what happened. Trevor had this rhythm and we just recorded it live, probably in one take. He heard the bass with this drum pattern and went, 'That's unbelievable. Let's go.' The rhythm track is what did it. We flipped out. It didn't exist, and then suddenly it did. The three patterns consisted of one that was the entire thing — hi-hat, bass drum, possibly a snare, and a little conga pattern — plus another without the congas, and then there was the fill: the 16's part where it goes mad. I think this was all done on the fly to a blank pattern, and we put it down with me playing guitar, JJ creating funny noises, Andy playing the chords and Trevor stepping through the presets. Then he called Holly and got him in to sing it there and then, that night, right in the middle of Studio One. I can't remember how many takes, but it was really quick.

"All of the effects on that song were overdubbed. The pissing sound, for instance, was Andy playing his [*Roland*] JP8, as were the explosions. He had a JP8 and [*Roland*] MC4, and he sequenced stuff as well. I myself added some guitar synth with a Roland GR300 — I think it was the second one they ever made — and Paul Rutherford did a whole load of stuff, such as the backing vocals, while JJ programmed all of the weirdness."

The fact that none of the other members of FGTH contributed to the track, along with their unavailability for touring throughout their halcyon year of 1984, would fuel 'Frankie can't play' rumours. And meanwhile, Trevor Horn initially adopted an attitude of 'Stevie can't mix'.

"I did mix 'Relax'," Lipson says, "but because Trevor didn't know me very well he wanted to get someone else in, and so Julian Mendelsohn ended up doing the mix. The same was true for 'Two Tribes', and also for 'War', which was mixed by Nick Ryan, the Sarm chief engineer, who did a mix at Sarm East. For some bizarre reason, he turned the toms up really loud and compressed the mix, and it sounded brilliant. Anyway, I think those are the only tracks that I did with Trevor that I didn't mix. He was just a bit unsure, so it was a comfort-zone thing and I think he was probably right. It really didn't matter to me. We were working on the album, and we just kept going, and there was also 'Ferry 'Cross The Mersey', which we were all really excited about because of the start — we thought the sounds and the whole atmosphere were magnificent."

Steve Lipson: "About three weeks into working on 'Relax', Trevor came in after dinner one night and said, 'We're going to start again.' I couldn't believe it."

Two Tribes

Upon its initial US release, 'Relax' only peaked at number 67 in the spring of 1984. Yet in the UK, where Frankiemania was in full cry, it remained on the charts for 42 consecutive weeks, including a revival in the summer of that year when it climbed back to number two while the band's follow-up single occupied the top spot for nine weeks, having entered the chart at number one. Not that the aforementioned follow-up had been very easy to find.

"Sarm was a busy and expensive studio, so rather than mess around there trying to come up with a new song, Trevor asked me if I knew a cheaper place where we could go with the band," Lipson recalls. "I told him about the Producer's Workshop in Fulham, and so we all decamped for there. Even though he'd already figured out what the next single should be, it was a real long shot. The song was 'Two Tribes', and I remember when we first heard it we all looked at him like he was mad, but he said, 'It's all about the bass line.' He completely got it. He was firing on all cylinders, whereas the rest of us were completely in the dark.

"For one thing, there wasn't much to the song, and for another, the demo wasn't any good. However, out of all the material it was the only track that he could envision being the follow-up single — it wasn't a positive choice, but one made out of necessity. The thing about the bass part on the finished record is


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that it drops an octave, whereas the original bass part didn't do that. It sounded like kids were playing the song, and they were. What Trevor loved about it was the beat, and so after we went down to the Producer's Workshop and the band members did their thing, they then left and again it was down to the four of us — Trevor, Andy, JJ and me — all feeling depressed as anything because it just sounded terrible. We therefore set up and played it ourselves, and interestingly the only part that ended up being retained from the Producer's Workshop was my guitar.

"I played a sort of harmony, and that, together with dropping the bass down on those notes and sequencing it, as well as Andy then coming in with the chord movement — a minor chord to a fourth and back to the minor — were the key elements. Once these were all in place, we then moved back to Sarm to work on the track, and while we were in Studio Two, figuring out how to make the bass sound good, Clive Langer and Alan Winstanley were mixing an album in Studio One, and they then went away to record another album, came back to mix it, and we were still working on the bass. We were looking for sounds, trying to get the articulation right.

"During the recording of 'Relax', having become more familiar with Trevor and his wife Jill, I had suggested that they buy this digital tape machine called a Sony F1. It was a Betamax two-track recorder and it wasn't that expensive, so they went for it, and that was a revelation because we could now record loads of stuff and it was pristine quality. At around the same time, digital multitracks happened, with Sony producing the 3324 and Mitsubishi the 32-track [*X850*]. We were in Studio Two, and I asked Trevor, 'Why don't we look at these machines?' He was always up for trying things, and so a Mitsubishi was wheeled in for us to try out, but it didn't work. A short time later, a Sony was wheeled in and it worked perfectly, so he bought one and we recorded 'Two Tribes' on that machine. In fact, 'Relax' was the only one of the songs on analogue."

In Xanadu...

According to Lipson, Horn panicked at the last moment and embellished 'Two Tribes' with a Linn pattern that would link it to 'Relax'. Then it was on to 'Welcome To The Pleasuredome'.

"That song was quite interesting because I had the idea of getting another tape machine in," says Lipson. "It suddenly occurred to me, if you could make digital copies, you could offset. I had no idea what I was talking about, but I did an offset so that, where the song ended, I had it start again, and when I demonstrated this to Trevor he absolutely flipped out. It was a remarkable thing. The concept of offsets freaked us out. Nobody had ever considered doing anything like this, and all of a sudden we were sort of inventing a recording equivalent of the wheel. It was like a super-sampler."

The producer had a similar recollection: "Steve had copied a multitrack and offset it eight bars, and it was like 'Wow! Let's use this on 'Welcome To The Pleasuredome'!' So he and I were basically upstairs for three months doing that track, which had started out as three minutes long, and we just kept overlapping it on itself, lengthening it and doing all sorts of stuff."

At one point, working in Studio One on the Monday morning after the surrounding Notting Hill carnival had taken place, Horn suggested hanging a microphone out of the window to record the general hubbub of the big clean-up — no one talking, just the sounds of bottles and cans as they were being swept and collected behind the studio along Lancaster Road.

"I literally hung a Neumann U87 out the window, went to the point in the song where the first round ended and the rhythm dropped out, hit 'record' and captured the noise of cleaning up the carnival," Lipson recalls. "It was a remarkable thing. It gave that moment in the song an amazing atmosphere.

"Steve Howe also played some acoustic guitar on 'Pleasuredome', a few funny chords that no one else could get their heads around. That song had so many vast expanses of music where nobody knew what was going on — one time, I was around the other side of the desk in Studio One, nowhere near the console, and I plugged in my old Strat, which was the only guitar I had in those days, and began figuring out a solo. I was telling Trevor how it could work, and he said, 'Show me what you mean.' Unbeknown to me, he went into record, and all the while I was talking to him, saying, 'Here it could go up, and then it could stop.' He said, 'OK, let's hear that back,' and what he played turned out to be the solo."

While it's a common engineering ploy to keep the tape running, it isn't often that you hear about the engineer being recorded without even knowing about this.

"That happened all the time," Lipson says. "It happened a lot on *Slave To The Rhythm*. Trevor was brilliant, but he also let me do whatever I wanted, and I think that's part of what's so great about him."

Talking in 1994, Horn described the aforementioned 1985 Grace Jones album as "probably the last really exciting thing" that he and Lipson did together. "I was acting almost like the artist and he was almost like the producer. I was having all of the mad ideas and he was executing them."

As soon as they began working together they were already embarked on that course. And in the case of 'Welcome To The Pleasuredome' the result was a three-minute

"With 12-inch remixes, I never wanted them to be boring, even though they were nine-or-so minutes long."

Three Times Twelve

In addition to Julian Mendelsohn's 7-inch mix, Trevor Horn also created a trio of 12-inch mixes, as he recalled in our 1994 interview:

"One of the reasons we did all the remixes was that the initial 12-inch version of 'Relax' contained something called 'The Sex Mix', which was 16 minutes long and didn't even contain a song. It was really Holly just jamming, as well as a bunch of samples of the group jumping in the swimming pool and me sort of making disgusting noises by dropping stuff into buckets of water! We got so many complaints about it — particularly from gay clubs, who found it offensive — that we cut it in half and reduced it down to eight minutes, by taking out some of the slightly more offensive parts. Then we got another load of complaints, because the single version wasn't on the 12-inch — I didn't see the point in this at the time, but I was eventually put straight about it.

"When I was out in New York producing Foreigner, I went to Paradise Garage. The Art Of Noise was happening and I'd just done 'Owner Of A Lonely Heart' [*for Yes*], which was huge in America. There was a great remix of it which made number two in the dance chart there, and yet it was only when I went to this club and heard the sort of things they were playing that I really understood about 12-inch remixes. Although I myself had already had a couple of big 12-inch hits, I'd never heard them being played on a large sound system, and so I then went back and mixed 'Relax' again and that was the version which sold a couple of million over here [*in the UK*].

"I wasn't being clever. It wasn't some great scheme that I dreamed up to make three 12-inches; I was just desperately trying to get the record right. I had a kind of ethic with 12-inch remixes. I never wanted them to be boring, even though they were going to be nine-or-so minutes long. I didn't see that as an excuse for them to be self-indulgent or boring, so we would work quite hard on them in order for them to make sense as pieces of music."

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Aphex 207D

Dual-channel Mic Preamp

Can a two-channel preamp at this price really give you the best aspects of tube and solid-state circuitry?

Paul White

Unlike many tube-based mic preamps, the Aphex Model 207D is a hybrid design that uses a solid-state front end followed by a tube gain stage — the idea being to offer the best of both worlds. The solid-state circuitry provides low noise with a good transient response, while the tube provides the 'flavouring'. Of course, every American technical innovation has to be given a trademarked name, and for the 207D the power under the hood is 'Tubessence'!

Overview

This 1U, rackmount, dual-channel preamp uses Aphex's established 'reflected plate' tube technology to give the performance of a tube running on a high voltage while actually running on a low-voltage circuit. Essentially they've taken an under-run tube and put solid-state feedback circuitry around it to counter the increase in plate resistance that otherwise occurs when a tube is run at a very low voltage. This approach saves on cost, extends tube life and also enables the unit to be powered from a switch-mode power supply that can adapt to local line voltages automatically — something that has the happy side-effect of keeping manufacturing costs down. It is also safer, because normal tube HT voltages are high enough to be lethal!

A delayed switch-on circuit mutes the audio path until the tube has warmed up, and to indicate the status the power LED at first glows orange, then turns green when the unit is ready for operation. The 48V phantom power source (which is independent for the two channels) also ramps up when activated, to avoid stressing the microphone circuitry.

In addition to its two channels of hybrid solid-state/tube circuitry, the 207D also features a switchable limiter and an inbuilt 24-bit digital converter with word clock connection that can operate at up to 96kHz, with both AES/EBU and S/PDIF digital outputs. The limiter is useful in situations where you can't accurately predict the maximum peak input level, and there's no setting up required: once it is switched in, the limiter automatically operates to prevent clipping. The circuit is Aphex's own optical design and comes from the much more costly Aphex 1100 two-channel and 1788 eight-channel preamps.

> **SOUND ON SOUND**
>
> **Aphex 207D £360**
>
> *pros*
> • Clean, pristine sound.
> • Switchable limiter.
> • Digital output.
>
> *cons*
> • None really — though the bright, airy sound won't be to everyone's taste.
>
> *summary*
> A great-sounding and very affordable dual-channel microphone preamp and DI that has analogue and digital outputs.

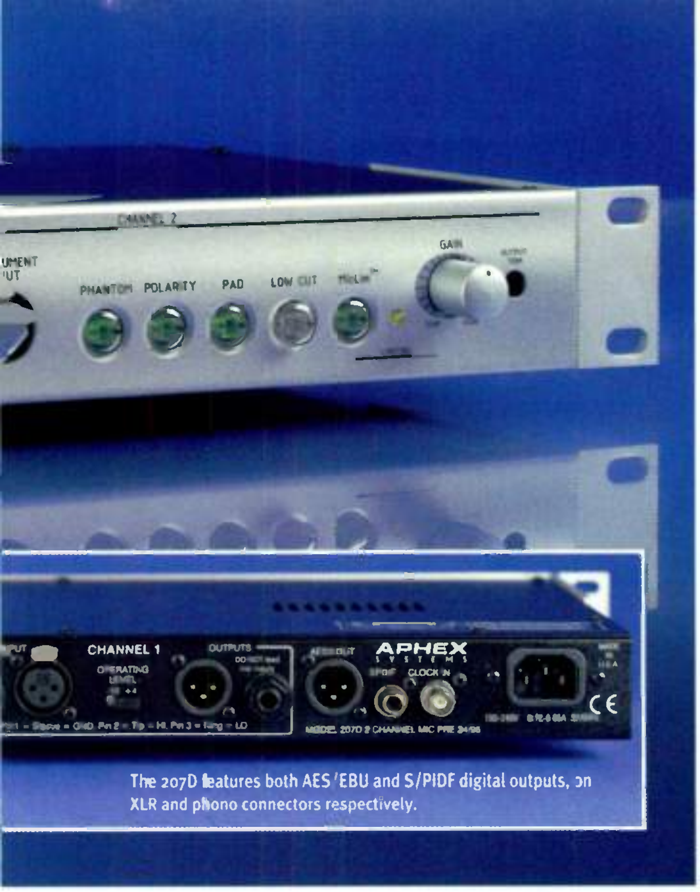
It detects the level at the preamp output, but works on the mic input line to limit the signal prior to preamplification. Because of this approach, it can increase the effective headroom by up to 20dB — though, of course, for the highest audio quality it is best to have a limiter only step in on occasional peaks. I'm not 100 percent sure if this limiter works fast enough to avoid *all* clipping, but it is certainly fast enough that you don't notice any.

The 207D is set out quite differently from the earlier 107. Every channel features a phase-invert switch, a 20dB pad switch, and a low-cut filter, as well as an LED display to indicate available headroom.

Each mic input offers 20 to 65dB of gain, and the high-impedance instrument input also allows up to 65dB of gain to be added. For line-level signals you use the mic inputs with the pad switched in and double-check that the phantom power is switched off! The output operating level can be set (using slide-switches) to -10dBv or +4dBu to suit semi-pro or pro systems, and the output level can be trimmed using the trim pots on the front panel. Power comes into the unit via a standard IEC mains cable.

All the audio connections accept balanced connectors — the outputs are available on both XLR and jacks, the mic inputs are XLRs, and the line and instrument inputs (the latter are, for ease of access, located on the front panel) are jacks. Both mic channels also have an unbalanced insert point on a TRS jack, which allows other processing to be patched post-preamp but pre-converters.

The digital section, which comes as standard on this model (the original 207 is no longer available), comprises both AES/EBU (XLR) and S/PDIF (phono) outputs, as well as



The 207D features both AES/EBU and S/PDIF digital outputs, on XLR and phono connectors respectively.

a word clock input on the usual BNC connector. When the 207D is hooked up digitally, the sample rate can be stepped through using the button between the centrally-mounted bar-graph meters, and the LED indicators display the available choices of 44.1kHz, 48kHz, 88.2kHz, 96kHz or external sync.

Testing Times

Aphex make a number of claims for their circuitry, including greater dynamic range and frequency response than traditional tube circuits, as well as very low noise — so I was keen to put this to the test. As I found with the earlier Aphex 107, when I reviewed it many years ago, the 207D warms things up in a very subtle way. It creates a sense of air around the sound, which makes me think that some subtle high-end enhancement also takes place. The result is a slightly more detailed, crisper sound than you'd expect from a strictly neutral preamp — something that can help the signal survive subsequent processing without becoming dull — but other than that the signal path is very clean and transparent-sounding, and this preamp seemed happy with all of the mics I plugged into it.

Of course, you don't have to use the 207D as a mic preamp: as a simple DI box, it also turns in a classy performance, and here the limiter becomes a very useful ally against more enthusiastic level excursions, especially if you were using the 207D for live recording applications.

Verdict

All in all, this is a very classy-sounding and versatile preamp, that gives you the unmistakably glittery and flattering Aphex sound. It also represents great value at its current price, and there are very few comparable units available for the same budget — see the SSL VHD Pre review on page 146 of this issue for a list of some possible alternatives, though bear in mind that some of these do not offer a digital output as standard. SOS

information

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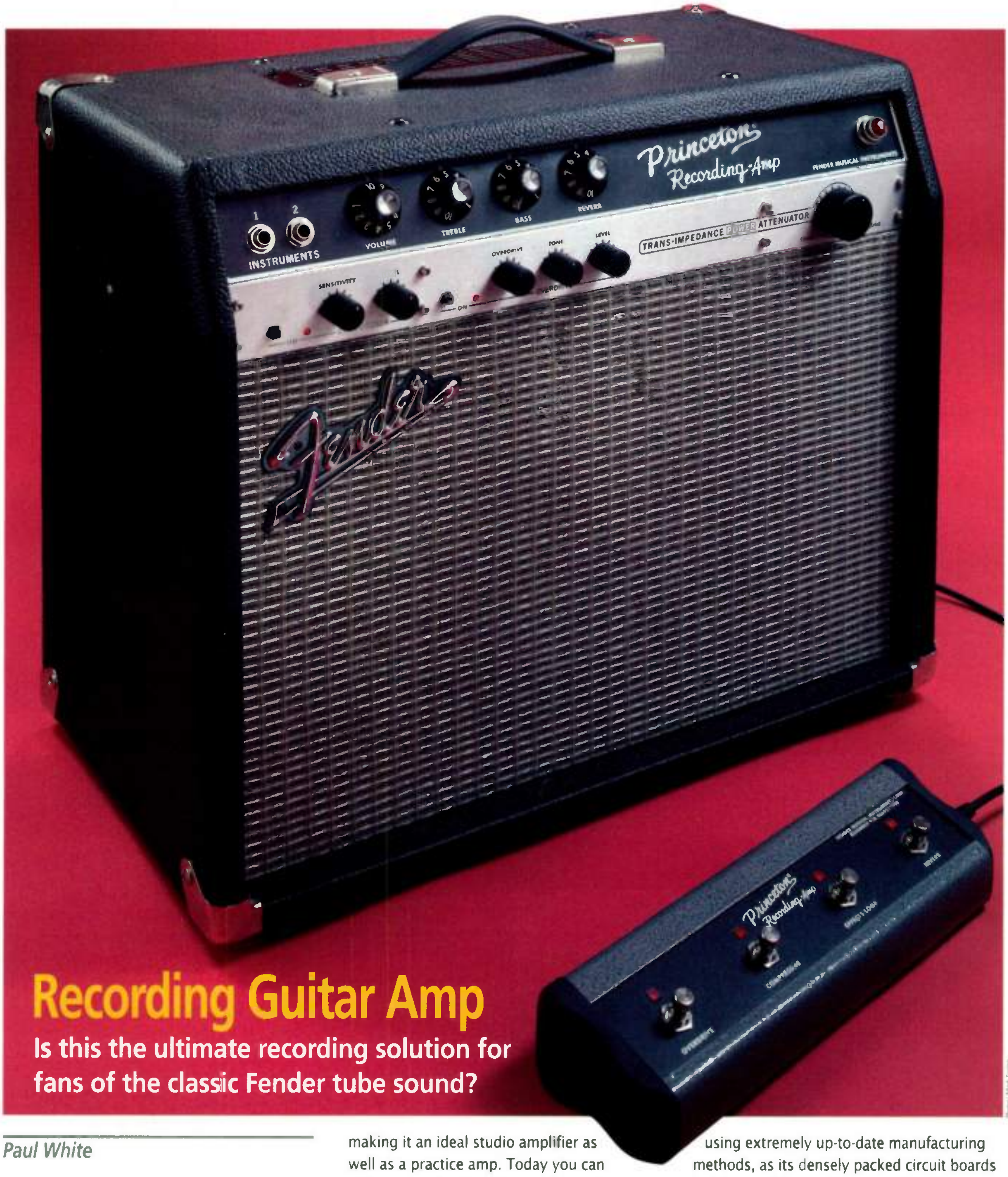
Fender Princeton



Recording Guitar Amp

Is this the ultimate recording solution for fans of the classic Fender tube sound?

Paul White

Fender's original Princeton was built in the 1960s, and although rated at just 15W it became popular for its ability to produce a great tone at moderate sound levels, making it an ideal studio amplifier as well as a practice amp. Today you can pick up a battered original and pay silly money for it — or you could take a look at the new Princeton Recording Amp...

Designed to recreate the tone of the original, this new version is built in Mexico using extremely up-to-date manufacturing methods, as its densely packed circuit boards testify: you'll find neither point-to-point wiring nor generously spaced, hand-assembled PCBs here. Nevertheless, its preamp and power-amp circuits are still based on the all-tube 65 Princeton Reverb model, and they drive a 10-inch Jensen speaker. It can provide a surprising amount of volume, considering the amplifier rating. The preamp stage comprises three 12AX7 tubes and one 12AT7 in the phase-splitter stage, driving a pair of 6V6s in a push-pull configuration.

What really sets this new design apart from its predecessor is the addition of a 1U 'studio' rack just below the main control panel, where you'll find a compressor, a stomp-style overdrive and Fender's 'Trans Impedance Power Attenuator', which lets you wind up the amp to achieve the required amount of output tube overdrive, but enables you to turn down the speaker level as low as you like — in other words, it is essentially a sophisticated power-soak. This circuitry replicates the way a hard-driven speaker reflects energy back to the power stage, which in turn modifies the feel of the guitar, so playing dynamics are not compromised by reducing level. When not required, this circuit may be bypassed completely. Reverb is provided by a tube-driven Accutronics spring (which sounds extremely sweet), and there's also a headphone output and a level-adjustable line output, with built-in speaker emulation for practising, recording or feeding to a PA system. There's a ground-lift switch to cut out ground-loop hum and a speaker output jack, which normally feeds the internal speaker but can be disconnected to allow alternative 8Ω cabinets to be used. This model also includes FX loop send and return jacks, plus a four-switch floor controller that is used to access the overdrive, compressor, reverb, and the FX loop bypass. The footswitches override the corresponding panel controls, and there are status LEDs above each switch.

The cabinet measures 420 x 510 x 280mm, weighing in at a hefty 21kg, and it is made from high-quality plywood, with the traditional Fender vinyl covering and grille cloth. As with many old Fender amps, the open-backed cabinet reveals the spring reverb tucked away in a vinyl 'bag' at the bottom of the cabinet, while a central slat shields the tubes. However, I'd expected a modern design to have a more effective physical barrier between the tubes and hands reaching up from the inside — especially as guitar players have a habit of using the back of the amp for storage and it is possible that the tubes could get damaged in transit by shifting 'cargo'. A perforated metal sheet would have been perfectly adequate.

SOUND ON SOUND

Fender Princeton Recording Amp **£1000**

pros
- Classic Fender all-tube tone.
- Effective Attenuator.
- Speaker-emulated phones and line outputs.
- Compressor and overdrive included, with footswitch control.

cons
- Expensive.
- No gain controls on effects loop or phones output.

summary
The Princeton Recording Amp blends the classic Fender small-amp tone with the convenience of built-in compressor, overdrive and Attenuator, as well as a speaker-simulated line out that can be used for recording.

The Controls

The main control panel is very simple, with controls only for Volume, Treble, Bass and Reverb. The two input jacks offer a choice of ▶

two sensitivities, though they can be used together — in which case the sensitivity of both becomes equal. Input 1 is the most sensitive, and as a rule players will generally plug into this regardless of what pickups they have, on the basis that you can never have too much of a good thing.

While the main control panel is pure pedigree Fender, with its black knobs and red 'jewel' mains lamp, the studio panel below looks a little like an afterthought. I suspect this is deliberate, though, in order to 'distance' the solid-state stomp effects within from the all-tube signal path of the main amplifier. The overdrive is very conventional and tonally fairly neutral, so it can be coaxed into fitting most styles, using its tone control. The other knobs govern the usual drive and output levels. With just two knobs to tweak, the compressor offers control only over sustain and level but actually sounds spot-on, whether you're simply increasing the density of the sound or, at higher settings, adding a pedal-steel-like sustain. Both effects have bypass buttons but they can also be controlled from the floor unit, and when inactive they are completely removed from the signal path.

On a practical note, I feel that the knobs on the studio panel protrude rather too far, and could be vulnerable to damage. They could have been half the height or less without compromising usability. In fact, the Attenuator knob on the review sample had an intermittent fault, even though there was no damage to the packaging and no evident sign of trauma to the front panel.

The Sound

Though an amplifier's sound is highly subjective, I rather like what this little amp has to offer. Using a Fender Strat to test the Princeton, its wiry tonality came over very nicely indeed, sounding bright without being hard or clangy, but still with the characteristic Fender 'pinginess'. At higher volume settings there's enough gain to coax a sweet blues sound out of the Strat's single-coil pickups, and this works particularly well if you turn the volume up full and use the Attenuator to regulate the actual playing level.

Although the overdrive sounds pretty anonymous, in combination with the amp's own natural overdrive it works well enough for beefing up solos. It is effective for dirtying up the sound further for vintage rock sounds, and while it doesn't stray into metal or shred territory, I wouldn't expect it to on an amp like this.

Inevitably there's some background noise audible at higher gain settings, or when the compressor is being used, but this is no worse than for other tube amps and shouldn't be a problem in the studio, as long as you take care with the gain settings and gate high-gain sounds.

I felt that the compressor really improved the feel of clean or just 'on the edge' sounds, and though there are few controls, the attack and release characteristics seemed spot-on, producing the desired result without choking the life out of the sound or making it seem dull, as some compressors tend to do. The speaker-emulated output was surprisingly close to the natural sound of the amplifier, though I'd always opt for recording with a good microphone where possible. It does seem a bit odd to me that the headphone outlet has no level control, though, especially with the current levels of awareness about hearing damage. Similarly, there are no level trims for the effects loop ins and outs.

Lasting Impressions

The Princeton Recording Amp delivers on both Fender tone and studio convenience, but it isn't exactly cheap, especially when you consider that the PVC cover is an optional extra. Cost aside, however, the Princeton does what it sets out to do rather well, with a very nice basic tone, a great spring reverb and a useful stomp-style compressor. And, unlike a modelling or solid-state amplifier, there's also the option of changing to a different brand of tube if you prefer a slightly different sound.

The overdrive is OK, though I'd probably still end up using my Tube Screamer in front of the amp in preference. The Attenuator works particularly well in maintaining the tone and feel of the amplifier at different power levels, so you save yourself the not-insignificant cost of a good power soak/speaker emulator, and the amp is still loud enough for gigs — even though you'd probably need to mic it up in all but the smallest venues.

On balance, I think Fender have managed to strike a good balance between a classic no-frills amp design and a modern studio-ready combo, and it should hold plenty of appeal for the recording guitarist who is in love with the traditional Fender amplifier sound. SOS

Alternatives

There are few, if any, direct alternatives I'm aware of, though there are many hybrid designs that sound good and lend themselves to recording, not least the Vox AD range. There are also numerous good-sounding tube combos that can be used with an additional power soak to achieve similar results in the studio.

The recording output comes as a line-level XLR, and there's a headphone jack for monitoring the signal.

information

£ £1000 including VAT.

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W www.fender.co.uk

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Mix Rescue

We help a home recordist on a budget to add that elusive polish to his jazz-trio recordings.

Mike Senior

Many of the songs sent in to Mix Rescue do indeed need salvaging, but occasionally something comes in that makes a pleasant break from the norm. Jesper Buhl's recording of his jazz trio was one such welcome submission, where he had managed to capture respectable signals from the ensemble performance, despite budget constraints, by careful placement and isolation of the piano, upright bass and drums within his home studio, which is a 7.5 x 7.5m converted garage.

The bass player, Rico De Jeer, was set up towards one corner of the carpeted room, allowing him to be isolated pretty efficiently from the other instruments using three large studio panels (complete with glazed sections to preserve sight lines). The drums and piano were set up more towards the other side of the room, and Jesper had tried to reduce the levels of drum spill on the piano mics by turning the instrument so that its lid opened in the opposite direction, as well as by arranging four 2 x 4-foot absorber panels around it.

The drums were played by Chris Barchet and recorded with a pair of Oktava MK012 cardioid small-diaphragm condensers running through Rane MS1B preamps into an Emu 1820M soundcard, the audio interface for Jesper's Cubase SX2 PC recording system (itself isolated in a purpose-built box to reduce noise). In addition to these mics, the bass drum had an AKG D112 in front of it and a Superlux ECOH6A large-diaphragm electret mic on the batter-head side. There was a Shure SM57 on the snare, as well as a separate small-diaphragm condenser on the hi-hat — although session gremlins ate the hi-hat signal before it could reach the recorder, so this last mic's signal wasn't available for mixing. All of the close mics were amplified by a Behringer ADA8000 preamp/converter before reaching the audio interface.

Another Superlux ECOH6A microphone was set up in front of the bass, after Jesper had tried and rejected both an SE Electronics SE2200A and a Studio Projects B3 in this role, while his Grotriam-Steinweg baby grand piano had two Studio Projects C3 mics (in cardioid mode) in a spaced stereo configuration over the strings. These mics also passed through the ADA8000.

Jesper's original mix was already sensibly balanced — and by no means bad — but I felt that the drums seemed a bit too wide and the bass wasn't quite present enough. There was also a slightly unnatural reverb added to the piano, and it seemed rather 'stuck on'. So I asked him to send over his original multitrack files and loaded them into my own Cubase SX2 system to see if there was a better result to be had.

Here you can see how the instruments were set up for Jesper's recording, with the bass isolated from the drums and piano using part-glazed screens.

Spill, Balance & Phase

Jesper's main concern with his original tracks was the level of spill from the drums on the piano mics, despite the steps he'd taken to reduce it, so I started my listening there. As I'd suspected, it turned out that the spill wasn't too much for this style of music. More to the point, it happened to sound fairly pleasant, albeit with slightly elevated cymbal levels. I felt that the biggest problem was that panning the piano mics evenly placed the drum spill over on the right-hand side of the mix, so I resolved to pan the piano mics a little to the left of the drums to re-centre the spill. Placing the low piano mic 75 percent left and the high piano mic 35 percent right did the trick, while still leaving lots of stereo movement in the piano sound.

I listened to the drum overheads next, which had very low levels of spill from the other instruments but a very wide stereo image, so I reduced the panning to 50 percent left/right. The timbre was pretty good straight away (although with very little bass drum level), so I had a quick listen to the bass mic. This had a fair bit of spill on it from both of the other instruments, but again Jesper had managed to keep this sounding fairly benign. I panned the bass 20 percent to the right, in order to balance the piano image, but without giving a seriously lopsided low-end picture.

Despite Jesper's efforts to isolate the instruments, the bottom line is that spill is an inherent part of recordings like this. With so many mics picking up the same sounds (either directly or as spill), phase-cancellation between the different mic signals becomes an important factor, so it's pointless trying to process any mic in isolation. For this reason, my first real mixing task was simply to fade up the piano, bass and drum-overhead mics to give a rough balance, and then listen to how they interacted with different polarity settings. After a little experimentation, however, I liked the default settings best — another point on Jesper's score-card! — so I turned my attention, first of all, to rounding out the balance of the drums coming through the overheads, by using the close mics.

The bass drum was most obviously missing in action, so I checked out the two relevant close mics. The AKG D112 at the front of the kick presented a rather unappealing and coloured sound (which I'd ▶

Rescued This Month...

First taking up piano lessons at the age of seven, Jesper Buhl joined his first band in his early teens, playing blues and rock & roll. By high school, however, he'd moved more into the jazz field, following the inspiration of such masters of the ivories as Oscar Peterson and Bill Evans. When Miles Davis went electric, he became interested in synths and jazz fusion and moved to Copenhagen to pursue his musical interests. For 10 years he played with his band Blue Turtle, before studying at the Hilversum Conservatory, and subsequently formed another electric jazz fusion combo called Dino On The Loose, as well as becoming involved with the trip-hop-meets-jazz group Warp Expansion Protocol. This month's Mix Rescue song is 'What Is This Thing Called Love?' and was performed by Jesper at the piano, with Rico De Jeer on bass and Chris Barchet on drums.

W www.jesperbuhl.com
W www.ricodejeer.com

The Jesper Buhl Trio (left to right): Jesper Buhl, Rico De Jeer and Chris Barchet.

probably have tried to remedy on the session in Jesper's position), so I faded up the batter-side mic instead. This didn't have the same amount of body to it as the D112, and also included an undesirable low-frequency ringing, but the spill from the rest of the kit was very well behaved indeed, and the resonance was quickly taken care of with an 8dB notch at 100Hz. Again, I checked the polarity of this mic against the rest of the mix, but this time found that an inverted setting gave a more solid sound.

As it happened, the level of snare spill on this mic was so high that I found myself reaching almost instinctively for some processing to reduce it. However, after a moment's thought I realised that this snare spill was, in fact, exactly what the overheads needed to bring the snare forward in the mix, so I stepped away from the plug-in menu with my hands in the air... In the event, this snare spill was something of a stroke of luck, as the snare sound coming through the close mic was a bit dodgy — woolly and resonant, with some nasty spill emphasising the stick noise from the ride cymbal. Although the drum sound was now pretty much in the right ball-park, I tried mixing in the D112 as well to see what it would sound like in context, and discovered that a little bit of it gave a more rounded sound to the kick, once I'd found the best polarity setting.

A spaced pair of Studio Projects C3 mics were used to capture the piano sound, while four acoustic absorber panels were arranged to try to reduce spill from the drums.

Here you can see the parallel compression setting Mike used to improve the sustain of the piano part. Similar settings were used on the other parts, independently.

Parallel Compression

With the majority of the mix balance in place, I could concentrate on some processing. My main aim was to try to increase the sustain and detail of each of the parts, and for this I turned to parallel compression — in other words, using compression as a send effect rather than as an insert. The advantage of the parallel approach is that you can smooth the levels without as much impact on the transients and performance dynamics that are so important to acoustic music styles.

By now I'd bussed each of the parts to its own Group channel in Cubase, and from each of these groups I set up a send to a separate instance of Buzzroom's Gran Comp. Starting with the piano compressor, I left the release time set to automatic but set the attack as fast as possible, effectively ducking the transients and therefore favouring the sustain elements in the compressed signal. With a fairly low ratio of 1.7:1, I adjusted the threshold to give a few decibels of gain reduction, before mixing some of the compressed feed with the uncompressed signal. I then made identical settings on the bass and drum compressors, as a starting point.

I liked the effect I was getting from the three compressors straight away, and while I chose to use more compression than some purists might, the changes were still fairly subtle, bringing up the ambience from the room and improving the audibility of softer details. However, I did make some tweaks to my initial settings, to improve things further. The cymbals were already quite prominent in the mix, so I gave them a little less support from the compressor by cutting 2.5dB from the compressed signal with a high shelf at 2.5kHz. In a similar way, I felt that the piano's low notes seemed to need more sustain than the high notes, so I cut a decibel from the piano's compressor signal with a 1.5kHz high shelf. However, this left the high notes slightly low in level, so I increased the level of the high-strings mic by 0.8dB to compensate.

Conversely, with the bass compressor I cut a decibel at 360Hz, using a low shelf to highlight the upper harmonics a little more. I also increased this compressor's ratio to 2:1 and switched to a release of 65ms, to make the sustain effect more overt. Despite this, the odd note would occasionally poke out unduly, so I bussed the compressed and uncompressed signals together and applied a traditional insert compressor to the resulting Group channel, just to catch these few

A little shelving EQ was used in each of the parallel compressor return channels, to help tailor the effect to each part.



MOTU MachFive 2

Software Sampler

Robin Bigwood

With software samplers playing such a crucial role in many modern styles of music production, competition amongst the major players is hotter than ever. Does MOTU's redesigned MachFive 2 have what it takes?

MOTU's first soft sampler, MachFive, was released in 2004 as the 'Universal Sampler'. It was developed in conjunction with the French company Ultimate Sound Bank and used their 32-bit UVI audio engine. As a virtual instrument that could run on Mac or PC, it was available in all the major plug-in formats, and could load samples and patches in almost any current or legacy format. It also supported multi-channel surround as well as conventional mono and stereo audio, which was an impressive feature at the time. In short, there was a lot to like about MachFive, and it quite rightly attracted a lot of good reviews (not least from *Sound On Sound*) and loyal followers.

The anatomy of MachFive 2. At the top, from left to right, are the File Manager, Display Area, and Master/Part/Layer parameters. In the lower half are the Part List, Keygroup settings (synth parameters) section and FX slots. Various alternative views and translucent overlays are used to provide enhanced functionality.

Four years is a long time in the world of music technology, though. It's still a useful tool, but the original MachFive looks less impressive now, compared with rivals like NI's Kontakt. MachFive 2, then, was eagerly awaited prior to its launch at the end of last year. As you'd expect, it works on Power PC Macs and Intel Macs, as well as Windows PCs, and offers a lot of new features, as well as an extensive sound library. Aside from broad GUI similarities between MachFive and MachFive 2, nearly everything is different or improved, so v2 can be pretty much regarded as a brand-new product.

System Requirements

To run MachFive 2 you need a Mac (Intel or Power PC, G4 1GHz or better if the latter) running OS 10.3.9 or later, or a PC (Pentium 4 1GHz or faster, or AMD equivalent) running Windows XP or 32- or 64-bit Vista. MachFive 2 will run as a plug-in inside any host application that supports MAS, AU, VST, RTAS or DXi formats, but it'll also run as a stand-alone application — a useful consideration for live use, amongst other things. A MachFive 2 Performance or Preset (of which more later) is cross-platform too, so Mac users can share sounds with their PC-based collaborators (and vice versa).

Copy protection relies on a Pace USB iLok dongle. When you buy MachFive 2 from new a pre-authorised iLok is included in the box, whereas upgrade copies get a little snap-off chip that is used to transfer a licence to your existing iLok. It seems people either love or loathe iLoks, so I won't waste time recording my personal feelings about them. However, it would be remiss of me not to report that the experience of using the MachFive's iLok for this review was 100 percent trouble free, and not having to jump through any internet authorisation or activation hoops was a refreshing treat. As they say, one man's treasure...

Sample Replay

For many users, the main purpose of a software sampler will be the playback of sounds from commercial sample libraries. Not surprisingly, MOTU have realised that too, and this is reflected in a range of features in MachFive 2.

Perhaps most important of these is the software's ability to work with a broad range of proprietary formats. These include its software-based rivals (EXS24, Gigasampler/Gigastudio and Kontakt) alongside widespread 'legacy' disc-based formats (Akai S1000/3000, Emu EIII/EIV and the Kurzweil K2 series). For the full list, see the 'File Formats' box. Impressive stuff for sure, but there are one or two notable absences, including Steinberg's Halion and Reason's NNXT. The lack of compatibility with NNXT is a great shame, I think, as there's a lot of good, wide-ranging material in this format, and much of it is often very affordable. NNXT's architecture isn't that complicated either, so I hold out hope that support might be added somewhere along the line. It'll also be interesting to see if and when MOTU add support for more recent versions, like Kontakt 3.

Just Browsing

Actually browsing and loading sounds is achieved in quite an ingenious way. After you double-click either the Preset pop-up menu at top left, or one of the slots in the Parts section below, the entire MachFive 2 window is overlaid with a just-translucent file browser. The left-hand column lists Volumes available to browse (for example, the hard disks in your system, or a legacy optical sample disc you've inserted) along ▶

File Formats

MachFive 2 claims compatibility with a wide range of sample, loop and third-party patch formats:

- **Audio files: AIFF, WAV, SD2, SND (all up to 192kHz, 24-bit).**
- **Loops: Acid, Apple Loops, REX 1, REX 2.**
- **Third-party presets: Akai MPC, Akai S5000/6000, EXS24 (including Garageband instruments), Gigasampler/Gigastudio 1/2/3, Kontakt 1/2, Samplecell, Soundfont, UVI soundbanks (.DAT and .UFS), VSampler 2.**
- **'Legacy' discs: Akai S1000/3000, Emu EIII/EIV, Ensoniq ASR, Kurzweil K2 series, Roland S7 series.**
- **Other MOTU sound libraries: Symphonic Instrument, Ethno.**

These can all be accessed directly, with no pre-conversion stage necessary, and consequently MachFive v1's ugly UVI Xtract application is a thing of the past. On the whole, my success rate at importing third-party sounds was good, and almost every straightforward EXS24, Giga and Kontakt 2 sound I could get my hands on worked great. Little problems can (and did) occur, though, ranging from a few spurious release triggers (which were easily tidied up and re-saved) to more unpredictable results when third-party presets relied extensively on a specific proprietary feature (like Kontakt's scripting) or the synth architecture. I also had a few difficulties with Akai and Emu CD-ROM imports, especially when programs used layered keygroups. MachFive would occasionally place keygroups wrongly, not layering them at all, with many having incorrect root keys, incorrect tuning, or glitchy loops. On the other hand, the majority of sounds imported perfectly.

As MOTU themselves point out, sample importing is a less than exact science, and MachFive 2 fares about as well as its competitors with it. You'll get excellent results alongside a few disasters, and your best chance for success is with fairly straightforward presets from modern sample libraries. For the very best reliability, stick to the bundled library, MOTU's other sample instruments (like Ethno), Universal Sound Bank's UVI soundcards, or other collections natively formatted for MachFive.

MachFive's translucent column-view browser makes navigating complicated file hierarchies quick and easy.

SOUND ON SOUND

MOTU MachFive 2 £299

pros

- Broad compatibility with a range of plug-in and sample formats.
- Excellent sound quality, a fine bundled library and useful effects.
- The ease of use hides deep programmability.

cons

- Poor processor efficiency and polyphony compared to rivals.
- Variable third-party sample import success rate.

summary

MachFive 2 is a hugely capable virtual instrument that can turn its hand to almost any style of music production. If efficiency was better it might just be the perfect software sampler.

with user-configurable Favourites (for frequently accessed files and folders) and disk images. When you click on one of these, the contents are revealed in the browser columns on the right, and by clicking further you can dig down into the hierarchy, with options to Auto Play any loops and to filter file types as you go. You can also right-click on items in the browser and apply certain 'file operations' (such as Delete, Add to Favourites and so on) via a contextual menu. After finding what you want to load, double-clicking it (or, alternatively, clicking the OK button) causes the file browser overlay to disappear, and your sound to be loaded.

On the whole, I found the new browser a great feature. It's quick and intuitive, gives easy access to samples stored in multiple locations, and can be personalised to match your individual needs. It could still be improved, though — a Back (and Forward) button would be great for those times you accidentally clicked out of a deep folder structure you were exploring, and a Search feature would be the icing on the cake. And I do have a gripe, about the way MachFive loads encapsulated formats like Giga presets and Soundfonts.

Let's say that you've downloaded a clutch of Soundfont presets (as I did) from the Internet. You see them in MachFive's browser as individual .SF2 files, but instead of just being able to directly load the single preset they contain, you have to double-click first to mount them in the Images list, and then select the preset from there. Not only does this seem like four mouse clicks too many, but you keep getting sucked out of the folder structure you were in so that you could explore the Soundfonts in the first place. What's more, the Images list soon becomes cluttered with all the presets you've auditioned, and extra effort is required at some point to eject these unwanted images. Please, MOTU, if an .SF2 or .GIG file contains just one or a very few presets, can't we load it directly, as with other formats?

Parts & Mixing

The Part List makes it easy to construct big multitimbral setups, and as there's no limit to how many Parts can be assigned to any individual MIDI channel, huge sound stacks can be created too. MachFive 2 can have an unlimited number of Parts driven by up to 48 MIDI channels (four banks of 16) in the stand-alone version and up to 256 (depending on the host software) in the plug-in. Expert Mode, accessed with a button above the Part list, allows you to set up key ranges and velocity switches for separate parts, allowing multiple parts and presets to be combined into splits and layers. Parts can be added or deleted using dedicated buttons, and although only eight are shown at a time, a scroll bar gives swift access to all that you're using. You also get to transpose and fine-tune Parts, set bend range and choose velocity-response curves and maximum polyphony. All this proves very intuitive and easy to use.

With so many Parts available to use in a single instance, MachFive has an improved mixing and routing scheme. In the Part List there are little volume and pan knobs for quick changes, but to access greater control and an overview of MachFive's internal mix there's a dedicated Mixer view, accessed by a button at the top of the window. This then gives a very familiar overview, complete with level meters and Mute and Solo facilities, and you can still browse and load sounds within this view as well. If I missed anything in this mixing environment, it was a way of grouping faders so that, for example, you could adjust the volume of an entire string section by moving only a violin Part's fader. But then this could also be achieved another way, by running MachFive as a plug-in and assigning all the string parts to an alternative audio output, which in turn fed a single channel in your DAW. MachFive is pretty flexible in this respect — the stand-alone version can drive up to 17 stereo hardware outputs, while Parts in the plug-in can be assigned to multiple hardware outputs or internal buses, depending on what your DAW allows.

The Mixer also hints at the extent of MachFive's effects architecture, and in fact there are five locations where up to four effects can be instantiated: Insert effects can be applied to a Part or even individual keygroups, Preset effects are saved and loaded along with .M5P presets, Part effects belong to individual Part slots, Aux effects can be shared amongst all MachFive Parts, and Master effects act on the main outputs (in both plug-in and stand-alone versions). Phew! All this adds up to a lot of flexibility, but are the effects themselves any good?

I counted 46 different effects types, spanning pretty much every kind of treatment. You call them up using pop-up menus or another translucent browser

The Mixer view replaces the usual main window interface but still gives access to a range of editing features.

overlay, which is very easy, but the selection scheme doesn't allow you to just choose the effect and then start dialling in your settings — you have to select a preset to get the ball rolling. That can feel a bit weird at first, as can working in the effect section and only seeing a row of generic knobs controlling effect parameters. But once you know how, these parameters and additional graphical feedback can be brought up on the central display area, where you also get to click around on an X-Y touchpad-like affair with some effects, but also get cheesy backdrops of guitar pedals and rack gear. It's pretty non-standard stuff, but I quickly got to like it. The effects themselves are surprisingly good, and include decent convolution and computational reverb, a useful range of guitar-oriented fuzz, distortion, amp and cabinet modelling, and effective single and multi-band dynamics processors. As with other aspects of MachFive, there's more breadth and depth than initially meets the eye. As a convenience feature, too, multiple effects chains can be stored as a Multi FX preset, and some are provided ready made.

Loops

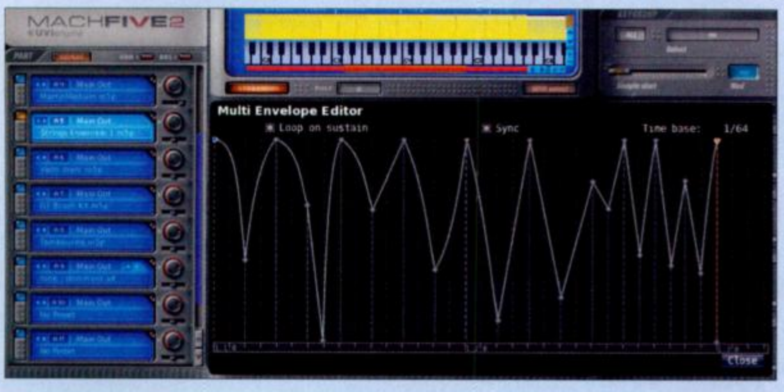
As well as conventional sample replay, MachFive 2 also sports new Loop Lab facilities. The idea here is that instead of loading a preset into a part, you load (or drag and drop) a loop or phrase in a compatible audio format (AIFF, WAV, SD2, REX, RX2, Apple Loops, ACID or UFS loops) and this is then played in response to a MIDI trigger, or even automatically in sync with your sequence. Also, you can work on a loop in Loop Lab and then drag it back to your DAW's sequence editor — and in this case MachFive acts more like an audio editor than anything else. Loop Lab Parts don't have any key-mapping functions, but you get alternative facilities, and there's no limit on the number of 'normal' Parts and Loop Lab Parts in a single instance of MachFive.

With a Loop Lab Part selected, you get some dedicated controls in a little section on the right-hand side of MachFive's window, and you also see your audio's waveform in the central Loop Lab window. But to really get down to business you have to expand the window to access crucial additional parameters, and this step is not initially obvious. Once you're 'in', you quickly discover that ▶

MachFive's effects can be tweaked with knobs in their FX slot, or more extensively in the display area.

The User Interface

Compared to a dozen EXS24s, Gigastudio's multiple windows or Kontakt's busy interface, MachFive 2's single window can appear a haven of calm. But first impressions can be deceptive — the interface can, in fact, change in many ways. First of all there's the file/effects browser, Expert mode and automation/modulation overlays, which temporarily replace the main interface. Then there's the Mixer view, which changes everything except the File Manager at top left. The display area (normally showing keygroup mapping, Loop Lab and sample info) can be enlarged to four times its normal size, popped out into a separate window, or switched to display a spectral analyser, tuner or effects parameters. And last but not least, a pop-up multi-envelope editor can occupy a good part of the lower half of the window.



there are three looping modes on offer: Sample, Stretch and Slice.

Slice mode is a lot like Recycle or Izotope's Phatmatik Pro — it looks for transients in audio (or reads them from an imported file type that already has them) and splits the audio into transient-defined slices. You can then trigger these via MIDI keys (Map mode), or drag them as audio slices or MIDI triggers into your DAW.

Stretch mode does away with any idea of slicing, but instead uses granular synthesis techniques to either change the pitch of the loop (or phrase) while maintaining tempo, or vice versa. I was half expecting to see Ableton Live-like Warp markers, to allow manipulation of the internal rhythmic structure, but Stretch mode doesn't go quite that far.

Finally, Sample mode offers a more traditional style of looping, where duration and pitch are linked. So here's where you can quickly get those grungy down-tempo beats going.

It took me a few minutes to really get my head around what Loop Lab could do, and the best way to interact with it, but once I had, I have to say I was hooked. Perhaps the most powerful features are those that are a little bit hidden. For example, Slice mode's Convert function allows a sliced loop to be dragged into a new Part, where its individual slices get mapped to chromatically arranged keygroups. That lets you really monkey around with it — transposing and applying the synth architecture to individual slices, looping within slices, and combining with other keygroup types. In all three modes, extensive editing and DSP operations on the loaded sample are available via right-clicking in the editor display. Exciting possibilities for sure, and delivered in a very streamlined way.

A Trip To The Library

MachFive 2 ships with a 32GB sound library on four DVDs. DVD 1 consists of a 'universal' soundset that shares a lot of content with Ultimate Sound Bank's Plug Sound Pro. DVD 2 offers various high-quality takes on a German grand piano, and DVD 3 has a series of 'Premium' instruments sampled in surround or at 96 and 192 kHz sample rates. Finally, DVD 4 is a specially licensed sub-set of the the Vienna Symphonic Library, with all orchestral instruments except for solo strings.

Whereas MachFive v1's library was patchy and ultimately disappointing, v2's is the business. DVDs 1 and 4 will probably get the most use, together fulfilling all normal requirements for a range of musical styles. There are really good pianos, basses, drum kits and loops, and they all sound great individually and in a mix.

Sound Design

If your idea of sampling is less about playing back commercial libraries and more about making your own multi-samples and mangling audio from your DAW, MachFive's sample and synth

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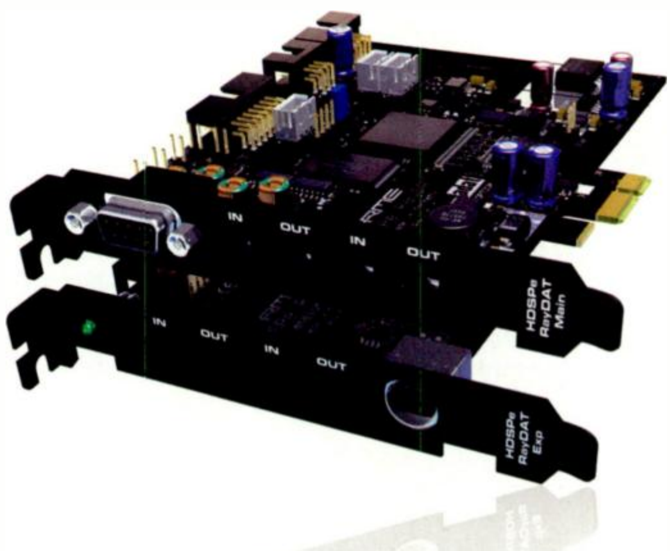

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The hammer falls again. HDSPe RayDAT is the newly developed PCI Express successor of the Hammerfall DSP 9652 which can be regarded as the studio standard for ADAT I/O cards. Some ten thousands of installations worldwide are proof that RME´s ADAT solutions are the perfect partners for all software-based DAWs.

Still based on the award-winning Hammerfall DSP technology, combining superb handling and stability with extremely low latency, the addition of PCI Express and several new features make RayDAT the ideal solution from recording up to the final mastering.

72 Channels. RayDAT offers no less than 4 x ADAT optical I/O, SPDIF I/O and AES/EBU I/O. All 36 inputs and 36 playback channels can be routed and mixed independently, including S/PDIF (phono) and AES/EBU (XLR), which are simultaneously operational due to separated hardware and record/playback devices.

On top, there are 2 MIDI I/Os and TotalMix, RME's unsurpassed DSP-based real-time mixer/router, with hardware-calculated level metering and complete MIDI remote capability. RayDAT also supports the optional TCO Module for synchronization to timecode (LTC/video).

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RME's unique SyncCheck and AutoSync technology has evolved into the new Intelligent Clock Control of the HDSPe system. HDSPe RayDAT measures and displays the frequency of all clock sources - even word clock! Based on validity and current sample rate the system then decides which clock source should be used, fully automated and performed in hardware. With this the HDSPe system offers the most easiest handling of the present clocks, although having a lot digital inputs, plus the most advanced support when configuring the clock setup.

The included DIGICheck for Windows turns the HDSPe RayDAT into a Spectral Analyzer, provides professional level meters for 2, 8, or 36 channels, offers Vector Audio Scope, Global Record Function and various other audio analysis tools.

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architecture will be of interest.

As with 'traditional' samplers dating back a decade or more, MachFive uses a familiar organisational hierarchy. Individual samples are loaded into keygroups, which can be layered and/or limited to particular pitch and velocity ranges. Each keygroup gets its own synth engine (and keygroup FX) and a single setup of samples, keygroups and other settings can be saved and recalled as a Preset. Finally, multiple presets loaded into a number of Parts can be saved and recalled as a MachFive Performance.

Having used MachFive v1 extensively, I found sample management and keygroup editing vastly improved in v2. Individual tools are no longer used in the keygroup editor; instead everything is contextual and automatic, depending on where your mouse pointer is, and right-click contextual menus give more options. In fact, in this respect I have no hesitation in saying that MachFive is the easiest, quickest and most intuitive sampler I've ever worked with. Batch sample placement and editing is flexible and reliable, and loops are easy to create and manipulate.

Interestingly, as well as normal sample-based keygroups MachFive can also create two 'synth' keygroups, 'Raw Oscillators' and 'Organ Emulator', and all three types can be layered and combined at will. A Raw Oscillators keygroup offers up to eight oscillators, each with five different waveforms and other facilities (like PWM and tuning) typical of a synth's oscillator section. Meanwhile, the Organ Emulator is a drawbar-based model, with nine drawbars (individually pannable) and percussion. I found these additional keygroup types extremely useful, and they certainly extend what MachFive is capable of, essentially turning it into a capable synth.

The rest of the synth architecture is, as you'd expect, based around a subtractive model, and it's both powerful and flexible. There are two good-sounding filters offering 14 different filter modes, and their topology can be changed with respect to a Drive circuit and keygroup FX, in 24 possible variations. There are four global and four keygroup LFOs, with variable rise and delay times and multiple waveforms (syncable to host tempo). Any of the six envelope generators can be switched between an AHDSR shape and a multi-segment unlimited breakpoint design. You can also save and load envelope types.

Performance

To get an idea of what sort of polyphony could be expected of MachFive, I ran a few tests on my dual 2GHz G5 and 2GHz Core Duo Macbook (both maxed out with RAM and running OS 10.5.1), using both the stand-alone application and the plug-in hosted in Digital Performer 5.13. In each case I used a 256-sample buffer size and created a DP project that gradually increased MachFive's polyphony by four notes at a time. MachFive had four 32-note polyphonic parts loaded with the 'Violin ens 14 (sustain)' preset from the bundled VSL library. Here's the polyphony (in stereo voices) I achieved before the audio started breaking up:

	Streaming	No streaming	No streaming Filter LP1 enabled
Stand-alone, G5	40	48	32
Plug-in, G5	60	64	48
Plug-in, MacBook	108	120	88

It has to be said, these are not the most impressive figures — especially for the stand-alone application. Obviously the 100 or so voices possible on the MacBook allow pretty complex arrangements to be built up, but by the time you've got other MIDI and audio tracks running and a clutch of plug-ins instantiated, that figure would be lower. As a quick reality check I constructed a similar test running Reason 4 on the G5, again at a 256-sample buffer size, and loading a similar stereo violin sample into multiple NNXTs. The G5 easily achieved over 360 voices before audio broke down — over six times the polyphony of MachFive's best performance, and much more in line with previous *Sound On Sound* G5 benchmarks using EXS24 in Logic. I wasn't able to test MachFive running in either Windows XP or Vista, but I've no reason to suspect the situation there would be radically different.

Loop Lab offers Recycle-like slicing and granular manipulation of audio files. They're powerful features that add hugely to MachFive's capabilities.

Modulation

On the face of it, modulation possibilities look to be fair, but perhaps not particularly impressive. For example, look in the Pitch section and there's a Pitch Mod knob with a pop-up menu giving access to a long list of modulation sources. It looks as if only one modulation source is selectable at any one time — until you click the little '+' button nearby... Then you discover that multiple modulators can work on this one parameter, and a modulation source can just as easily be an external MIDI controller as an internal LFO or envelope. Additionally, nearly all other knobs and sliders can be right-clicked to bring up their own modulation assignment windows — organ drawbars, FX parameter knobs, Aux Send levels, envelope Decay times, or almost anything else you can think of.

Far from being limited, MachFive's modulation and MIDI control possibilities are about as extensive as you could possibly imagine, even offering custom value mapping that allows all sorts of subtle and interesting effects like modifying pitch-bend and mod wheel response. The drawback, of course, is that many assignments are hidden most of the time, and that's where a modulation matrix display (as in MOTU's MX4 synth) wins over. But this is a scheme that makes some very nice things possible, especially in terms of programming expressive instruments that work with MIDI controllers, as well as providing MIDI-based automation with DAWs.

Layer Rules

Keygroup Layers and the rules for switching between them are amongst MachFive's most powerful and least intuitive features, and while they're entirely optional when programming your own sounds, they're essential in providing support for sample libraries that have multiple instrument styles or semi-automated or keyswitched articulations within one preset. The idea is that any one preset can have multiple keygroup layers, themselves containing actual layering or other normal arrangements of keygroups. To give a real world example, this can be used to offer multiple articulations (pizzicato, sustain, marcato and so forth) within one string preset, with some pre-defined MIDI keys used to seamlessly switch between the articulation layers. Other things are possible too — guitar strum samples that alternate up and down strum direction, solo violins that sound different depending on what musical interval was just played, and so on. I had a quick bash at setting up a simple alternating rule with some hi-hat samples and it was easy enough to figure out. Anything much more complex than this I'd prefer to leave to the sample design pros, thank you very much, but still, the features are there, should you need them for your own sounds.

Supersonic?

I'll get straight to the point — I really enjoyed working with MachFive over the course of this review, and I'll gladly make it a part of my normal workflow from now on. Like all the best-designed software, it's easy to learn and immediately useful out of the box, but somehow keeps on delivering as you ask more of it.

Of course, there are some flaws, and right at the top of the list is processor efficiency, or lack of it. To work up a really complex arrangement, with filters and effects enabled, you'll need to have a good computer. MachFive is still useful on older machines, but you must have realistic expectations of what can be achieved. I'd also like to see a few more sample formats supported, the anomalies I experienced with a few legacy discs cleared up, and more native MachFive libraries being released commercially. But set against these drawbacks is the very fine sound quality of the UVI engine and the included library, and the sheer usability of MachFive in most other respects. MOTU have been careful to ensure that it's relevant to a wide range of users too, so whilst Loop Lab might never see the light of day in an orchestrator's studio, it could easily become a crucial feature to a writer or producer of pop or dance music.

MachFive 2 excels wherever good sound quality, broad compatibility and ease of use are high priorities, and the perception that it's less 'geeky' than NI's Kontakt is, I think, a valid one, although immense programming depth is there when you need it. It's a mature product that you can buy confident in the knowledge that — so long as your computer can cope — it'll help you get things done. And we can all use a few more of those. SOS

information

- £ £299 including VAT.
- T Musictrack +44 (0)1767 313447.
- E info@musictrack.co.uk
- W www.musictrack.co.uk
- W www.motu.com

Radial Re-amping Kit

Re-amping is an old technique that allows the producer to keep his or her options open in regard to electric guitar or bass sounds until the last minute. The idea is that you record the dry feed from the guitar, even though the player may be monitoring the sound of an amp. Then, when you come to mix, you feed this dry signal back through the amp of your choice, stick a mic in front of it and add the miked signal back into your mix.

It sounds simple, so why would you need to buy hardware to achieve it? Well, if you already have a suitable high-impedance DI box, you can have a stab at it without using more hardware, but the results may be less than optimal for a number of reasons — not least that the output of an audio interface or soundcard doesn't 'look' the same as the output from a guitar pickup as far as a guitar amp is concerned. It's all a matter of signal levels and source impedance.

Radial take their usual thorough approach by using two pieces of hardware to solve the problem — a high-quality DI box for recording and a specially developed amp driver for feeding the recording back into the amplifier. In fact, there are several Radial DI boxes suitable for recording, specifically the JDI, J48 (reviewed in *SOS* October 2007) and JDV models, as all that is needed is a high-impedance input, a link through to the amp being used for monitoring and a balanced line-level output for connecting to the recording system.

Radial's X-Amp amp driver looks after the other half of the process and is available on its own (£147 including VAT) or as part of a re-amping kit (£294), which also includes a J48 (£147) and a carry-case. Built with a similar form factor to the DI boxes, the X-Amp runs from an included power adaptor. The power inlet and balanced XLR input are on the rear panel, along with a ground-lift switch that acts on the input side. The construction of the box, which is made from heavy, folded steel, affords protection to the connectors by means of a 'book-style' overhang, and a non-slip rubber mat on the underside helps keep it from sliding around.

The output end is designed to face the user and includes a power-on LED, a clip LED and a screwdriver-adjustable (though fingertips also work...) output level control. There are two outputs, both on unbalanced jacks, the first labelled 'Direct'. This is a transformerless feed and should always be used, as it provides the ground connection to the amplifier. The second output is transformer-isolated, allowing a second amp to be fed without the risk of ground-loop hum, and this has a phase-invert switch, which would normally be used if one guitar amp happened to be wired with its speaker

in the opposite phase to the other. It can also be used to create deliberately phasey sounds, which may be creatively valid if the two amps sound quite different or are miked from different distances.

Up & Running

The normal way to use this system is to place the DI box in line with the guitar cable, between the guitar and the amplifier, so that the guitarist can play normally and the clean signal from the guitar can be recorded regardless of the amplifier settings. The output from the recorded guitar track is then fed to the X-Amp via an XLR cable of any practical length, after which you adjust the output level from the recording system until the clip LED starts to blink on peaks. Backing off the level by a few dB should stop the clip LED coming on and still give you plenty of level. You can now feed Output 1 into your guitar amplifier and then adjust the output level control on the X-Amp to get the best level into the amplifier. A practical way to do this is to try the guitar directly into the amplifier with a fairly clean sound, then adjust the X-Amp to give the same subjective level. If there's any hum, the ground-lift switch, which isolates the input signal ground, can be used. The manual warns that any connected amplifiers should be properly grounded (which, for safety reasons, they should be anyway).

Once you're set up, the recorded signal fed back through the amplifier behaves just as it would if you were playing the guitar through it directly — except, of course, that the output from the amplifier doesn't interact with the guitar, as it can do at high volumes during performance. If you want the extra sustain that being close to feedback gives you, simply play the amp at a high level when making the initial recording. The X-Amp can also be fed through any chain of effects pedals, again just as if it were a guitar, so it is pretty flexible.

As with all the Radial products we've tried, this system works exceptionally well, and while it may not be the cheapest solution, it won't compromise your sound and should give a lifetime of service. The re-amped sound is essentially indistinguishable from the sound you get when plugging the guitar directly into the amplifier, though you may get very minor differences depending on the type of guitar cable you use with the amplifier. The only extra thing I would have liked is an input level control, as most DAWs have no physical output level control, which means that levels have to be adjusted in software. This reduces the signal resolution slightly — although, of course, if you're working at 24-bit, as most people now are, that won't be an issue in practice. In all, this is a beautifully engineered re-amping package. *Paul White*

SUMMARY

The Radial re-amping kit provides a very elegant and effective way of re-amping your guitar and bass signals. If you already have a good DI box, the X-Amp would be a great addition.

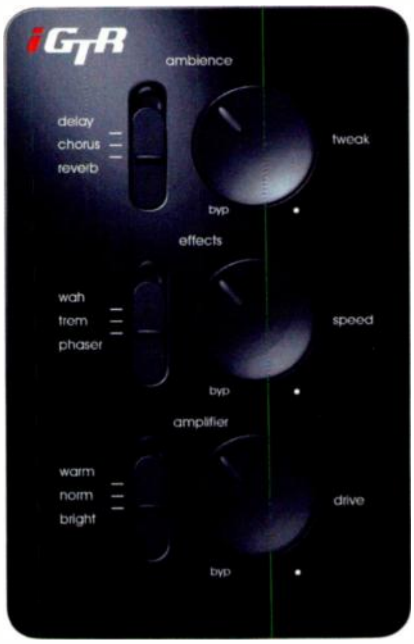
Shure Distribution UK +44 (0)1992 703058.
www.shuredistribution.co.uk
www.radialeng.com

Waves iGTR

Personal Guitar Processor

Waves' iGTR Personal Guitar Processor is said to be a spin-off from their GTR software and interface. This pocket-sized hardware box uses digital processing to provide high quality basic effects, clean and dirty guitar sounds, and has a stereo mini-jack input for your MP3 player so you can jam along on the tour bus or in your hotel room.

Around the size of an original Apple iPod, and with a non-slip casing, the iGTR is powered by four AAA batteries or an optional power adaptor, and it also comes with a detachable belt-clip. Its simple controls are divided into three sections, each with a rotary control and a three-way slide switch. Thumbwheel controls on the edge of the box adjust the guitar level and aux input level, and there are two headphone mini-jacks — so you can inflict your playing on a third party! A slide-switch on top of the iGTR is used to power it up, and there's the usual quarter inch input jack for your guitar cable.



The top section offers a choice of Delay, Chorus or Reverb, for which a single knob adjusts either the amount or speed of the effect, as appropriate, with the fully anti-clockwise position bypassing the effect. Next there's a choice of wah (auto), tremolo or phaser, again with a single-knob adjustment (in auto wah mode this sets sensitivity) and bypass, all of which sound absolutely fine.

Finally, we get to the amp models which are simply designated Warm, Normal and Bright, with the rotary control adjusting the drive level. Waves claim that the iGTR offers realistic sounds that are inspired by vintage models, and the clean sounds are very nice, with distinctive tonal characters. However, turn up the drive and it sounds as though the designer has omitted the speaker-simulation stage — the sound morphs gracefully from sparkly clean to 'wasp-in-a-paper-cup', as though it is simply being clipped. The fixed level of background hiss from the output stage isn't affected by the guitar volume control: using a sensitive set of Ultimate Ears ear-buds, the background hiss level was always very obvious, whereas with less sensitive phones the hiss only becomes really noticeable at drive levels of more than 12 o'clock, at which point the noise from the overdrive effect starts to build up.

The practical outcome is that to keep the hiss down you need to use less sensitive headphones and crank the guitar level up — which will reward you with some nice-sounding effects and some very usable clean tones in the 'glassy' Rockman style. But the overdrive sounds are simply too fizzy to be pleasant — even allowing for the fact that US guitar sounds usually have more edge than European ones. You may be able to record passable overdrive guitar sounds by feeding the output via a software speaker-emulator, which would also cut down on the hiss, but as things stand anything above a very mild overdrive sounds quite dreadful, especially when you consider that for only a little more than the iGTR's £69 retail price you could buy something like a Line 6 Pocket Pod, which is superior in all respects other than physical size. As a convenient practice aid to throw into your guitar case, the iGTR has a definite appeal if you play mostly clean sounds, but the gritty overdrive lets it down badly. *Paul White*

SUMMARY

The main benefits of the iGTR are size and simplicity. The clean sounds are good too, but the overdrive sounds are a real let-down, as is the constant background hiss when using sensitive earphones. Though it's cheaper than its rivals, an extra £20 or £30 can buy you something better.

Sonic Distribution +44 (0)1582 470260.
www.sonic-distribution.com
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The New Atlantis

The Story Of The BBC Radiophonic Workshop

Fifty years ago this month, the most celebrated electronic music studio in the world was established. We trace the history of the Radiophonic Workshop, talking to the composers and technical staff who helped to create its unique body of work.

Steve Marshall

I was 10 years old. As the last 'whoosh' of the *Doctor Who* theme dissolved into a wash of tape echo I sat transfixed by the light of the television, eagerly reading all of the end credits. "Wow!" I exclaimed. "I want to get a job in the BBC Radiophonic Workshop when I grow up!"

"I'm sorry, son," said my father. "You won't be able to do both."

Although it never felt like a 'job', I did eventually get to work in the Radiophonic Workshop. I was only there for three months, but I've never stopped going on about it. Wouldn't you too, if you'd been lucky enough to have worked in the most famous electronic music studio in history?

The story of the Radiophonic Workshop began half a century ago, in 1958. Britain in the 1950s was a bleak place, as the nation struggled to rebuild itself after the devastation of war. Food rationing had continued right up until 1954, when bananas finally came back on sale; anything worth having was still in short supply. We now think of the '50s as the rock & roll years, but the UK charts for 1958 tell quite a different story. Elvis was there for a few weeks; so was Jerry Lee Lewis — but

Delia Derbyshire, with Workshop co-founder Desmond Briscoe in 1965.

the chart is mostly dominated by the likes of Perry Como, Connie Francis and Vick Damone. It was a dull time for music, but things were about to get more interesting...

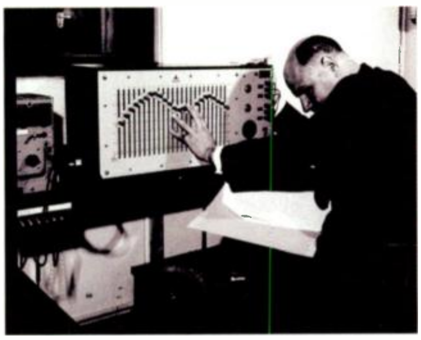
Defects Of The Brain

One of the few benefits of wartime had been that some women had an opportunity to work in jobs previously denied to them; Daphne Oram was one. Daphne had started working for the BBC as a 'music balancer' during the war, turning down a place at the Royal College of Music to do so. After her promotion to studio manager in the '50s, she began pestering the BBC to follow the lead of the French broadcasters, and to provide a facility for the production of electronic sound

Before the Workshop: Daphne Oram manipulates a tape loop at Broadcasting House, watched by Frederick Bradnum, 1956 or '57.

and *musique concrète*. Desmond Briscoe (1925-2006) was also a studio manager, with similar interests, so in 1957 the pair teamed up to produce some innovative programmes for the BBC Drama Department. Using borrowed test oscillators and tape-splicing techniques, they produced sounds that had never been heard before on the BBC.

Their nagging finally paid off, and in April 1958 Desmond and Daphne founded the Radiophonic Workshop in the BBC's Maida Vale Studios (a former ice-skating rink). They were joined later in the year by 'technical assistant' Dick Mills. Brian Hodgson came along in



Desmond Briscoe at work, 1960.

1962 and he eventually ended up running the place. Brian adds: "Workshop was then a very popular word among theatre 'types', and it gave away the Drama Department origins. It was originally going to be called the Electrophonic Workshop, but it was discovered that 'electrophonic' referred to some sort of defect of the brain, so it had to be changed! A board was set up to see that the place was run properly. Unfortunately, one board member had a doctor friend, who advised that three months should be the maximum length of time that anyone could work there, as staying any longer could be injurious to their health; they'd go mad, or something. This problem recurred throughout the Workshop's history — just as a recruit was getting into the swing of things, they'd have to leave."

Daphne Oram was the first to fall foul of this rule. After three months in her new job, she was ordered back to work in a control room at Broadcasting House. But for some reason Desmond Briscoe was not required to leave: instead he was appointed as the

The New Atlantis

"Wee have also Sound-Houses" became the Radiophonic Workshop's motto. Taken from *The New Atlantis* by Francis Bacon, it was rediscovered by Daphne Oram, and for many years was pinned to the Workshop's office wall. It is an extraordinary piece of writing, seemingly a vision of some recording studio of the future; yet, incredibly, it was written in 1624.

"Wee have also Sound-Houses, wher wee practise and demonstrate all Sounds, and their Generation. Wee have Harmonies which you have not, of Quarter-Sounds and lesser Slides of Sounds. Diverse Instruments of Musick likewise to you unknowne, some sweeter than any you have; Together with Bells and Rings that are dainty and sweet. Wee represent Small Sounds as Great and Deepe; Likewise Great Sounds, Extenuate and Sharpe; Wee make diverse Tremblings and Warblings of Sounds, which in their Originall are Entire. Wee represent and imitate all Articulate Sounds and Letters, and the Voices and Notes of Beasts and Birds. Wee have certaine Helps, which sett to the Eare doe further the Hearing greatly. Wee have also diverse Strange and Artificiall Eccho's, Reflecting the Voice many times, and as it were Tossing it; And some that give back the Voice Lowder then it came, some Shriller, and some Deeper; Yea, some rendring the Voice, Differing in the Letters or Articulate Sound, from that they receyve. Wee have also meanes to convey Sounds in Trunks and Pipes, in strange Lines, and Distances."

John Baker was another stalwart Radiophonic Workshop composer.

Workshop's Senior Studio Manager. For the BBC's women, it seemed, the war was over. A lengthy and bitter row ensued, and eventually, Daphne left the BBC for good in 1959, moving to an oast-house that she'd bought in Kent and establishing her own Oramics Studios for Electronic Composition. She was replaced by Maddalena Fagandini.

Fag-ends & Lollipops

The Workshop's reputation grew over the next few years, and the ranks swelled with the addition of Brian Hodgson, Delia Derbyshire and jazz pianist John Baker. The equipment at their disposal was minimal, to say the least, as Brian recalls. "In the very beginning, Desmond had been given £2000 and the key to 'redundant plant' [*the BBC's junk pile*] and that was it! The place kept going for years on what we called 'fag-ends and lollipops'. 'Fag-ends' were the bits of unwanted rubbish that other departments had thrown away; 'lollipops' were the much rarer treats that were occasionally sent down to keep Desmond quiet. Like the vocoder, for instance: it was very nice, but we hadn't asked for one and didn't really need it. It was like the icing on a non-existent cake!"

The Workshop's equipment consisted merely of a lot of old tape recorders and a few pieces of test equipment that could make noises. The tape recorders could be used for echo, and reverb was also available — it came from an empty room downstairs with a microphone at one end and a speaker at the other. Maida Vale Studios is an unusual building, long and thin with one of its two floors below ground. The Radiophonic Workshop's rooms were at street level, spanning an extremely long corridor.

One room was occupied by a succession of dedicated engineers who had the tools and the know-how to fix all the broken rubbish that arrived; they also built special equipment to order. First was 'Dickie' Bird; then came Dave Young, and finally 'The Two Rays' (White and Riley). Dave Young started a tradition of visiting the nearby Portobello Market every week to buy bits and pieces for the Workshop, and this continued long after he'd left. In the '60s, a lot of ex-military kit from the war was still being sold off; Dave would return with items such as a genuine aircraft's joystick!

Brian Hodgson plays a tune on the Workshop's home-made keyboard, controlling 12 individual oscillators.

Much of the Workshop's output then was produced simply by using the techniques of *musique concrète*: natural sounds were recorded and manipulated on tape by editing, pitch-changing, and very often by reversing the tape. There was a standing joke that a Radiophonic composer could enthusiastically churn out original compositions for several years. When the inspiration ran out, all their old tracks could be re-used (and improved?) by playing them backwards!

Wobbulating The World

In the early '60s, synthesizers simply did not exist. Producer Joe Meek was using the monophonic, valve-operated Clavioline but the Radiophonic Workshop, oddly enough, never had one. What they did have, though, was all the test oscillators that they could beg, borrow or steal from other BBC departments. A method was devised for controlling 12 oscillators at a time, triggering them from a tiny home-built keyboard of recycled piano keys. Each oscillator could be independently tuned by means of a range switch and a chunky Bakelite frequency knob.

There was also the versatile 'wobbulator', a sine-wave oscillator that could be frequency modulated. It consisted of a very large metal box, with a few switches and one very large knob in the middle that could sweep the entire frequency range in one revolution. They were used in the BBC for 'calibrating reverb times in studios' apparently. And as far as the Workshop's electronic sound sources went, that was it!

Yet, curiously, it is the work produced in those early years that the Radiophonic Workshop's reputation still hangs on. The *Doctor Who* theme was first recorded in 1963, and still there are fans who insist that the original is the best of many versions made over the years. What's more, some of the sound effects made for the first series of *Doctor Who* are still being used! When the newly revamped *Doctor Who* appeared in 2005, hardcore fans recognised the original effects and wrote to Brian Hodgson: "How nice to hear the old original Dalek Control Room

The Radiophonic Workshop name would become indelibly associated with a certain long-running science-fiction TV series...

Brian Hodgson with dismembered piano, as used to create 'the Tardis sound' from *Doctor Who.*

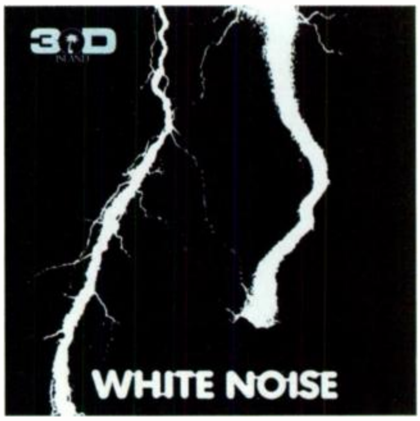
again, after all these years!"

Brian's 'Tardis' sound, dating from 1963, is also still used. "I spent a long time in planning the Tardis sound," says Brian. "I wanted a sound that seemed to be travelling in two directions at once; coming and going at the same time." The sound was actually made from the bare strings of a piano that had been dismantled. Brian scraped along some bass strings with his mum's front-door key, then set about processing the recordings, as he describes it, "with a lot of reverse feedback". (By this, I assume he means that tape echo was added, then the tape reversed so that it played backwards.) Eventually, Brian played the finished results to Dick Mills and Desmond Briscoe; at their insistence he added a slowly rising note, played on the wobbulator.

Working Up A Storm

Brian and Delia Derbyshire were, as he says, "best mates. We used to go on holiday together." In 1966, together with the founder of synth maufacturers EMS, Peter Zinovieff, they formed Unit Delta Plus, a band of sorts, and began performing on London's psychedelic underground scene. As one Workshop member remembers it, "At the end of their day at the BBC they used to race off to the West End, changing into their kaftans in the taxi." Unit Delta Plus split in 1967, but some of their gigs sound like crackers: how about the two-day 'Million Volt Light and Sound Rave' at the Roundhouse? I'm sorry to have missed that one! In 1969 the pair teamed up with David Vorhaus as the White Noise, releasing the cult classic album *An Electric Storm.*

Meanwhile, the Radiophonic Workshop was going through some changes. The three-month rule ensured a steady throughput of staff, but some managed to become permanent. David Cain arrived in 1967, Malcolm Clarke in 1969; Richard Yeoman-Clark, Paddy Kingsland, Roger Limb and Peter Howell all joined in the early



'70s, just as Brian and Delia were leaving. The association with Peter Zinovieff had already led to the BBC buying three VCS3s, but in 1970 the Workshop took delivery of an EMS Synthi 100 modular system. It was the biggest voltage-controlled synthesizer in the world! Christened 'The Delaware', after the road outside the studios, it had 16 oscillators and even incorporated its own oscilloscope and frequency counter. As with the VCS3, there were no messy patch cords: instead were provided two 60x60-way 'pin patch boards'. There was a digital sequencer too, which could store up to 256 events. The massive control surface presented a sea of knobs to twiddle, but one of them, labelled 'Option 4' was actually a dummy.

Recording The *Doctor Who* Theme

"We got a phone call from Verity Lambert, the first *Doctor Who* producer," says Dick Mills. "She said she had a little sci-fi series that would only run to six episodes, but she'd like some special electronic effects. So me and Delia went along to Ealing for a meeting with her, and we said we could do the effects, but we could probably help out with a signature tune as well, as we'd just been working with Ron Grainer — a composer who was coming quite into vogue (he'd done themes for *Steptoe* and other shows). So Ron was hired to write the sig, and us to record it. Ron had originally come to us first, so we were returning the favour. We'd done a TV show called *Giants of Steam* and Ron had got us to make loops of train effects and process them to different tempos for his musicians to play along to. He had great confidence in us — for *Doctor Who,* he just handed Delia one foolscap sheet of manuscript paper and said off you go! Then he cleared off to Portugal for a fortnight — he said it was for the sake of his health..."

So how was the theme recorded? "Well, we started with the bass line. You know those 19-inch jack-bay panels? You could get blank panels too, to fill in between them. They were slightly flexible, so Delia found one that made a good musical twang, and played it with her thumb. We recorded it then vari-speeded up and down to different pitches, copied them across to another tape recorder, then made hundreds of measured tape edits to give it the rhythm."

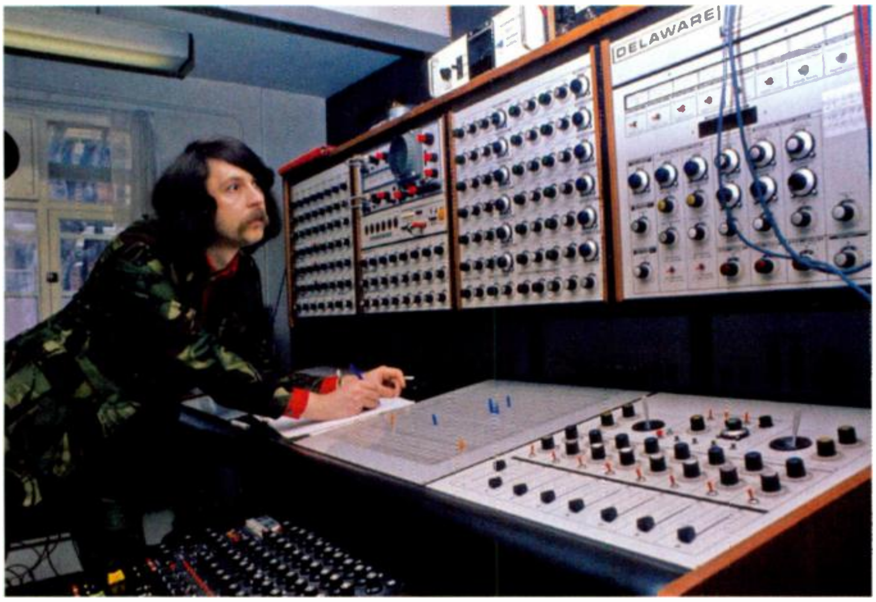
And what was the main tune played on? Was it some early synthesizer? "No," says Dick, "it was just a load of oscillators — signal generators — that someone had connected to a little keyboard, one for each note. Again, we had to make lots of tape edits."

But what about that distinctive portamento? How could you bend the notes like that without a synth? Dick sighs: "Well you just twiddled the frequency knob, of course — how else? It was all done with actual knob-twiddling then — there was no other way! We did it in lots of little pieces, then joined all the bits of tape together."

Eventually, after some pre-mixing, the elements of the entire composition existed on three separate reels of tape, which had to be run somehow together in sync. "Crash-sync'ing the tape recorders was Delia's speciality," says Dick. "We had three big Phillips machines and she could get them all to run exactly together. She'd do: one, two, three, go! — start all three machines, then tweak until they were exactly in sync, just like multitrack. But with *Doctor Who* we had a bum note somewhere and couldn't find it! It wasn't that a note was out of tune — there was just one little piece of tape too many, and it made the whole thing go out of sync. Eventually, after trying for ages, we completely unwound the three rolls of tape and ran them all side by side for miles — all the way down the big long corridor in Maida Vale. We compared all three, matching the edits, and eventually found the point where one tape got a bit longer. When we took that splice out it was back in sync, so we could mix it all down."

Ron Grainer returned from his holiday and famously asked if it was the same piece of music that he'd written. The theme was an instant success, as was the programme. But success brought its own problems, as Dick remembers. "The trouble was, because it was a hit show, every producer wanted to put their stamp on it, so they'd ask us to record another version. We did loads and no-one ever liked them. One was laboriously done on the Delaware. The sounds were great, but no-one liked it. I remember Delia did one version herself, where there was very heavy tape echo on the rhythm that gave it a new and different groove. The first time it was played in a dub all the technicians complained. 'Oh no — what's wrong with that?' they all said. 'Let's have the old one back!' And we also had to make a 45-second version when the show got popular. Anyone who's worked in TV music knows how difficult it is to turn a 30-second sig tune into a 45 — it's a very unnatural thing to do, musically."

As a footnote, there is still a difference of opinion on how the *Doctor Who* bass sound was created, 45 years ago. Dick Mills remembers Delia twanging a blanking panel in a rack, while Mark Ayres offered two versions — a plucked string and a rubber band (he heard both from Delia!). Peter Howell, meanwhile, told me: "The bass twang was a plucked bass string on a home-made electric pickup device (a piece of wood with a string on it). That sound appears on several early Workshop recordings."



Radiophonic Workshop composer Malcolm Clarke (1943-2003) with the EMS Synthi 100 modular synth known as Delaware.

Not connected to anything at all, it was occasionally tweaked to appease awkward producers who wanted to get 'just the right sound'.

Desmond Briscoe's retirement in 1977 saw Brian Hodgson returning as Workshop Organiser, after five years away. Brian finally managed to prise a reasonable annual budget out of the BBC and he set about systematically renovating the place, eventually providing a customised studio for each of the five composers. Apple Macintosh computers were introduced, and a lot of the new kit was identical to what could be found in any studio of the time; there were growing mutterings about the Workshop having somehow deviated from its original purpose to become a 'music-writing factory'. This was not really true: the Radiophonic Workshop had been founded because the equipment needed for electronic music production was not generally available. Mass-produced synthesizers did become affordable with time, but remember that when the first 8-bit digital sampler, the Fairlight CMI, appeared in the early '80s, it cost over £30,000: you could buy a house for that! The Workshop's composers were all producing work in their own styles, using equipment that may have been available to outside composers, but was prohibitively expensive for most. Elizabeth Parker joined in 1978 and her trademark sound came from the pricey and unreliable PPG 2.2. Richard Attree, who, in 1987, was the last composer to be taken on, made good use of the Yamaha TX816, which was effectively eight DX7s in a rack. Just one DX7 cost £1200 when it was new.

Peter Howell told me: "There's still this prevailing idea that we were somehow almost traitors for using modern gear and computers! Some people still believe that the original Workshop, with virtually no equipment, was the only incarnation that mattered. But we were there to do a job. With the Fairlight I could play something live, in real time; why on earth should I spent three weeks chopping up little bits of tape to get exactly the same result? We had to catch up with the real world — otherwise we'd never justify the time and cost."

Dick Mills (left) and Brian Hodgson compare the lengths of two sections of tape; watching is Desmond Briscoe.

The Cost Of Everything

Ultimately, it was costs that killed off the Radiophonic Workshop. The controversial appointment of John Birt as the BBC's Director-General in 1992 was the writing on the Workshop wall — for Birt brought 'producer choice' to the BBC. The asylum would be run by lunatics no longer: the accountants were taking over.

With 'producer choice', staff producers at the BBC could now either use the BBC's carefully costed in-house facilities, or they could choose to go outside — all that mattered was the cost. And everything in the BBC was costed. So what happened? In London, staff producers and directors cleared off to Soho in droves, to work with their old mates who'd already taken redundancy and gone freelance. For about a year, many BBC buildings felt empty. Everyone was eventually recalled and producer choice was 'modified', but the damage was done — it resulted in a catastrophic lowering of morale within the BBC.

Brian Hodgson struggled for a long time to keep the Workshop alive, but it was a losing battle. Under the Birt regime, every BBC department was assessed for profitability, and if running costs were found to be greater than profits, extermination followed swiftly. The Radiophonic Workshop had been doing a fine job providing quality music for many programmes that didn't have big budgets — schools programmes, in particular. But now the Workshop was expected to compete on the 'open market' with freelance composers like myself. Brian spent many months calculating the cost of finished music per minute and searching for ways to reduce it. I didn't even bother costing my music per minute: I didn't have to. If a director asked me for a quote, I could just say "Well, it depends... How much have you got?"

Despite this approach being the most obviously competitive, it was not permitted under BBC rules, and so in 1998 the Radiophonic Workshop finally closed its doors. John Birt was awarded a Life Peerage, by the way, and now sits in the House of Lords.

Daphne Oram (1925-2003)

There would have been no Radiophonic Workshop without Daphne Oram, despite the fact that she worked there for less than a year. She was a remarkable woman and a true pioneer, whose achievements have never been fully recognised. As well as her work in electronic music she also composed many orchestral pieces, all of them as yet unperformed. This year though, Sonic Arts Network (www.sonicartsnetwork.org) are to mount an exhibition and concert celebrating her life and work. Details will be announced in

Daphne Oram with the wobbulator (centre of shot), 1958.

SOS, or see www.daphneoram.org.

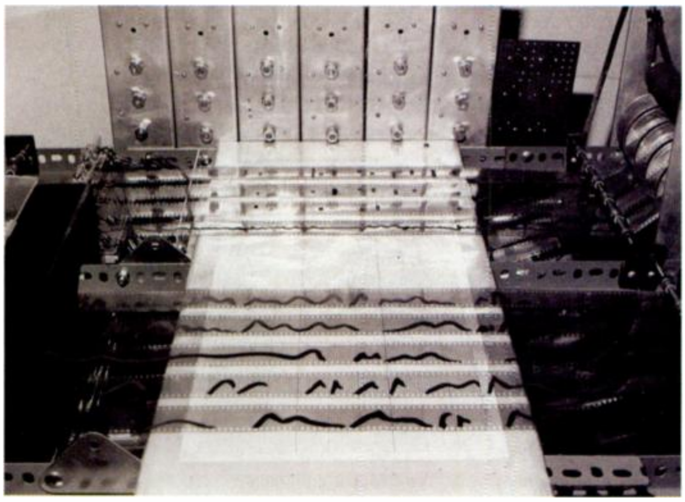
Daphne left the BBC in 1959 and moved to Tower Folly, a Kent oast-house that she had already started converting into a home and studio. Here she produced music for film and theatre, using the techniques of *musique concrète* and primitive electronics. Over the next years she was to develop her own Oramic Synthesis, an extremely novel way of producing electronic sounds.

At that time, the most advanced electronic instrument in existence was the RCA Electronic Music Synthesizer Mark II. Built in 1957, it consisted of a huge array of steel racks and was bigger than the average living room. The machine (which still exists) was controlled, or 'programmed' by means of a roll of paper, punched with holes. It also offered an alternative: the parameters could instead be drawn onto transparent film that passed over a series of photo-cells. Daphne's Oramic system was similarly controlled by drawing, but for each parameter there was a separate roll of 35mm transparent film (known as 'clear leader' in the film industry). The 10 rolls of sprocketed film were mechanically linked, and passed over a horizontal 'drawing table' where the operator could make marks on the film to control pitch, envelope, intensity, and so on. Additional rolls of sprocketed recording tape or 'mag track' could be used to record the results; this section was referred to as the 'multitrack' recorder.

Daphne seems to have preferred to draw onto the film using a brush and special ink, but felt pens or sticky tape could be used. Her machine had several oscillators with variable waveforms, again controlled by photo-electric cells. This part of the machine was even more bizarre: a selection of glass plates, each with a cut-out pattern, could be fixed to 'cathode ray scanners' to change the waveforms. It was effectively an oscilloscope in reverse! Reading contemporary accounts of how the Oramic system worked is confusing nowadays [*see photo and diagrams,courtesy Sonic Arts Network, overleaf*], as the words analogue and digital are used, but not in the sense that we know them. Continuously variable parameters were regarded as analogue, while those that could only be switched on or off were 'digital'. However, Daphne did eventually go digital in the modern sense.

I met Daphne Oram once, in 1989, and inquired whether she still used the Oramic system. Surprised and delighted that I'd even heard of it, she laughed "Oh no, not that old-fashioned thing!" She then explained that her old Oramic system had been swept away and replaced by something far more modern! She'd been working with 'some clever young chaps' who had helped her to build a new, computer-controlled synthesizer. "It's a huge improvement!" she said. "Now, when you draw the parameters, they're digitally scanned into a micro-processor..."

Yes, it still used rolls of 35mm film! ▶



The unique Oramics synthesizer was controlled by drawing onto 35mm photographic film.

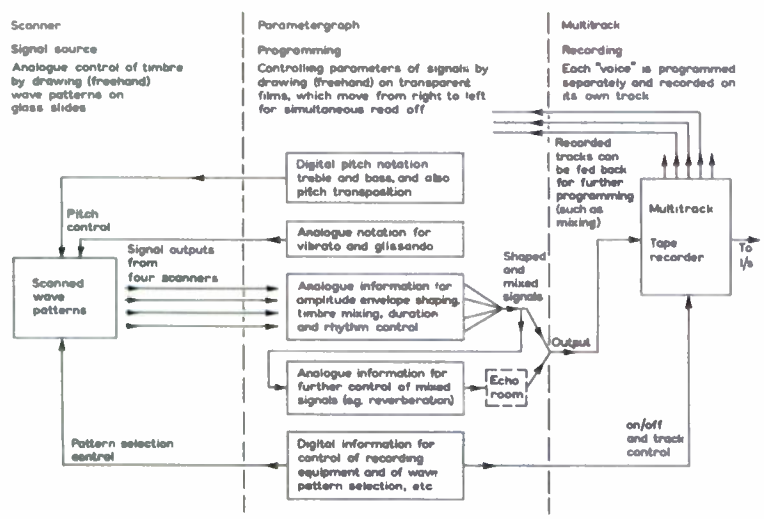


Fig 67 Oramics graphic sound

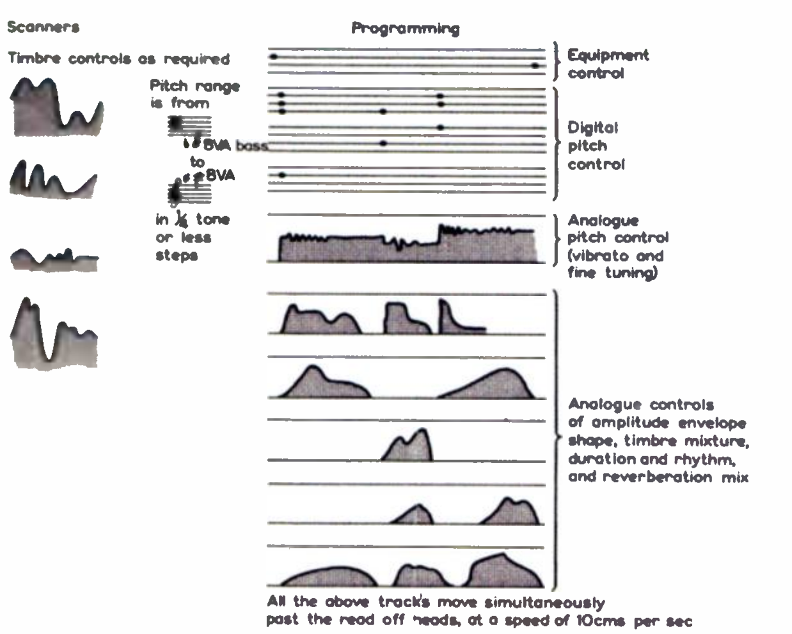


Fig 68 Expansion of Fig 67

Delia Derbyshire (1937-2001)

Although her name will be forever associated with her 'realisation' of Ron Grainer's *Doctor Who* theme, Delia Derbyshire (pronounced 'Darbyshire', by the way) proved herself to be an extremely original and sensitive composer. She had a degree in Music and Maths from Cambridge that may have accounted for her unusual and analytical approach to sound: she is reputed to have always carried a book of logarithm tables that she used in her work! During her time at the BBC, the Workshop composers were not always properly credited, so consequently there is no complete catalogue of her music. She also 'moonlighted', producing library tracks under various pseudonyms. Some of her music is available on CD, though, and she has a MySpace tribute page. Blue Veils & Golden Sands and The Delian Mode, two pieces that she made in the Radiophonic Workshop, are particularly outstanding, featuring organic sounds that seem to 'shimmer' as their harmonics slowly change. She claimed to have made the sounds by analysing the partials of her favourite metal lampshade and replicating them with sine-wave test oscillators! Before Delia, electronic music had a reputation for sounding 'ugly'; she proved that it could also be extremely beautiful.

In 1973 she left the BBC and gave up composing, working for a short time with Brian Hodgson at Electrophon Studios. Then followed a series of curiously directionless moves: she went to Cumbria to work as a radio operator on the gas pipeline; was briefly married; she ran an art gallery, and made a disastrous attempt at teaching music in York. Eventually she settled with a partner who brought much-needed stability. In the late '90s her interest in electronic music returned and she began working on an album, but sadly, it was never finished, as she died at the age of 64 after cancer treatment. In a 2001 obituary Brian Hodgson wrote of her: "One night many years ago, as we left Zinovieff's studio, she paused on Putney Bridge. 'What we are doing now is not important for itself,' she said, 'but one day someone might be interested enough to carry things forwards and create something wonderful on these foundations.'"

Dick Mills

Dick Mills spent virtually his entire career in the Radiophonic Workshop and holds the record for the most *Doctor Who* credits. Now retired, he finds himself increasingly in demand for *Doctor Who* and sci-fi conventions.

"I joined the Workshop in November of 1958, after it had been going for only about six months, so I'm now the oldest surviving member. As a duty engineer, I'd worked once with Desmond Briscoe on a very silly drama thing he'd done, which was set on the moon! It was about a couple of astronauts who went there and fell in love with a moon woman and when they got back to Earth one of the men was pregnant... It had weird sound effects and was great fun to do. So when I saw a notice

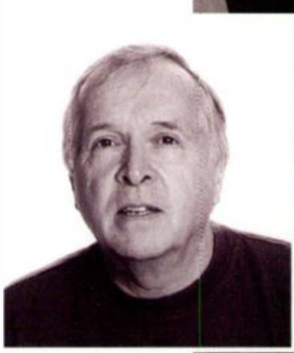
Dick Mills (left) and studio manager John Harrison attempt to control a very long tape loop!

Dick Mills today.

asking for someone to help out at the new 'Radiophonic Workshop' I jumped at it. I stayed there until I retired in 1993."

Dick's speciality at the Workshop was sound effects — not just sci-fi ones, but outrageously funny ones too. I asked Dick where his interest in effects came from. "From the early '50s there used to be a Saturday night DJ called Jack Jackson who did amazing things with records — cutting and mixing between music, comedy and sound effects. He was much appreciated by those in the business. Then there was the Goons — remember, all that stuff was done live, with a studio manager spinning in sound effects from 78rpm records."

For the Goons, Dick famously produced one of the best comedy effects of all time: 'Major Bloodnok's Stomach' was an outrageously long impression of a tortured digestive system. It has appeared on several BBC FX discs and was even sampled by the Orb! Dick remembers recording it: "I always wanted to work on the Goons, but Desmond Briscoe was in charge and he said no, because he thought they'd be unreliable and a nuisance (he was probably right). Anyway, Desmond was away on holiday that week... A producer came in and asked if we could do something for the Goons, so I just said yes! The finished thing was hysterical, and originally even longer, all cut-up burps, gloops and explosions. We just fell about laughing every time we played it. The producer sat and listened in silence, then said 'Well, it's all right, but we've only got half an hour for the show. We can't spend 30 seconds on one effect!' So we had to cut it down to 10 seconds for him."



Roger Limb, 2006.

Because the Workshop had a couple of in-house technicians, some of Dick's experiments would involve them building custom pieces of equipment. "I got obsessed with crossfades at one time," he laughs. "I wanted to be able to do longer ones, so I got them to make me a splicing block that was 18 inches long! No, it didn't catch on... We tried all sorts of variations on tape loops: I once tried splicing a Moebius strip. That didn't work either. The tape changed sides at the splice, so half of it was bright but the other half had top-cut because the tape was now upside down. Someone else invented a vibrato unit for tape! It consisted of a gramophone motor, attached to a biscuit-tin full of sand, to make it heavy enough; the motor had some sort of gearing, probably Meccano, to make an arm press periodically against the tape and give the vibrato effect. It worked, sort of."

Eventually, the Workshop began to be seen as uneconomic and unnecessary. *Doctor Who* had finished and there was no need for sci-fi effects any more. What could the Workshop provide that couldn't be found in studios anywhere else? The answer turned out to be intelligent noise removal: it was new and extremely expensive in the '90s (even hard drives cost a fortune then). So Dick's last few years at the Workshop were spent running a Sonic Solutions No-Noise system. No-Noise was a useful tool for TV production — from one small sample it could automatically remove hiss, camera noise, hum, and so on. Dick was set up in a brand-new new computer studio and was kept busy with archiving work, remastering video sound for DVD. "It was very interesting and satisfying work," says Dick, "but quite ironic really. I started my career with the BBC paying me to add horrible noises to their programmes; then in the end they were paying me to take them off!"

Roger Limb

"I'm 67 this year," Roger declares, "but I like to keep busy. I play keyboards with a rock & roll band and we gig regularly..." Roger Limb has always been busy: with his phenomenal output from the Workshop he must count as one of the most prolific composers in history! He spent his first few years with the BBC as a TV announcer. Then, some time in 1972, he bumped into Paddy Kingsland in the street outside Broadcasting House.

"Paddy and I had been studio managers together," he says. "He told me he was working at this fantastic little department in Maida Vale and that I should apply for an attachment — so I did! What they were ▶

Roger Limb attacks an empty tank with a mallet.

The Voice Of The Daleks

One of the most famous Radiophonic Workshop effects was the voice of the Daleks in *Doctor Who*, which was created by Dick Mills and Brian Hodgson. "We used a ring modulator," explains Dick, "the old-fashioned type, with two centre-tapped transformers and four diodes. Same as a bridge rectifier. They were 'improved' years later with a transformerless design, but the old ones could be distorted better. We spent a long time finding the right frequency to modulate the voice with, and eventually settled on 30 Hertz. But it's not as simple as all that, because they needed the actor who did the Dalek voice to perform live as they filmed. We set them up with a ring modulator in the studio (which they eventually lost!) and provided a reel of tape with a 30Hz tone on it. They'd run the tape, the actor spoke into a mic, both went through the ring modulator, and it sounded like a Dalek.

"But if the tape was supposed to run at, say, seven and a half inches per second, they'd sometimes run it at 15ips by mistake, or at three and three quarters. So that's why, for all you *Doctor Who* anoraks, the Dalek voices are slightly different in some episodes — if so, it was a mistake! I did other experiments with modifying the tape containing the tone — distressing it and removing bits of the oxide. It was a good effect, but was never used."

doing was what I'd been dabbling with at home for several years. I'd been dangling microphones inside pianos and just playing with interesting noises. It had never occurred to do me that this could be a career." Roger had been with the BBC for over five years, but before the Radiophonic Workshop he'd never heard of such delights as tape loops. "What impressed me the most," he says, "was vari-speed. I'd never thought it possible! There was 15 ips [*inches per second*], seven and a half, three and three-quarters — but it hadn't occurred to me that there could be anything in between!"

Roger was yet another victim of the dreaded three-month rule: after his allotted time he duly left, and was only able to return when a place in the Workshop was advertised (internally, of course) in 1974. Roger remembers the instruments of the early '70s: "There was the VCS3 and the Delaware, both of them certainly ground-breaking, but not terribly reliable. The VCS3, in particular, used to drift out of tune all the time. I was told that this was due to their being made with poor components. But you must remember that although we now call them all 'keyboards' they were often played, or controlled, without a keyboard, just by twiddling knobs. I do remember there was an attitude back then that using keyboards as controllers was probably just an interesting cul-de-sac, almost a passing fad! I did love the ARP Odyssey, though — it had a decent keyboard and it was very musical. It felt like a real instrument."

Roger says that the mid-'70s saw crucial changes in the way that the Radiophonic Workshop was run: "The original tape-splicers, John Baker and Delia Derbyshire, both left and it became much less experimental. With the likes of Paddy and myself coming in as musicians, it became more of a music-making factory."

The equipment was changing, too. Paddy and Roger began recording their tracks onto the Workshop's two eight-track recorders, which speeded up the business of making music considerably. "In 1985," says Roger, "the Fairlight arrived, and I think that one instrument changed music, and the way it was to be made, forever. I was a big fan of the Fairlight, and once when I travelled to Australia I called in at the factory to meet one of the inventors, Kim Ryrie."

I asked Roger if he had any other favourites. "The Yamaha CS80 was a lovely instrument — very expressive. I had an Oberheim that I was very fond of; I loved the Prophet V. The Delaware was an amazing instrument, but so labyrinthine that you could disappear for weeks just making sounds! We never had any Moogs, you know — although I believe that Mr Moog himself once visited. We did get an awful lot of visitors, particularly musicians who were working in the other studios. One day I was leaving my studio for a coffee break and as I opened the door I almost knocked over Marc Bolan, who was listening outside! He looked very sheepish and apologised. 'I've always wondered what went on in here,' he said, so I invited him in to have a look around. He had an appointment and said he'd love to have a tour the next time he was at Maida Vale, but it never happened. Two weeks later he had his fatal car crash."

So how would Roger sum up his time at the Radiophonic Workshop? "I feel very fortunate that I had the best job in the world for 20 years — I'd have done it for nothing! Well, maybe not *absolutely* nothing..."

Peter Howell

Now a lecturer in Screen Music at the National Film & Television School, Peter Howell started his musical career in the late '60s, playing 'psychedelic folk' with Agincourt and other related bands. Peter

eventually got a proper job as a BBC Studio Manager, but after a few years he managed to become a full-time member of the Workshop. "I started in '74 — the same year as Roger Limb," says Peter. "John Baker was still there, but we sort of crossed over. It was a funny period, really. I saw how to do the tape-splicing techniques, and had a go myself, but this was just when synthesizers were becoming available, and that was what really interested me."

Peter later became known for his work on the Fairlight, but he was happy to be the guinea pig for any new gear that came into the Workshop. "What I really found satisfying," he laughs, "was making beautiful sounds from ugly, clinical-looking machinery. The Fairlight was one of the ugliest instruments ever! I enjoyed using the VCS3 a lot; with the eight-track recorder I could make a whole piece using only the Odyssey, which I was very keen on.

"Then polyphonic synths appeared. I tried the Polymoog and really didn't like it; I liked the Prophet V, but my favourite was the Yamaha CS80. When I did Jonathan Miller's TV series *The Body In Question* I really wanted one, as I'd just seen it demonstrated. There was no money for one at the Workshop but we got the programme, which did have a decent budget, to hire one for me. It became a hit series, so later the BBC was later shamed into buying me one. It was a wonderful machine: polyphonic, though only eight-note; it was so expressive, with soft-action pads, and a great long pitch ribbon that you could play like a violin string. My party piece was to play the hornpipe just on the ribbon! I used the swell pedal constantly and this became crucial to my technique. Later, when we got MIDI sequencers, I used a volume pedal in the same way — so I ended up with files that were huge with all the Controller 7 changes."

Peter Howell with his beloved Fairlight.

Peter Howell today.

Peter was an early convert to making music with computers: "I did love having a room full of actual things that made noises, but what appealed to me most about computer instruments was the fact that all the settings could be memorised. Previously, I used to dictate all my studio settings into a cassette recorder, especially if there was a chance that someone else might come in to use my studio and change something. People would call in as they were leaving at the end of the day, and I'd be crawling around on my knees, calling out 'Attack seven; decay three; sustain nine...' It could take me 20 minutes to do the whole studio!"

Paddy Kingsland

Of all the composers who passed through the Radiophonic Workshop, Paddy is possibly the best known, because of his prominent credit on the end of each episode of *The Hitchhiker's Guide To The Galaxy*. Paddy was originally a guitarist, playing in several semi-pro bands; after several years as a Radio 1 Studio Manager he joined the Radiophonic Workshop. "When I started in 1970 there were three rooms — 11, 12 and 13 — plus the 'Piano Room' and an 'Organ Room' that housed a great big electronic organ that someone thought might be useful. It wasn't. John Baker was in room 11: he had three Phillips tape machines and the room was lined with hooks that had hundreds of tape loops hanging from them. John had a playback machine (a Leevers-Rich?) with vari-speed, and the speed control had been marked up in semitones. He would play his original loops on this, change the speed and run off copies onto a standard 15ips machine. In this way he'd make all his notes first, then splice them together to make the music. And he used to listen to Radio Four while he did it! I tried his technique myself and really enjoyed it — one track I made used DIY effects like hammers and drills.

"Next door in Room 12 were Brian Hodgson and Delia Derbyshire with a VCS3. They had Electrophon Studios and a connection with EMS, so they'd persuaded

The Radiophonic Workshop's Greatest Hits

When asked for a discography of the best ever Radiophonic Workshop releases, Mark Ayres came up with his top seven 'in no particular order'.

- ***BBC Radiophonic Music* (aka 'the Pink Album')**

Early Radiophonic wonderfulness from Delia Derbyshire, David Cain and John Baker. Originally released as a mono vinyl album in 1971, catalogue no. REC 25M; now also on CD (BBC REC25MCD), remastered with two extra tracks.

- ***The Radiophonic Workshop***

Compilation of material from the early '70s, released as a stereo LP (REC 196) in 1975; remastered CD (BBC REC196CD) includes two additional tracks.

- ***21***

Don't be put off by the terrible birthday-cake cover, this is a quality compilation of material from the Workshop's first 21 years. Released as a stereo LP (REC 354) in 1979.

- ***Fourth Dimension***

Theme and test-card music from Paddy Kingsland. The Workshop goes lounge. Stereo LP (RED 93S) from 1973.

- ***Through A Glass Darkly***

Peter Howell's solo album from 1978 (stereo LP, catalogue number REC 307). Side one is 'a lyrical adventure' (ie. one long track, done 'after hours' for the fun of it). Side two is comprised of shorter tracks, including Peter's classic *The Astronauts*.

- ***Doctor Who At The BBC Radiophonic Workshop, Volume One: The Early Years***

CD compilation of *Doctor Who* music and sounds from the '60s. Originally released by BBC Music, later re-released on the Grey Area sub-label of Mute.

- ***Doctor Who At The BBC Radiophonic Workshop, Volume Two: New Beginnings***

Continuing from *Volume One*, this dives into the '70s. Includes Malcolm Clarke's music for the 1972 story *The Sea Devils*. In effect: 43 minutes of Malcolm fighting with the Delaware. The jury is out as to who won.

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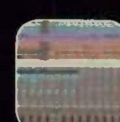

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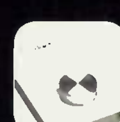

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Saturday 12th April	DV Cambridge:	12pm, 2pm and 4pm
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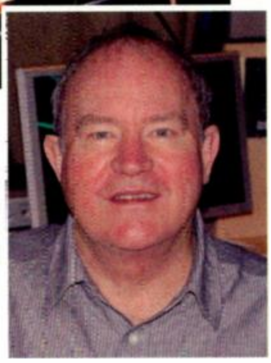


Paddy Kingsland, at the Radiophonic Workshop in 1980 (left), and today.

the Beeb to buy some VCS3s. I always found them great for effects but not very tuneable. Then in room 10 there was the Delaware. Composer Dudley Simpson used the Delaware a lot for *Doctor Who*: he would arrive with an eight-track tape he'd recorded with live musicians in Lime Grove Studios, then, working with Dick Mills, he'd somehow sync up to the Delaware and add extra electronic tracks. He used the sequencer a lot."

Paddy, though, along with Roger Limb, was largely occupied with getting as much finished music out of the door as possible — every day. "I enjoyed it, but as I wasn't really into 'weird', I didn't feel I was doing anything that couldn't have been done anywhere else. Not until I started on *Hitchhiker's*, that is! Then suddenly I thought the Workshop came into its full potential: it was using the place properly."

The first episode of *The Hitchhiker's Guide To The Galaxy* was actually a one-off pilot to test the idea. Actors' voices were recorded onto eight-track tape in the studios at Broadcasting House, then sent over to the Radiophonic Workshop to have the effects added. Paddy really went to town and created some extremely original sounds, many of them using his latest gizmo: the Eventide Harmonizer. "I did use it a lot," says Paddy. "For processing voices, mostly. Marvin the Paranoid Android used it; Eddie the Shipboard Computer, the Vogon Space Captain... It was the first real-time digital pitch-changer. The Vogon voice was treated with an echo that went up in pitch with each successive repeat, as the Harmonizer had been patched into a delay line."

Douglas Adams' witty script and Paddy's innovative sound effects proved to be a great combination; the pilot was a success and a further five episodes were commissioned to make up series one. But there was a problem: "All of this took ages, and I'd been moved on to a radio series that was to take six months — I just wasn't available to do it, so Dick Mills and Harry Parker came in and took over." Producer Simon Brett had put the first episode together and introduced Douglas Adams to the Workshop. The writer immediately saw the potential. "The first series was a big hit," says Paddy, "and I came back on board for the Christmas Special — you know, the one where the robot falls down a lift shaft... This time I made the music too [*apart from the Eagles' signature tune*]."

Paddy went on to make effects for another radio series of *Hitchhiker's* and then the TV series. "My biggest mistake when I did the TV series," he admits, "was to add squiggledy 'computer' noise to the book sequences as the letters drew across the screen. It looked great, though! I did it for the first episode and the director loved it. 'Great effect,' he said. 'We must have it for all the other episodes,' So I was stuck with the laborious task of cutting the sound to picture, using 16mm magnetic track, for the entire series! It took ages, doing it all by hand using an old-fashioned film splicer."

Paddy left the Workshop in 1981 and set up his own PK Studios in London (www.pkstudios.co.uk), where he now works. "It's a great mix," he says. "My son works with me and we do TV post-production, some music, a lot of film dubbing. What I enjoy most is Foley work — making sound effects, live to picture. I could happily do that all day."

Mark Ayres, Radiophonic Archivist

A life-long *Doctor Who* fan, Mark Ayres first visited the Radiophonic Workshop as a schoolboy! He kept up contact and eventually returned years later as a freelance composer, now working on *Doctor Who* himself. Mark is now a member of the BBC's unofficial '*Doctor Who* Restoration Team' — a group of dedicated fans, some of whom are BBC staff. The team has been responsible for restoring 'lost' episodes and remastering many DVD releases. Mark Ayres has done much of the audio restoration, and was also responsible for rescuing the Radiophonic Workshop's tape archive when the place was closed in 1998. "I suddenly got phone calls," says Mark, "from Brian, then Peter Howell, then Paddy... They all said 'Someone's got to get in there and save the archive before it ends up in a skip!' — so I did."

Doing so took a great deal of time and effort, almost costing Mark his career. "I'd just done my first feature film score," he says, "and I should have been out promoting it and trying to get another. But instead, I spent 18 months in Maida Vale, cataloguing tapes."

The Radiophonic Workshop was unique within the BBC, as it was the only department to hold its own archive. Absolutely everything was kept. When DAT tape came along in 1988, composers were ordered to continue making quarter-inch copies, as no-one knew then how long DAT tapes would last. The tapes were all stored in three cold, dark, tomb-like rooms. "The Workshop had closed and no longer existed," says Mark, "but they had a system whereby it was still being charged rent by the BBC for storage! So all the tapes were taken out of the three store rooms and crammed into Dick's old studio. And they were now all out of sequence." Mark was told that some of the later tapes had been thrown out to save space, but that wouldn't matter, because "it will all be on DAT anyway".

"So," says Mark, "having messed up the archive, the BBC paid me (not very much, I might add) to sort it all out again." He started with the oldest tapes and worked his way through the pile. When he got up to 1983, all the rest of the tapes were missing. "They'll be the ones that are on DAT," he was told. Pointing out that DAT had not yet been invented in 1983, he set about scouring the building. The tapes, he discovered, should have been put in a skip, but by some fluke the paperwork had not been done — so they must still be in Maida Vale somewhere. "It took a whole week," he says, "of borrowing keys and opening rooms that no-one had been in for years. Eventually I opened a room labelled Band Store and there they all were!"

The tapes are now safely stored in the BBC's main archive, but are 'non-accessioned', meaning that no-one apart from Mark really knows what is there. "They all need properly digitising and cataloguing," he declares, "but it takes forever to do. There are three and a half thousand reels of tape. Ten of the reels are John Baker's sound sources — his sample library, if you like. But they're 40 years old, and full of splices that are either dry and falling to bits, or gone sticky. You have to copy a little bit, clean the heads, copy another bit..."

He started by concentrating on the *Doctor Who* tapes. "There are about 250 reels of sound effects," he says, "each up to

40 minutes long and containing about 100 sounds. It's an enormous task."

Mark has remastered four CDs of Radiophonic music so far. He started by pulling out the quarter-inch masters for the first two albums that had originally been released on vinyl, and discovered that they came with extra unreleased tracks. He hopes to continue, but as he says: "The funding just isn't there. I started remastering them as a labour of love, really. I'd work slowly on remastering an album, then deliver it to BBC Music when I'd finished. I phoned them up one day and said that I'd got another album ready for them, after several months of work, and there was an embarrassing silence. 'Sorry,' they said, 'we don't have a label any more!' Mark hastens to add that this problem has been resolved: BBC releases are now licensed to other labels, and he also has the support of an enthusiastic music department. "What this project needs, though," he says, "is lots and lots of time. And some money!"

To celebrate the 50th anniversary of the founding of the Radiophonic Workshop, Mark is compiling a two-CD set of Workshop music. It will comprise the two classic Workshop compilations *21* and *Soundhouse*, as well as an hour of previously unheard material. Details will be announced in *SOS*, and you can find out more at Mark's web site: www.markayres.co.uk.

Ray White, Engineer

"I had several attachments to the Workshop," says Ray, "They interviewed me a few times for a permanent job, but I was very bad at interviews. Desmond Briscoe really wanted me to stay, so eventually he just fiddled it! We had a 'rehearsal' for the interview and sure enough, next time I passed."

Ray spent most of his BBC career as an engineer in the Radiophonic Workshop, fixing, building and modifying anything electronic. Arriving in the early '70s, he stayed for 20 years. "In the early days it was almost like a club," he says. "It was great fun, going to work. If they thought you were right, the management would welcome you in — then recommend that you join the Union! That would just not happen nowadays."

Ray is proud of his association with the Radiophonic Workshop but points out that not all the music produced there was good. "There was some awful dross came out of the place at times," he says, "and no-one mentions that. I think it was at its most successful when it combined electronic innovation with something more traditional. Like a tune... The *Doctor Who* theme is the best example. Could you imagine anything like that ever coming out of, say IRCAM in Paris? They've produced so much stuff in that place that is clever, and pushes the limits of music technology, but it all sounds horrible! You wouldn't want to listen to that in your lounge, would you?"

Ray White, whose engineering expertise made many of the Workshop's experiments possible.

In 1993 Ray decided to take early retirement. "As soon as Birt was appointed, I could see what was to come," he says, "The Workshop had gone as far as it could and it had served its purpose. Looking back, it was so difficult for those early pioneers to achieve what they did." Ray cites film composer Tristram Cary: "He was making electronic music in his home studio in the '50s - building his own gear too. He'd get stuck halfway through a composition, then have to get out his soldering iron and build some new machine, just so he could finish the track! By comparison, it's so easy to make electronic music today. But that means it's even easier to produce rubbish!"

Ray White's web site contains the most detailed account of the Radiophonic Workshop and its equipment: http://whitefiles.org/rws/index.htm.

Better Late Than Nedder

My own three months in the Radiophonic Workshop in 1988 were spent in Malcolm Clarke's Studio C, which was at the end of a short corridor running past Dick Mills' Studio D. I was covering for Malcolm, who was off sick, so I never got to know him (Malcolm died in 2003). Dick's approach to sound work was extremely practical and no-nonsense: his small studio was brightly lit with fluorescent tubes and resembled a laboratory. Malcolm's studio, on the other hand, was dark and moody; decorated entirely in red, at his insistence (something to do with the primal nature of creativity, apparently). Some witty technician had installed a tie-line box on Malcolm's studio wall; it included a dummy jack socket embossed with the words Fine Art Output.

One morning, Dick showed me his party trick. "Have you ever seen this before?" he chirped, producing a full 10-inch NAB spool of quarter-inch tape. In the centre of the hefty aluminium spool was a large circular hole, with three more sharp indents. Holding the spool balanced on the flat of his left hand, he deftly laced the tape into a Studer A80, winding it onto an empty take-up spool. He jabbed a button and put the Studer into fast-forward. The Studer is a huge, heavy beast of a machine, mounted flat on its back in a wheeled caddy. The enormous size of its reel motors means that 'fast forward' is terrifyingly fast. As the machine whizzed into action, Dick gently patted the full NAB reel into the air and kept patting to make it hover just above his hand as it spun faster and faster. As the spool emptied, it began spinning even faster still. "Now the tricky bit!" shouted Dick above the whooshing and whirring sounds that rose steadily in pitch. The tape had almost all come off the spool; it was spinning dangerously fast already. The last bit of tape came off and whipped the spool like a top. With that, Dick tossed the reel up into the air above his head, then suddenly clapped his hands together and caught the empty spool between them. The spinning and the noise immediately stopped. "You do have to be careful not to catch your fingers," he said.

Finally, an opportunity to work with Dick Mills came with a radio sci-fi show for BBC Schools called *Slambash Wangs Of A Compo Gormer*. Dick was to make the sound effects and I was to start with the music and make some effects if I had the time. Eventually, schedules slipped and all I managed was a signature tune. One of the effects was the sound of 'a galloping Nedder'. A 'Nedder' was a six-legged horse, in the alien world in which the series was set, and Dick and I agreed that whoever had some free time first would make the Nedder effects. I kept thinking of complex and sophisticated ways to do this, most of them involving samplers and/or coconut shells.

One day I saw Dick as I passed his studio. "I've done the Nedder," he said, and proceeded to play me it. It was perfect — exactly like a six-legged horse.

"How did you do it?" I asked. "Samples? Library discs?"

Dick reached out to his bench and picked up an empty plastic cassette box. He held it close to my ear, then rapidly drummed his fingers on it.

"Voila!" he said. "There goes a Nedder!" SOS

Euphonix MC Mix

DAW Control Surface

Paul White

The new Artist series brings Euphonix's innovative Eucon technology to the project studio market. Does the MC Mix put existing fader controllers in the shade?

At the NAMM show earlier this year, Euphonix gave us a sneak preview of their Artist Series MC Mix and MC Control units, two new control surfaces that support most mainstream Mac-based DAW software packages. Where the software vendors have implemented Euphonix's own Eucon protocol, this is used to connect the Artist controllers, but they are also capable of emulating Digidesign's HUI protocol and the Mackie Control protocol. Each Artist may be used independently, or they can be combined as part of a larger system. We were sent the more conventional fader-based MC Mix controller for review; the intriguing MC Control, with its touch-screen interface, should be following imminently, along with Windows support.

Until now, Euphonix products were mainly aimed at the professional working on a professional budget, but these new Artist controllers are pitched to appeal to both pros needing a compact control system and to project studio owners. The technology behind the interfaces is drawn directly from the Euphonix high-end control surfaces, as is the Eucon protocol itself. Like the larger MC Pro controller reviewed in last month's *SOS*, the MC Mix hooks up to the host computer via Ethernet. A Eucon control utility is continuously active while the controller is connected, but doesn't seem to impose a significant CPU load. A separate MC Client application routes the Eucon control data to all the supported applications.

I say 'all', because Eucon makes possible the simultaneous control of multiple applications or even multiple workstations. The use of Ethernet means that data bandwidth is vastly higher than MIDI, which most controllers in this price range use, and Eucon also offers better data resolution than MIDI, where controllers are commonly restricted to 128 discrete values.

The MC Mix is designed to work to a limited extent with any application, but for fader and knob control the application must either support Eucon directly or via the HUI or Mackie Control protocols. Where applications lack support for any of these protocols, the buttons can still be mapped to send out user-specified keystroke commands, so although you wouldn't buy an MC controller to work with unsupported software, it may still be useful.

Most of the major DAWs now work with the Artist series of controllers directly, by incorporating Eucon support, among them Apple's Logic, Steinberg's Nuendo (and soon Cubase), MOTU's Digital Performer and Cakewalk's Sonar. The big name missing from this list is Digidesign, and Pro Tools users will need to use the HUI emulation. Mackie Control emulation should take care of other non-Eucon DAW software.

Size Isn't Everything

The first thing I noticed about the MC Mix was just how slim and compact it is, despite it having eight 100mm, touch-sensitive motorised faders, eight touch-sensitive rotary encoders with integral push switches and a clear LED display above each fader, using Organic LEDs (OLEDs, which have various advantages over LCDs). Overall the unit measures 16.5 x 9.5 inches and is barely an inch thick, yet with its neat layout, clean lines and uncluttered panel, it looks thoroughly professional.

Up to three further units, or a setup incorporating an MC Control and up to four MC Mix units, can be linked to provide a larger-scale control surface. Short swivel legs beneath the unit allow it to be angled up slightly, and there are also plastic clip-in pieces that can be used if a bit of extra height is needed. Where multiple units are used together, they can be physically joined, after removing the relevant end-cheeks, to produce a single entity, which makes for a very neat installation. Power comes from a separate switch-mode power adaptor, and the connector for this is on the rear panel, along with the Ethernet socket and a jack socket for an optional punch-in/out footswitch. Most modern Macs need only the included crossover Ethernet cable to get up and running, but an Ethernet switch or hub is needed when hooking up two or more units or for enabling control over a network.

Once the supplied software is loaded from its CD-ROM, there is little or no setting up to do, depending on what DAW software you're using, though I had some teething problems getting my Ethernet connection to work. If your host computer is already configured for network use, you might need to give the MC Mix a fixed IP address too, rather than relying on the default dynamic allocation. This had been necessary in order to get the review model working for the photographs, and the manual didn't make it entirely clear how to change it back again — though a few tentative button presses and choice swear-words soon fixed it!

A grey 'E' symbol appears in the menu bar at the top of the Mac's screen on start-up, and this turns green when communication has been established with the MC Mix. This is supposed to happen automatically, but on both my MacBook Pro and my studio G5 I found I had to click on the 'E' symbol to bring up the Eucontrol ▶

SOUND ON SOUND

Euphonix MC Mix £699

pros

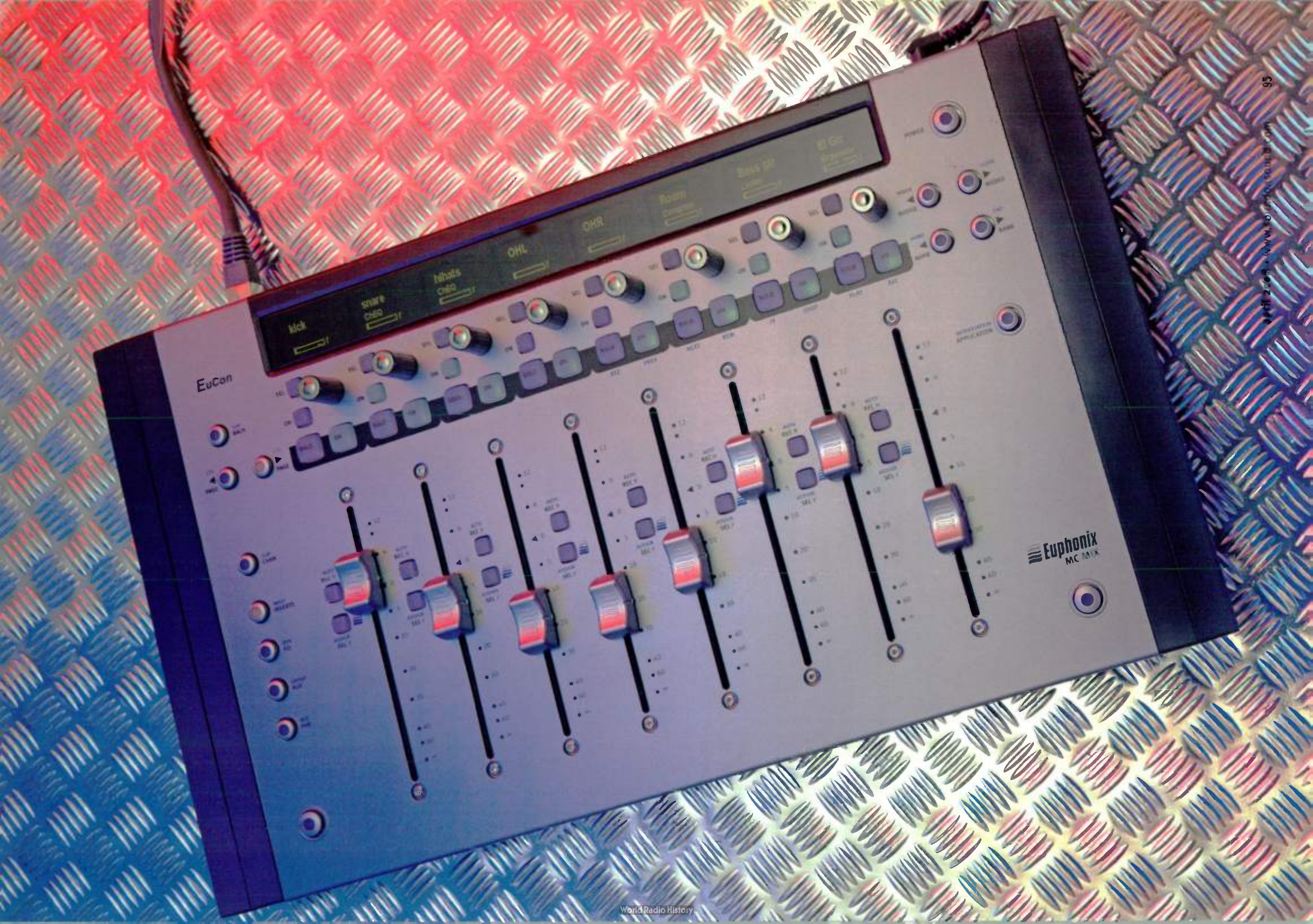
- Intuitive but powerful user interface.
- Supports multiple DAWs, automatically recognising which one is active.
- Affordable.
- Stylish and compact.
- Multiple units can be mechanically locked together to form a single control surface.

cons

- Panel legending unnecessarily small.
- Button LEDs in transport mode could be more helpful.
- No jog/shuttle wheel or display zoom buttons.

summary

This is a very serious controller that looks great, takes up little desk space and supports all the mainstream DAWs.


kick
snare
hihats
OHL
OHR
EuCon
Euphonix
MC Mix

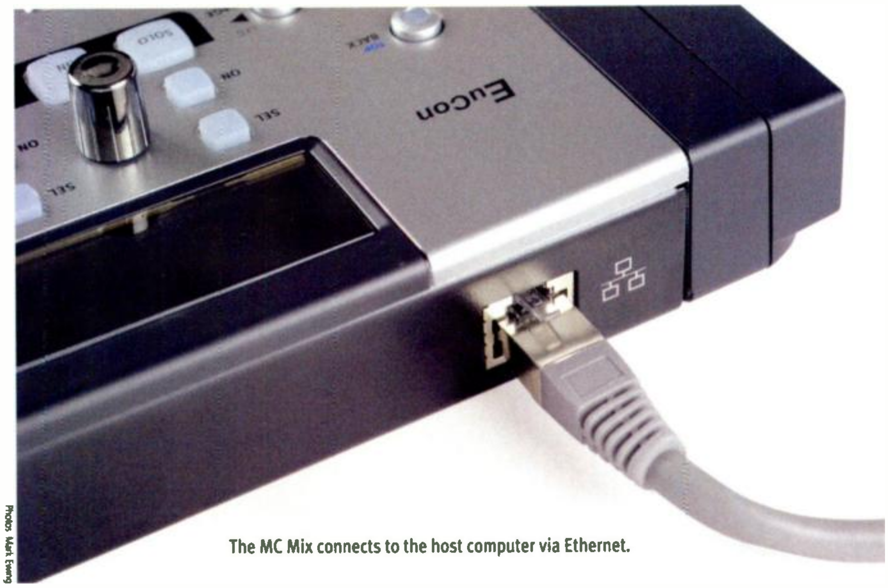
Settings panel. This has a list on the left called All Surfaces and a list on the right called My Surfaces, and I had to add the MC Mix manually by selecting it in the All Surfaces list and hitting the Plus button, which then puts it into the My Surfaces list. Once the above step had been completed, the 'E' symbol glowed green, the MC Mix burst into life and everything was happy.

What's On The Menu

In addition to its small size, the modest number of buttons on the MC Mix comes as something of a surprise, and a dedicated transport section is conspicuously absent, though this isn't an insurmountable problem as will be revealed shortly. At the two lower corners are Shift buttons which activate a second function for each control (printed in blue), but so well has the operating system been devised that relatively few buttons are actually necessary. You can lock the Shift keys by pressing both together.

Each channel has Select and On buttons next to the rotary encoder, Solo and On buttons below the encoder, and Record and Select buttons alongside the fader. The encoder On key toggles dual-state parameters, such as switching EQ bands on or off, while the Sel key changes the knob function or, for example, switches from pre- to post-fader, depending on the context. The individual LED display above each fader strip displays metering and track details for the associated channel in your DAW. Though the LED windows are quite small, the displays offer very high definition with good contrast, and show a lot of information, including level metering, automation mode, track name and pan position. If you're working in surround, they even show very skinny 5.1 metering to the left of the window, and whenever a track is selected as the 'attentioned' track a horizontal line appears in the display. The displays dim when not being used, to extend LED life.

Touching a fader automatically selects the corresponding track, though the touch function can be disabled if required, in which case the channel Select button does the same job. The Shift function of the Fader Select button is used to assign parameter control to a fader in cases where an application isn't directly supported, so you can, in effect, create your own control system for these programs, albeit to a more limited extent than those that are directly supported.



Photos: Mark Ewing

The MC Mix connects to the host computer via Ethernet.

Three sets of left/right cursor buttons deal with nudging through channels or stepping through banks of channels or edit pages, with a separate Back button for retracing your steps. Down the left of the panel are five further knob mode buttons designated Channel, Inserts, EQ, Aux and Pan. When Channel is off, the unit is in 'Normal' mode, where the knobs control the same function across eight different channels. When Channel is on, the eight knobs control up to eight different parameters relating to a single 'attentioned' track. Channel mode is indicated in the LED display by a grid of small yellow dots around each parameter title and knob position indicator, and the 'attentioned' track has further dots around its entire display area to highlight it.

The other four mode buttons get you directly to the named functions, enabling plug-ins, EQs and aux sends to be inserted or created, and their parameters adjusted. Under some circumstances, knobs have dual functions that may be toggled by pressing the knob's Select button, EQ frequency and bandwidth being one example. The displayed parameter name changes when Select is pushed, in this case.

Shifted functions include Flip, which makes the faders and rotary encoders swap functions, Input, Dynamics, Group and Mix. Shifting the left Nudge button opens and closes the mixer page of your DAW, where this function is supported, and the shift button also turns the last four channels' Solo and On buttons into a full set of transport controls, while leaving the On and Solo buttons of the first four channels functioning normally. Engaging the Shift lock keeps the transport available all the time, but the LED status within the buttons now functioning as transports doesn't change accordingly. I think this is a bit of a shortcoming, as it should have been possible to make these emulate a conventional transport section (either all lit or all off) and for the record button LED to turn red instead of green when active.

Shifting the track Rec buttons accesses the DAW's track automation and steps through the available automation modes. Another very neat touch is that you can assign the 'attentioned' track to a dedicated fader, while you to 'bank' through tracks with the remaining faders. This is a good way to keep the currently selected track always to hand as you change banks. To get back to the first eight channels quickly, you just activate the Home function (Shift plus left Bank).

In Use

When your DAW is the active application the MC Mix jumps to attention, and if you have multiple DAWs running at the same time the MC Mix simply aligns itself with whichever one is in use. Apple's Logic needed no setting up at all — you just fire it up and go. If you switch to an unsupported application such as a word processor, the faders return to zero and the displays and LEDs go blank, other than a small rectangle in each display window. There are also protocols for manually switching between multiple open applications via a dedicated button to the right of the front panel and a Shift

System Requirements

- Apple Macintosh G4 with a 1.25GHz or faster Power PC G4 processor; Power PC G5, Intel Core Duo or Intel Xeon processor highly recommended.
- Mac OS 10.4 or later.
- One available 10/100 Base-T Ethernet port.
- An Ethernet hub or switch is required to connect additional MC Mix or MC Control unit(s) or to connect to a network.
- 1GB RAM.
- 100MB of free hard disk space for full installation.

function for closing open windows. However, there's no fader kill mode that I could find; when you're doing a final mix it would be nice to be able to disable the faders so you can listen to the final product without the sonic accompaniment or visual distraction of moving faders.

Overall, I love the straightforward approach of the operating system. It is extremely intuitive and seems to be more obvious than, say, a Mackie Control, though not sufficiently different that it would take a user long to move from one to the other. With my Mackie Control, for example, selecting channel mode to see all the sends associated with a single channel is easy, but when you want to see specific controls associated with multiple channels, this is less obvious, and the default mode seems to be to allow you to change the designation of a send rather than its value. To this day it confuses me, but the Euphonix MC Mix just did what I expected it to do.

The default start-up mode controls levels and pans across eight channels, but if you hit Aux, you see the control for the uppermost send slot in each channel, and using the Page arrow buttons, you can move up or down through the sends — very straightforward. The function of the rotary controls then changes to adjust the send levels, just as you'd expect, while activating Channel mode shows the first eight aux sends associated with the selected channel across the display section.

When you select Inserts, the Page buttons can again be used to move up or down the insert slots in the eight channels. Pressing a knob opens the associated plug-in window and shows the parameter values across the LED windows — again very intuitive. Doing the same thing with Channel mode active shows the first eight plug-in slots associated with the 'attentioned' channel (any more are shown on a second page), and this choice of 'horizontal or vertical' access extends to most channel functions.

MC Control

If the MC Mix looks a bit short on transport control features, that's because the forthcoming MC Control looks after all that and more. It's based around a data wheel and a colour touch-screen interface, with several function keys that access the shifted functions of other buttons on the MC Control, nine rotary controls, a punch in/out footswitch jack and four of the same high-quality motorised faders as on the MC Mix. The touch-screen can also display things like EQ curves and song data, and it is possible to add up to four MC Mix controllers to create a 36-fader control surface. A number of ergonomically placed buttons around the data wheel enable it to control multiple functions, though the actual operating system and the extent of its transport control abilities won't be known until we get hold of one to test.

Reading & Writing

All the buttons include bright status LEDs that leave you in no doubt as to what mode you're in, which is essential in a product such as this, but I do have an issue with the tiny legend size on the MC Mix's front panel. In my office, which is illuminated by one of those energy-saving bulbs that laughingly claims to give the equivalent of 100 Watts of light (as seen from Mars, perhaps!), I couldn't read any of the panel artwork other than the Euphonix name and the Eucon symbol without reading glasses, and even then, the

In Insert mode, each display shows the contents of a single plug-in slot on its associated channel; if Channel mode is active, the eight channels show the first eight plug-in slots on the 'attentioned' channel.

writing still looked about the size of the text in a busy dealer ad! It's perfectly readable in bright light, but under typical studio illumination you don't need to suffer from less-than-optimal eyesight to experience problems. There's absolutely no reason for this, as the panel layout would allow a typeface of twice the size without crowding, and the buttons that double up as transport buttons have plenty of space for proper transport-button-style graphics below them.

I've already mentioned that the MC Mix is more intuitive and straightforward to use than Mackie Control, but on the down side, it does miss out a few features that Mackie Control includes, such as the row of function buttons that take you directly to your screensets, marker navigation and a single button for saving the current project. I really missed having a data wheel for scrolling the cursor and, for me, having dual-function transport buttons halfway up the panel is not as immediate as a well-labelled, dedicated transport section. The transport section is the most-used area of my Mackie Control, and as well as starting and stopping playback and recording, it allows me to zoom the vertical and horizontal display resolution, turn the metronome on and off and to enter or exit Loop mode in Logic. In my view, Mackie got this exactly right, although the way the limited number of buttons on the MC Mix is utilised is very effective.

With no dedicated transport controls, the Solo and On buttons for the last four channels double in this role.

"Getting Euphonix quality in such a classy, cost-effective controller, built around the proven Eucon technology, is a big deal and should be recognised as such."

Both are fantastic pieces of kit for the price so I don't want to knock either, but equally both have their strengths and their weaknesses and they may appeal to different types of user. Maybe adding an MC Control to the MC Mix would provide a more complete solution for those who have similar requirements, but with a Mackie Control you get the eight faders plus a master fader, as well as a solid transport section, all in the same unit.

Leaving the limited transport section aside for the moment, even if you're one of those people who doesn't use a remote control surface to its full extent, and I have to confess I'm one, the MC Mix makes it very easy to mix, pan and access plug-ins and to change automation modes, which seems to be what many users spend most of their time doing. There's no visibility problem with the MC Mix's LED readout panels, as they are pin-sharp and high-contrast, even if the rotary controls are so close to them that they're usually obscured by your knuckles when you're making an adjustment. All the motorised faders feel smooth and comfortable, with the touch mode behaving very positively, at least for me. Furthermore, even though my first test system was based around a laptop where nothing had a mains ground (everything ran from power adaptors), the touch faders still worked reliably, which isn't always true of controllers that use touch-sensing technology.

Wrapping Up

The MC Mix is perhaps the most elegant compact controller I've yet seen for use with Logic, which is the DAW I'm most familiar with, and it should be equally welcome to users of the other fully supported DAWs. Before working on this review in my own studio, I spent a little time with the Euphonix product specialist. He had multiple DAWs on his system, and it hopped between them with no problem. Furthermore, the MC Mix is priced very sensibly to be within the reach of home and project studio owners. It doesn't offer quite so much functionality as my Mackie Control, but what it does, it does very professionally and intuitively. It also takes up a gratifyingly small amount of desk real-estate. It is probably fair to say that as the Euphonix MC Mix has a more professional provenance than the more populist Mackie Control, yet is priced within the same ball park, it's not unreasonable that it should have a more streamlined feature set.

The forthcoming MC Control adds four further motorised faders and a flexible touch-screen control section, so having one of each would make the operation even more intuitive, but even as a stand-alone controller the MC Mix does a surprisingly good job. The ability to address multiple DAWs without having to manually change modes or reboot the system is excellent, though having the faders clatter to the bottom every time you switch from a DAW to a word processor is slightly irritating. Perhaps just leaving them where they are when switching to an unsupported application would be a kinder option, both to the user and the faders?

Ultimately, however, such niggles as I have are minor, and even the poor legibility of the legending would only be a problem during the short time it would take to learn what all the buttons do, as there aren't that many of them. Getting Euphonix quality in such a classy, cost-effective controller, built around the proven Eucon technology,x is a big deal and should be recognised as such. I'm really looking forward to checking out the MC Control when it becomes available. SOS

information

- £ £699 including VAT.
- T Euphonix +1 650 855 0400.
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The latest iteration of Tannoy's Precision 8 includes an intriguing on-board DSP room-correction system.

Hugh Robjohns

Tannoy have earned a good reputation for their range of active nearfield monitors with the well-proven Precision series, and the range was augmented last year by the Precision iDP model. It is available in two sizes, one with a six-inch and the other with an eight-inch bass driver. It's the latter model, the Precision 8iDP, that is reviewed here.

The iDP is really an amalgamation of two pre-existing products: the iDP system (borrowed from parent company TC Electronic, and first introduced by Tannoy in their Ellipse monitors); and the original Precision 8 monitor, which employs Tannoy's Wideband and Dual-Concentric technologies. The Precision monitor was first seen as long ago as 2005, with the passive Precision 6 being reviewed in the pages of *SOS* in August of that year.

The iDP acronym stands for 'Interactive Digital Programming', which is a digital technology that provides digital inputs and easy interconnection of multiple monitors, with facilities for accurate level-matching and remote control, as well as accommodating a host of EQ functionality for general tonal tweaking, room alignment and bass management purposes. More of that later; I'll turn first to the more conventional aspects of these monitors.

Tannoy's Precision monitors are very solidly built from MDF, with stable tongue-and-groove joints and a 40mm-thick sculpted front baffle (which helps reduce cabinet-edge diffraction). The overall size is 440 x 272 x 369mm and each cabinet weighs a sturdy 17kg. The baffle supports three drivers on a brushed-aluminium panel, and the cabinet is ported to the rear. Magnetic compensation for the drivers is included as standard (so you can place these monitors near 'legacy' CRT displays without problems).

The main driver is an eight-inch dual-concentric design with paper cones, the claimed advantage of the dual-concentric approach being more accurate stereo imaging and a wider 'sweet spot'. The idea is that the majority of the spectrum is reproduced from the same small source area — the HF emerges from the centre of the LF driver — rather than from two spatially separated drivers, as with most speaker designs.

The upper driver on the baffle is referred to as a wideband 'super-tweeter' and is a one-inch titanium-dome device. Tannoy are not alone in producing wideband monitors, and while the reproduction of audio content to 50kHz may seem unwarranted in many cases, there are valid technical benefits, including more accurate phase and transient responses. It seems incongruous to combine a separate and spatially removed super-tweeter with a system that trumpets the sonic advantages of its dual-concentric design, but in practice the system does retain very stable imaging, so separating the super-tweeter appears not to be detrimental.

The two crossover points are given in the specifications as a DSP-performed crossover at 1.7kHz for the dual-concentric unit, and a passive analogue crossover at 16kHz for the super-tweeter. The complete system boasts an overall response (within a commendable ±2dB margin) of 43Hz to 51kHz, and the maximum continuous SPL is stated as 120dB. Interestingly, both the frequency-response margin and the SPL figures appear to have improved slightly on the original active Precision 8D model.

"The Precision 8iDP is quite a powerful performer, and it can certainly handle big transient peaks without difficulty."

The built-in amplification comprises two 200W Class-D amps, one driving the bass unit and the other driving both the dual-concentric tweeter and the super-tweeter. Class D is a very power-efficient topology, and this design consumes a mere 45W on average. The power supply is a switch-mode type, able to accommodate mains voltages between 100V and 240V AC.

Digital Facilities

The iDP system is in use on a wide range of monitors now, both from Tannoy and TC Electronic's other studio speaker manufacturer, Dynaudio Acoustics (in the Air series). It is a well-thought-out system that is reasonably intuitive to use and immensely powerful, and works with pairs of master and slave speakers, plus optional subwoofer(s). Only the master monitor carries the audio inputs (analogue and digital), with the signal reaching the slave via a 'TC-link' Cat 5 cable. More Cat 5 sockets provide a daisy-chain link to other speakers in larger systems (for example, 2.1 or 5.1 arrays), as well as to the supplied remote control unit.

The Cat 5 networking interface carries more than just audio to the slave speakers — there is also real-time control data, including volume control and mute/solo for each speaker, and static control data used to configure each speaker's DSP facilities for alignment and room-correction purposes. Although it is technically feasible to set up the entire system from the small display on the master unit, it is infinitely easier and faster to use the supplied iDP Soft configuration software. A far more comprehensive 'Installer's' software suite (PC-iP) is also optionally available. This includes much more sophisticated alignment facilities, as well as providing access to the room-correction EQs and delays.

In the Tannoy iDP implementation, the standard master speaker can accept stereo digital audio via an AES3 (XLR) input, with sample rates between 32 and 96 kHz (an option caters for 192kHz, if this is required). The default internal sample rate is 96kHz, and a BNC word-clock input is also provided. A removable panel accepts analogue inputs, again on a pair of XLRs. Four selectable operating levels are provided to set the DSP full-scale point to +9, +15, +21 or +27dBu. The converters are 128x oversampling, dual-bit delta-sigma types, with a dynamic range of 113dB (unweighted, 20kHz bandwidth).

If required, the analogue input card can be swapped out for a dual digital-input card, so that all six channels of a 5.1 signal can be connected to the one master monitor and distributed from that to the other speakers via Cat 5 cables. If multiple master monitors are in use (for example, in a 5.1 array using analogue inputs), then one master monitor has to be configured as the system master controller, and this is done with a simple push-button on the rear panel.

When the system is plugged up and turned on for the first time, the master monitor scans all the connected speakers (identified by their serial numbers) to identify what is connected, and then the user can program the appropriate role of each monitor (front right, rear left, or whatever).

The discreet backlit display on the front of the master monitor is surrounded by four buttons: Exit, Enter, Up and Down. These enable the configuration menu to be navigated and the appropriate system parameters established. The high-level menus are divided into bass management, setup, recall preset, and utility options. Pressing 'Enter' at the desired menu then opens the appropriate submenu in a very intuitive way.

The bass-management menu provides options for crossovers of 50, 80 or 120 Hz, and for no crossover. The setup menu enables the speaker array to be defined with ▶

The main function of the remote is to provide volume control for the entire system (stereo, 2.1 or 5.1). It also enables users to access three configurable reference listening levels and four system presets.

Alternatives

There are plenty of active monitors around, and a growing number are using DSP facilities to enhance sonic performance — such as the Digidesign RM-series monitors we reviewed last year. However, at present only Tannoy and Dynaudio Acoustics can offer the kind of integration and remote control/configuration facilities provided in the iDP system. Consequently, the only speakers that can be compared directly to the Precision 8iDPs are Dynaudio's Air series.

SOUND ON SOUND

Tannoy Precision 8iDP £2585

pros
- Easy room alignment and configuration.
- Superb integration of complex monitoring systems.
- The iDP Remote panel is very user-friendly.
- Accurate imaging and wide sweet spot.
- Good resolution at normal listening levels.

cons
- Biased towards a slightly forward sound.
- Sound tends to harden at very high levels.
- The technology makes a simple stereo setup expensive.

summary

Tannoy's Precision monitor is available in three forms: the basic passive model, the 8D active, and now the 8iDP active with DSP control. The iDP technology has been proven on more up-market Tannoy models and brings an impressive level of control and configuration to the Precision range, which is particularly effective in 2.1 and 5.1 setups.

modes for analogue, digital or the 192kHz digital input options, and in stereo or 5.1 surround configurations. The necessary arrangement of monitors is specific for each mode, but the manual makes everything very clear, with good diagrams and descriptions of how everything has to be wired up — although, once you grasp the basic premise, it's really not rocket science.

Once the system knows what kind of setup it is working as, you can allocate the required channel function of each speaker. Further options allow you to select the external word-clock reference, set the analogue input levels and enter the calibration mode. This last setting allows the position of the speaker to be defined (wall, corner, console, and so on) and the DSP to generate pink noise for room measurements to be made. The calibration level can then be set, along with simple ±6dB bass and treble equalisation.

The remaining menu options allow specific configurations to be stored or recalled from memory (with 15 factory and 15 user locations), and for the standby and power-save time durations to be established. The monitors will go into standby mode after between 15 and 90 minutes (although there is also an 'off' mode), and power-save after between one and five hours.

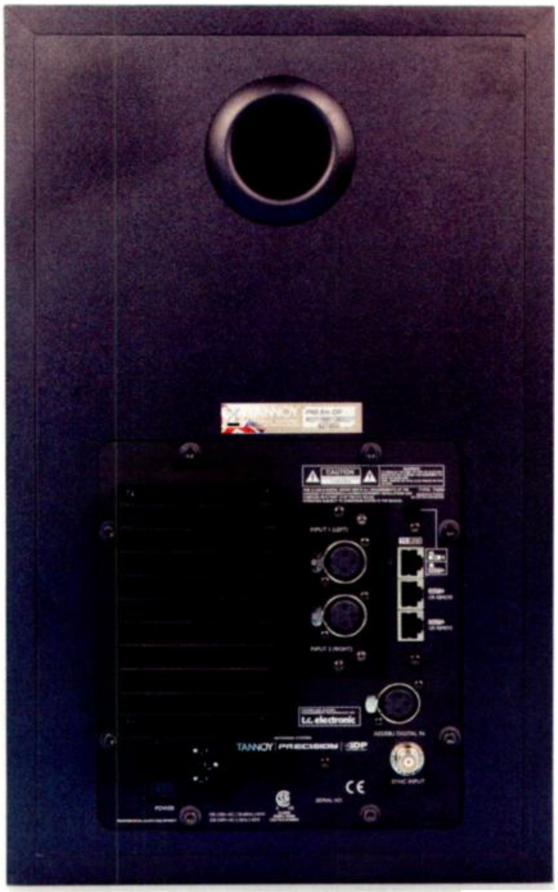
The monitors are supplied with the iDP Remote panel, which is a neat handheld or desktop panel (see photo on previous page) with 12 push-buttons and a rotary control. It connects via any spare RJ45 socket on any speaker, using another standard TC-link cable. The main function is to provide volume control for the entire system (stereo, 2.1 or 5.1) and, to make life easier, the top three buttons provide access to a trio of configurable reference listening levels, while the second row of buttons accesses four system presets.

At the bottom of the panel, six more buttons serve as mute or solo buttons for each speaker in a 5.1 channel configuration. Although arguably a luxury in a simple stereo system, this iDP Remote really comes into its own in a surround setup.

Software Configuration

The supplied iDP Soft is a configuration editor that makes setting up an iDP array much easier and quicker than all that front-panel button pressing! I ran it on a PC laptop, but it will also work on a Mac running OS X. A serial-port-to-TC-link cable is provided as standard, although a USB serial-port adaptor is also available.

The graphical interface is, once again, very intuitive and easy to use, with just three sub-pages from the main page. The latter allows the volume of the system to be controlled, the reference listening level buttons to be programmed, the presets to be edited, and each individual speaker's parameters to be accessed (through separate sub-pages). The other main sub-pages cover system configuration, speaker calibration and network setup.



Audio is fed into one (master) speaker. This is carried to others via Cat 5 cables into the RJ45 connectors on the rear of each slave speaker. The same system also relays real-time control data to each speaker.

In conjunction with a sound-level meter positioned at the listening position, I was able to configure, control and align the level of each speaker in a stereo pair within a few minutes, and a full 5.1 system wouldn't take much longer. The iDP Soft editor also made it easy to experiment with the overall EQ, and I ended up turning the treble down by 1dB to tame a perceived tendency to brightness in my room.

Listening

The Precision 8 has always been a good-sounding monitor, with a reasonably neutral and detailed sound, possibly tending a little towards a bright or forward character. The bottom end is smooth and well controlled, but I found it tended to sound a little shy on some material. After some extended listening I came to the conclusion that, in fact, the bass is quite accurate and well extended for a box of this size, but the slightly bright tonal balance tends to give a misleading impression. Trimming the built-in equalisation by -1dB to tame the upper end fractionally seemed to produce a more balanced sound (to my ears), and removed my earlier concerns about a weak bass response. In my case, the monitors were also sited well away from walls in a fairly large room — and in a smaller room with closer wall proximity the bass would tend to fill out more anyway.

However, a monitor of this size will never be able to deliver the 'liver quiver' experience, so for those who like having their internal organs gently massaged by low bass it would be sensible to budget for one of the matching subwoofers — the TS212 iDP or the TS112 iDP — which can be integrated very well, thanks to the DSP alignment facilities.

The Precision 8iDP is quite a powerful performer, and it can certainly handle big transient peaks without difficulty. At normal, modest listening levels the resolution is very good. Cranking them up to big-room levels tended to undo all the good work, though, as the sound started to get messy and more veiled the harder I pushed it. But, of course, this is unlikely to be an issue in the average home studio, with the speakers working in their intended nearfield role.

Verdict

The Precision 8iDP is based on a well-established and capable performer, and the application of the iDP technology extends its flexibility further. In a simple stereo setup, the additional cost of the iDP version over the standard active 8D model probably outweighs the benefits. However, in a more complex 2.1 system — and especially in a 5.1 array — the iDP technology really comes into its own, and the more so if your room acoustics are sufficiently good to warrant a full digital response alignment. SOS

information

£ £2585 (pair); 2.1 system with the TS112 subwoofer, £4342. Prices include VAT.
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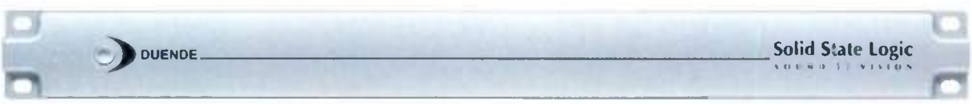

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Secrets Of The Mix Engineers: Renaud Letang

Aided by its memorable video and starring role in an iPod ad, Feist's '1234' has been a refreshingly different worldwide hit. Renaud Letang was behind the (vintage) desk during the recording and mixing sessions.

Paul Tingen

"What is cool," muses Renaud Letang cheerfully, "is that I'm considered a bit of a dinosaur here in France, but that I'm new and fresh for the British and Americans. I'm not old, but I've worked in the French studio industry for 18 years and I'm established here as a big-name producer. But outside of France people are just getting to know my name, so it's a really different experience for me. It's funny."

Renaud Letang was speaking from Studio Ferber in Paris, where he works "seven to eight months per year". Born in 1970 in Iran, of French parents, Letang began his studio career working as a tape-op in a small studio in Paris at the age of 18. After six months, in 1989, he moved over to France's leading studio at the time, Guillaume Tell. He worked there for three years, and the seeds for his international outlook were sown during that time when Paris, and particularly Guillaume Tell, became a fashionable place to record for Anglo-Saxon stars. A presumably wide-eyed Letang found himself in the same room as the likes of Phil Ramone, Prince, Sting, Peter Gabriel, Paul Simon, and others. "It was crazy, crazy, crazy," he says, recalling the unrelenting influx of foreign stars.

In 1990, concurrently with his apprenticeship at Guillaume Tell, Letang began working as Jean Michel Jarre's engineer, both live and in the studio. He remained with the electronic musician until 1999. After leaving Guillaume Tell in 1992, Letang quickly spread his wings as an independent engineer and also increasingly as a mixer and a producer. His name can today be found on whole swathes of French hit recordings in all kinds of different genres, ranging from French chansons to world music, electronica and hip-hop. He's also built up a considerable overseas pedigree, with artists such as Peaches, Beck, Jamie Lidell, Mocky, Gonzales and Feist.

Gonzales, *née* Jason Beck, hails from Canada, but moved to Europe and currently lives in Paris, where he set up a production duo with Letang called VV. It was through Gonzales that Letang met Canadians Peaches, Mocky and Feist. "Gonzales and I co-produced Feist's *Let It Die* album and also got her

The success of '1234' was boosted by its memorable video.

The Neve A646 desk at Studio La Frette, where most of the tracking for *The Reminder* took place.

signed to a record company," recalls the Frenchman.

Gonzales and Letang went on to co-produce the follow-up *The Reminder*, which featured the hit '1234'. Helped by its appearance in an Apple iPod Nano commercial, the song, and subsequently the album, has been a hit in many countries, and resulted in four Grammy Award nominations for Feist.

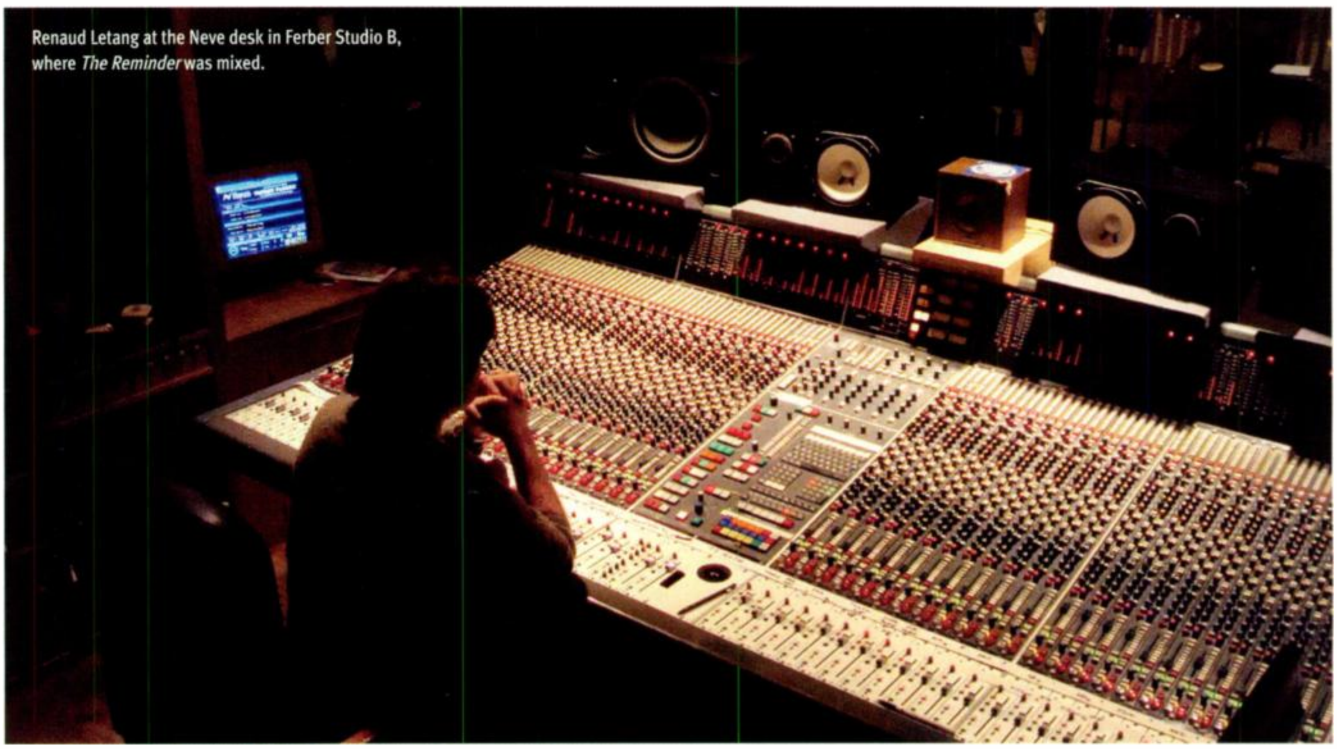
The Numbers Game

Letang relates how he applied his talents on the recording and mix of '1234'. Most of the backing tracks for *The Reminder* were recorded at Studio La Frette, just outside Paris, after which the project moved to Letang's own recording and editing room at Ferber Studio D for 'clean-up'. The company then travelled to Canada, where overdubs took place and a few more songs were recorded at the Woodshed Studio in Toronto. Following this, everyone returned to Letang's room in Ferber for 'post-production' editing and further clean-up, as well as the recording of the song 'I Feel It All'. Finally, Letang mixed the album at Ferber Studio B, his favourite mix room.

One of the common denominators in all these studios is Neve desks: a 1972 36-channel Neve A646 in La Frette, a 1978 32-channel Neve 8108 in Renaud's own studio, a Neve 8014 desk and Neve 1073 sidecars at the Woodshed, and a 48-channel Neve V series at Ferber Studio B. It turns out that Letang is quite a fan of vintage equipment in general, and only very occasionally uses plug-ins. "I use a combination of analogue and digital," he explains. "I use Pro Tools for what it's good for, recording and editing. I'm not going to use a Focusrite plug-in if I have a real one standing next to me. In fact, I used no plug-ins at all for the recording and mix of *The Reminder*.

"I've never wanted to use the old Pro Tools thing, the 888, because it didn't sound good enough to my ears. If I got a project to mix in on Pro Tools, I always transferred it to 24-track analogue. But since Pro Tools has gone HD, I like it. HD sounds really transparent, there's no colour. You get back what you put in. And I prefer the Pro Tools A-D converters to the Apogee ones, because I find that the Apogees do weird things to the mid-range. It's cool for a stereo mix, but when you record with them and use it on many tracks the middle gets too rich.

"I like using Pro Tools HD because I record with old desks and old microphones, so everything already sounds fat and warm. I get people coming in who have mixed in Pro Tools, and they say 'I don't know, but there's something strange about the sound, but we don't know what.' It's always the same thing. ▶



Renaud Letang at the Neve desk in Ferber Studio B, where *The Reminder* was mixed.

And when I then mix their project on a real desk, with analogue EQ and everything, the difference is crazy."

The Vintage Approach

Studios La Frette was set up in the late '60s by French production icon Eddie Barclay, also known as the founder of Barclay Records. Located in the Parisian suburbs in a classical-looking 100-year-old house with high ceilings and wooden floors, it continues to be a haven for lovers of vintage equipment and natural acoustics. Feist apparently chose to record there because of the natural ambience, and to further this feeling the team recorded in the parlour and living rooms. "Feist's band had already played '1234' live," explained Letang, "so they had the beginnings of an arrangement. We recorded the bass, drums, piano, guitar, and a guide vocal live at La Frette, using a click track, and added the brass there as well."

In the process of recording *The Reminder*, Letang used everything at his disposal at La Frette in terms of vintage gear and acoustics. "I had many ambient microphones, and I used many of the vintage microphones at La Frette. There were a Neumann M49 and U67 for the drums, a Coles ribbon for overheads and guitar cabinets, a U47 FET and an [*Electro-Voice*] RE20 on the bass. I also added a lot of dynamic mics to double things up; for example, I also used SM57s on the guitar cabinets. I could then later choose which sound I liked best, or mix the two sounds together to change the colour of the instruments. In general I used a lot of ambient microphones, and just did really simple EQ on the Neve desk, just low cut, things like that. I used the Neve desk preamps, and a Millennia preamp if I wanted a cleaner sound. I also had four Fairchild 670 compressors in the signal chains.

"Many people have asked me how I did Feist's vocal tracks on the album. We had two setups for the whole album. I had a Neumann U67 and an SM57 both going into a Vox guitar amp, on which we EQ'ed and added reverb. The Vox was miked up by an SM57 and a U87 and then went into the old Neve A646. This gave us what we called the 'dirty' vocal. The other setup was just one U67 going into the Neve preamp in Canada, which was similar to the Neve A646. In the end we used the 'dirty' vocal on all tracks, apart from '1234', which was done with the 'clean' signal chain. The reason was that the banjo on '1234' had the same mid-range frequencies as the 'dirty' vocal, so they got in each other's way. We then went to Canada, where we overdubbed strings, banjo, a new guitar track, and the vocal to '1234'. After that we added some keyboards and small things, and edited and cleaned up the track in my studio D at Ferber."

'1234'
Written by Sally Seltmann, Leslie Feist
Produced by Gonzales, Renaud Letang and Feist, with Ben Mink

"One important aspect of the mix was to take out many things," says Renaud Letang. "There were a lot of things playing — for example, a whole brass section — and it had a lot of energy, but it made the track too full. There was too much cream on the cake. It needed to be much lighter. It's a cool song, with a positive and happy feel. Related to this was the fact that I wanted to create space around the acoustic music and give the track a modern shape. I'm not sure how to say this in English. I wanted the vocals and drums to be really large, like hip-hop size, but I couldn't make the whole song fat-sounding, because it would be too much. The song had to be *légère* — light — and I achieved this by using the acoustic of the room microphones a lot. So I was trying to give the instruments a normal size, ie. big vocal and big drums, and the whole track a normal, acoustic and light sound.

"The Neve V-series desk I mix on at Ferber has only 48 channels, and I like to come out of Pro Tools with a maximum of 32 tracks, so I also have channels on the desk for automation and effect returns. Pro Tools Sessions usually consist of many more than 32 tracks, so part of my mix preparation involves pre-mixing. I never pre-mix the drums, I always like to have them separate. The only exception is the bass drum: if it's recorded with more than one microphone I'll submix to one track. The same with bass and the main guitars. If they are recorded with more than one microphone, I'll often pre-mix these as well.

"I find that if you apply EQ or compression to individual string or horn tracks, you lose the impression of the horns or the strings as a unified section, and it becomes really easy to lose the overall balance. But when you pre-mix and then EQ and compress the whole section, the sound remains much more natural. The ability to do pre-mixes is one of the reasons why I love working with Pro Tools. For '1234' I pre-mixed the backing vocals, the horns, the strings, and the banjo, which was recorded with two or three microphones.

"When I start with the proper mix on the desk, my method is very traditional. I begin with the drums, and then I add the bass, and then the other main instruments — the piano and banjo on '1234' — and then I add other supporting instruments, like the electric guitar in the chorus. After that I add in the vocals, to see what's going on, and to see whether the vocals work with the backing tracks. If not I will change the balance. Sometimes when you mix, a snare will sound cool in the backing track, but with the vocals it may need a different dynamic, so you need to start doing rides. After that, when I have the main instruments and vocals in place, I'll add things like the horns and the strings and the backing vocals. One important aspect of this way of working is that I add all the tracks in, rather than work on them in isolation. If you work on things in isolation, they make no sense. A bass drum alone is nothing. A vocal alone is nothing."

Drums: API & Neve EQ, Neve compressor

"As I said, I start my mix working on the drums — but I won't spend four hours just on the bass drum! On '1234' I used an API 550 EQ on the bass drum, pushing 100Hz and 5kHz. I used another 550a on the snare, pushing 1.5k and 5k. That's it for inserts. The rest was just using EQ and compression on the Neve V-series board. When you compress

Letang's collection of vintage gear also includes a number of keyboards.

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the ambient microphones and add EQ to them, you can make the whole drum section sound really compact and alive and together. So I added quite a bit of compression to the ambient microphones, but unless I have problems I don't like to apply much compression on the drums themselves. I prefer to do the main compression on the drums during final mixdown. It's the same as what I told you earlier about EQ'ing individual microphones in a horn or a string section: sometimes you get a weird sound. I prefer to apply EQ and/or compression to the whole track during mixdown — not heavily, just enough to tighten the dynamics of the drums."

Bass, guitars & banjo: Pultec EQ, Urei 1176, TC M5000, API 550 EQ, EMT 140, Manley Variable Mu

"'1234' is really acoustic, so I'm using normal stuff, like reverb, EQ, compression, to get a really natural sound. For the acoustic guitar I used a blackface Urei 1176 and the desk, just a little compression and EQ. I also added a very minimal amount of chorus from a TC M5000, to spread it out a little bit in stereo, because the guitar was in mono. For the banjo I used two API 550 EQs and a Manley Variable Mu tube stereo compressor. After that I put a short delay and reverb on the banjo, the latter from an EMT 140 plate, with a short decay time. The banjo was recorded with natural acoustics, so I mixed these in as well.

"Do I like the API EQ? Yes, I do, though I have more Pultec equipment than API. I have six API 550a EQs, but don't use them for everything. I find that each equaliser has certain frequencies at which it works best. For example, a Neve sounds crazy at 500Hz and also really crazy at 300Hz, really musical and really warm. But if you push these frequencies with an SSL EQ it sounds bad, horrible. On the other hand, if you push 80Hz and 100Hz on an SSL it sounds really clear and dynamic and nice. With the API you can push 400Hz, 1.5kHz, 7kHz, and it sounds great. What I like is that the steps are discrete. The API makes a statement, because you push +2dB, or +4dB, or +6dB. When you work on a stereo channel, for example on a submix of the horns or the backing vocals, you can boost left and right exactly the same and this means that everything is still in phase. With non-discrete EQs you never know exactly what you're doing."

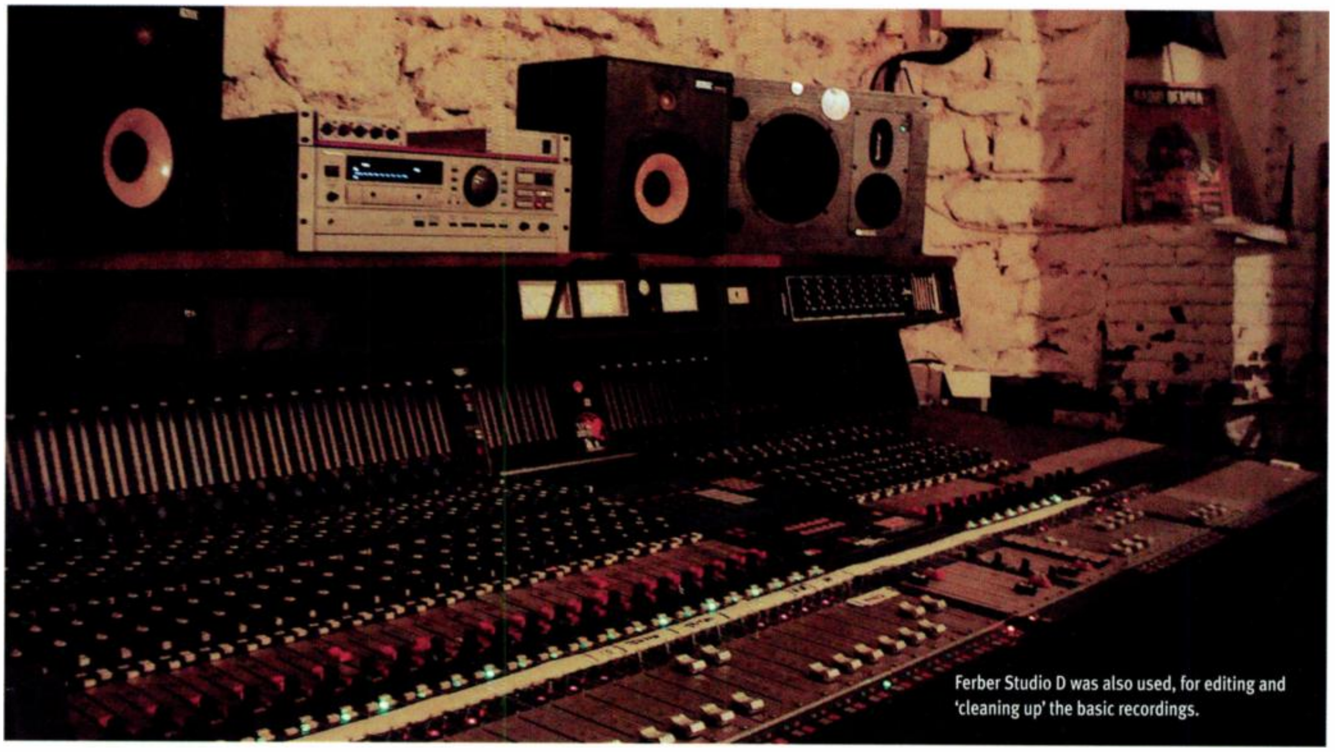
Pultec and API EQs figure prominently in the gear list...

Keyboards: Neve compression & EQ, Yamaha D1500

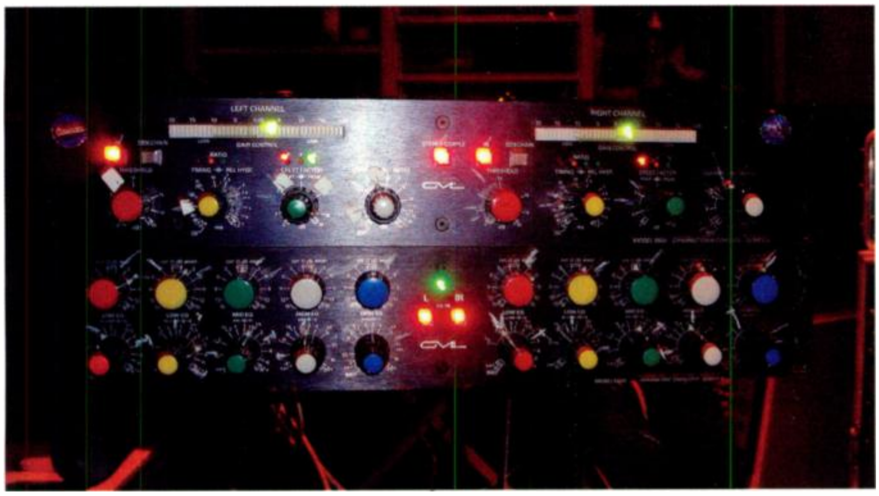
"I used compression and EQ on the desk, nothing special. The piano is playing really hard, it's creating a lot of energy, so I EQ'ed the piano to make it really bright and really pushing the track, really honky-tonk. There's also a Mellotron, which is going through a Yamaha D1500 delay, to give it some more space. The D1500 was one of the first delays, and it still sounds really good. The delay was compressed and filtered, taking out high and low end, to give the delay a different space and make it less normal."

Vocals: Urei 1176, Pultec EQP1A, Dbx 902, Korg SDD 3000, AKG reverb

"My lead vocal chain during the mix was a Urei blackface 1176 going into two Pultec EQs, one EQH2 for low and high end, ie. 100-330Hz and 5k to 12k, the other Pultec



Ferber Studio D was also used, for editing and 'cleaning up' the basic recordings.



The GML stereo compressor and EQ were used on the master bus.

MEQ5 for the mids, 200Hz-2kHz and 200Hz-8kHz. With these two Pultecs I had everything: I could cut or boost high, middle, or low end. After the Pultecs the vocals went through a Dbx 902 de-esser, and then back into the Neve insert. I also had a small delay on the vocals, using an old Korg SDD3000, and for reverb I used an old AKG reverb. It has no high end, but it sounds warm and really musical and hyper-chic."

Brass, strings & backing vocals: Neve compression & EQ, EMT 140, Lexicon 480, Avalon EQ

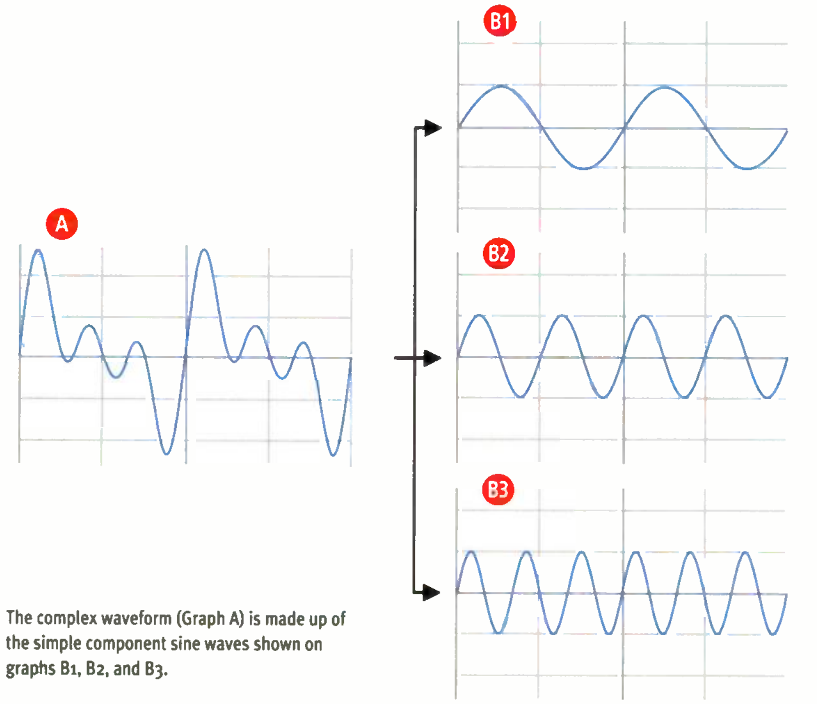
"I just had desk compression and EQ on the brass, and then the same EMT 140 as I had on the banjo, with a short reverb, though I changed it to a long reverb for the parts of the song where it's really alive and it sounds like a lot of people playing. There was no insert on the brass. As for the strings, I wanted to give the impression of a film soundtrack, with the strings coming from nowhere and with super high-quality effects. So I used a Lexicon 480 reverb. The 480 makes everything sound produced, but in a cheesy way. For this kind of effect it's really good. The backing vocals were treated with an Avalon stereo EQ and I compressed them on the desk.

"Finally, I put a George Massenburg 8200 EQ and a George Massenburg 8900 compressor over the stereo mix. I finalise the colour of the mix with the EQ and the compressor. I don't use the compressor for the dynamics of the track, but more to finalise the groove of the track. When you compress you change the groove a little bit. By adjusting the attack and the release times you can make a track more or less bouncy. So I finalise the colour and the texture with the EQ and the groove with the compressor.

"I mixed down to two formats: an old Ampex half-inch analogue two-track and the other mix went through an Universal A-D converter back into Pro Tools. So we had really good quality digital and analogue. During mastering we choose which one is the best. For '1234' we used the digital version. I already use so much vintage equipment, and everything is already so warm and the high frequencies are so smooth, that it is sometimes better to use the digital mix, to make things more dynamic." SOS way sound reflects from solid surfaces such as walls. For example, if you close-mic an electric guitar cabinet, a significant minority of the sound picked up will actually be reflections from the floor. If the distance from the cabinet's speaker cone is only six inches, and the floor is a foot below the mic, the direct and reflected sounds of the cone will meet at the mic capsule with around 1.5ms delay between them. In theory this will give a comb-filtering effect with total phase cancellation at around 300Hz, 900Hz, 1.5kHz, 2.1kHz, and so on.

But it doesn't work out exactly like this, for a number of reasons. For a start, the reflected sound will almost certainly have a slightly different timbre by virtue of the sound-absorption characteristics of the floor. Sonic reflections will also arrive at the mic capsule off-axis, which will alter their frequency balance. Then there's the contribution made by reflections from other nearby surfaces, which further complicate the frequency-response anomalies. However, even though you don't get a perfect comb-filtering effect in practice, reflections from the floor are still an important contributor to the sound of a close-miked cab, and many producers experiment with lifting and angling cab in relation to the floor for this reason. If you want to hear this for yourself, surf over to my article on guitar recording at www.soundonsound.com/sos/aug07/articles/guitaramprecording.htm and have a listen to the audio files in the 'Room & Positioning' box, which demonstrate how much difference moving the cab relative to room boundaries can make.

Of course, phase cancellation between direct and reflected sound can cause problems when recording any instrument, and with acoustic instruments it becomes, if anything, more troublesome — listeners tend to have a less concrete expectation of how an electric guitar should sound, so phase cancellation can be used to shape the tone to taste, whereas with acoustic instruments the listener tends to have clearer expectations, so the tonal effects of comb-filtering are usually less acceptable. Fortunately, it's not too difficult to avoid problems like this, as long as you try to keep performers and microphones at least a few feet from room boundaries and other large reflective surfaces. This can be a bit trickier where space is limited, in which case it can also help to use soft furnishings or acoustic foam to intercept the worst of the room reflections. Our extensive DIY acoustics feature in *SOS* December 2007 (www.soundonsound.com/sos/dec07/articles/acoustics.htm) has lots of useful advice if you find yourself in this situation.



The complex waveform (Graph A) is made up of the simple component sine waves shown on graphs B1, B2, and B3.

Another thing to try in smaller rooms is boundary mics, because their design gets around the phase-cancellation problems associated with whichever surface they're mounted on.

Multi-miking Single Instruments

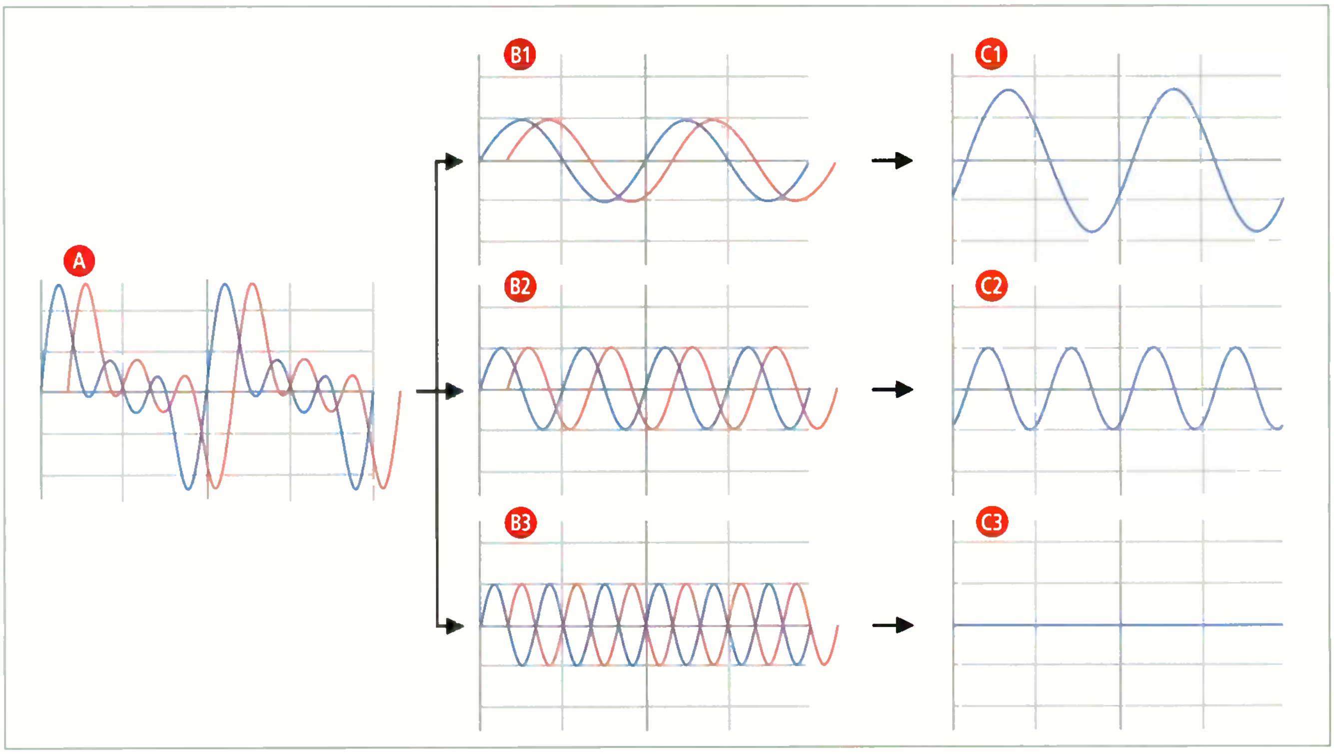
If you use more than one mic to record a single instrument, the simplest way to minimise the effects of phase cancellation is to get the mic capsules physically as close together as possible — what is often called coincident miking. However, given that it only takes about an 8mm difference between the capsule positions to produce a deep phase-cancellation notch at 20kHz, it pays to line up the mic capsules quite carefully — a process a lot of people refer to as 'matching the phase of the mics' or 'matching the mics for phase'. In practice, this is best done by ear, because it's often difficult to tell exactly where a mic's capsule is without taking it apart. One handy trick for doing this is to invert the polarity of one of the mics and then adjust the positions of the mics while the instrument plays, to achieve the lowest combined level. Once the polarity is returned to normal, the mic signals should combine with the minimum of phase cancellation.

Clearly, spaced mic positions that are equidistant from a sound source will also capture its direct sound without time delay, but some phase cancellation of reflected sounds will inevitably occur, so some adjustment of mic positions can prove useful here to find a suitable ambient timbre. However, it's worth bearing in mind that if you mic up any source of sound vibrations from in front and from behind simultaneously, the polarity of the signal from the rear mic may be inverted. This is very common when, for example, miking the front and back of an open-backed guitar cabinet or the top and bottom of a snare

Combining Mics & DIs

A number of instruments are routinely recorded both acoustically and electrically, simultaneously. For example, electric bass is often recorded via both a DI box and a miked amp, while an acoustic guitar's piezo pickup system might be recorded alongside the signal from a condenser mic placed in front of the instrument, in some situations. In such cases, the waveform of the DI or pickup recording will precede the miked signal because of the time it takes for sound to travel from the cabinet speaker or instrument to the mic. The resultant phase cancellation can easily wreck the recording.

A quick fix for this is to invert the polarity of one of the signals, and see if this provides a more usable tone. It's almost as easy to tweak the miking distance for more options if neither polarity setting works out. A better method, though, is to re-align the two signals in some way by delaying the DI or pickup channel, either using an effect unit during recording, or by shifting one of the recorded tracks after the fact using your sequencer's audio-editing tools.



Sine waves phase-cancel when delayed and undelayed versions of the same waveform in Graph A are mixed together. The red traces show the delayed versions of each waveform in graphs A, B1, B2 and B3. Graphs C1, C2 and C3 show the result of combining the different sine-wave components: the waves in Graph B1 only slightly phase-cancel, producing a combined sine wave of nearly twice the level of each of the individual sine waves; the waves in Graph B2 phase-cancel more heavily, producing a combined sine wave of only the same level as each of the individual sine waves; and the waves in Graph B3 are 180 degrees out of phase with each other, so completely phase-cancel.

drum, and you'll usually find that the two mics will combine best (especially in terms of the bass response) if you compensate by using a phase-inversion switch on one of the two channels while recording.

Despite the potential for phase cancellation, many producers nevertheless record instruments with two (or more) mics at different distances from the sound source. Where the two mics are comparatively close to each other, this provides some creative control over the sound, because tweaking the distance between them subtly shifts the frequencies at which the comb-filtering occurs. Inverting the polarity of one of the mics yields another whole set of timbres, switching the frequencies at which the sine-wave components in the two mic signals cancel and reinforce, so the potential for tonal adjustment via multi-miking is enormous.

The severity of the comb-filtering is usually reduced a little here, though, because two completely different models of ▶

you're unconcerned about mono compatibility in principle, you could still come unstuck if you find that you want to narrow the stereo image of a spaced-pair recording at the mix — because panning the mic signals anything other than hard left and right will combine them to some extent, giving you a tonal change alongside the image width adjustment.

Where mono compatibilty is paramount, coincident stereo techniques are clearly the most suitable choice. However, a lot of engineers find that the lack of any time-difference information in such recordings makes them sound rather clinical and emotionally uninvolving, so they have come up with a number of ways of maximising the mono compatibility of spaced-pair recordings instead.

Although it might appear to be a good idea to reverse the polarity of one of the mics to achieve a better mono sound, this creates a very weird effect on the stereo recording, so this is of little use here. The simplest thing you can do, therefore, is first set up the spaced stereo pair to your satisfaction, and then quickly switch to mono monitoring and subtly adjust the distance between the mics to massage the tonal balance of the mono sound. Small mic movements will make quite large differences to the mono mix, but without making a huge difference to the stereo sound. The goal is to find mic positions that keep the tonality as consistent as possible as you flip between mono and stereo monitoring.

Another trick is to position the spaced mics equidistant from the most important elements of the ensemble (putting them exactly in the middle of the stereo picture), ensuring that they transfer to mono as well as possible. This principle is used by some engineers when setting up drum overhead mics, and is particularly associated with über-producer Glyn Johns — the idea being to place the mics equidistant from the snare drum, kick drum, or both.

The famous Decca Tree technique is an example of a different tactic: spacing the mic pair more widely and then setting up another mic between them, panned centrally. (For more on this, have a look at the second part of Hugh Robjohns' stereo mic techniques article back in *SOS* March 1997 (www.soundonsound.com/sos/1997_articles/mar97/stereomictechs2.html).

Phase Cancellation When Layering

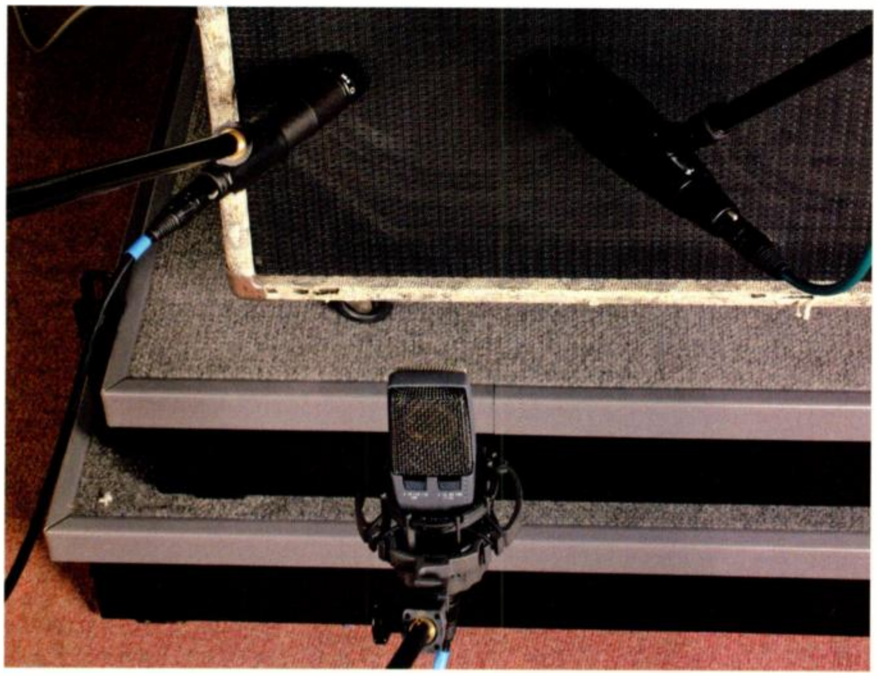
Throughout the main body of this article I've concentrated mostly on the phase-cancellation problems that occur when two versions of the pretty much the same sound are mixed with a delay between them. However, phase-cancellation can also occur to some extent between the sine-wave components of any two similar sounds that are layered together. When layering human performances on real instruments, this is not a problem; on the contrary, when you're layering up vocal or guitar overdubs the fluctuating phase-cancellations between the different, naturally varying tracks is all part of the appeal. However, if you try to layer sampled or synthesized sounds together within a programmed track, you can encounter all sorts of pitfalls.

The first common problem occurs when you're feeding the same MIDI notes to two different sample-based instruments, either hardware or software. The sounds you select for the two different parts will play much more accurately together than would human performers, and the notes will be more uniform in tone than those of real instruments. Even if there is ostensibly no delay between the onset of the two different sounds, the phase relationships of their different sine-wave components may still be very different, and the resultant phase-cancellations won't vary in a natural way, as they would with layered live performances. This often gives a kind of 'hollowness' to the layered patch, which you'll rarely find appealing.

However, I've found that it's not too tricky to avoid unpleasant combinations in these situations as long as you steer clear of pairs of sounds which are both percussive and similar-sounding (say two different pianos) — and with things like evolving pads you can get away with all sorts of combined sounds without difficulties. You should also take care when layering sounds with prominent low frequencies (such as basses and kick drums), because it can really suck the power out of the track if the combination cancels out even a single powerful low-frequency sine-wave component. (In fact, this is as much a concern with live instruments, and accounts for the comparative rarity of layered bass sounds on record.)

The problem can be compounded if you also get a delay between the two instrument layers. You might ask 'How can this occur if I'm sending each one the same MIDI data?' The most common way it can happen is if you're running hardware sound-modules alongside a computer sequencer. For a start, the computer's internal instruments don't have to deal with the output MIDI latency of your soundcard, but there's also the fact that hardware sound modules suffer from latency too, and each one will probably have a different latency value. If you're then monitoring your hardware sound modules' outputs through spare inputs on your audio interface, you'll have to contend with further latency from the soundcard.

The biggest difficulties arise, though, when you're layering bass or kick-drum sounds and there's a delay between the two layers that varies from moment to moment. The result is a sound that is practically unmixable: the timbre of the instrument varies from note to note, and some notes completely lose their power through cancellation of important low frequencies. You'd have thought that this would be a fairly uncommon problem, but I've encountered it twice in recent Mix Rescue projects, which leads me to suspect that it might actually be quite widespread. One workaround is to use only one of the layers to supply the low frequencies, by high-pass filtering the other, but in my experience this tends to be more successful with bass sounds than with kick drums. For the latter, it makes much more sense to layer the two kick samples within a single sampler instrument (hardware or software), as this is more likely to ensure that they always trigger at exactly the same time, thereby keeping phase cancellation between the sounds, and their composite timbre, consistent.



Phase cancellation isn't necessarily a bad thing. The 'phase EQ' technique uses three mics positioned to form a triangle. The faders on the desk (or in the DAW) can then be raised or lowered for each signal, altering the phase relationship between the signals — and this can be a less intrusive alternative to conventional EQ.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By relying on the central mic for the majority of the recording and the left and right mics for the stereo effect, when you sum the recording to mono the sound from the central mic remains unchanged (it already was mono!), and the phase cancellation between the spaced mics makes less of an impression because they're lower in level. The down side with such techniques, though, is that the phase cancellation between the central mic and each of the other mics can make finding suitable positions for the stereo rig a bit trickier — there's no such thing as a free lunch!

A final worthwhile option is trying one of the techniques that combine time- and level-difference information by using spaced directional mics — setups such as the ORTF or NOS standards, for example. Although the sound of off-centre instruments will still arrive at the two mics at different times, the levels of the two signals will be differentiated by the mic polar patterns, which will help reduce the audibility of the comb filtering. However, that's no excuse not to still check the sound in mono while recording.

Multi-miking Ensembles

Where phase cancellation can really mess things up is when you start miking individual instruments in an ensemble separately. You'd ideally like each instrument mic to pick up just the instrument it is pointing at, but in reality it will pick up spill from all of the instruments around it. The sound of each instrument through its own mic will, to some extent, phase-cancel with its spill on every other mic, so that moving any single mic has the potential to change the sound of the other instruments in an incredibly complex way.

Phase can be a problem with stereo miking of sources that aren't stationary. An acoustic guitar player, for example, will usually move the guitar at least a little during performance — which is why closely placed coincident pairs tend to be preferred over spaced stereo techniques.

There are some engineers who (by dint of golden ears, years of experience, and a pact with a certain horned gentleman) have acquired the skill of managing this mass of phase cancellations such that they can actively use spill to enhance and 'glue together' large-scale multi-mic recordings. Such luminaries are often happy to use primarily omni mics, despite the increased levels of spill, for this reason. For the rest of us mere mortals, the key to success on this kind of recording session is reducing the levels of spill as much as is feasibly possible, thereby minimising the audible effects of the comb filtering.

There's a lot that can be done in this regard simply by careful positioning of the mics and ensemble instruments, and in general it makes sense to keep each mic closer to the instrument it is covering than to sources of spill. This idea is often encapsulated as the '3:1 rule' — namely, in order to keep spill manageably low, the distance between mics on different instruments should be at least three times the distance between each mic and the instrument it is supposed to be covering.

Although the 3:1 rule is a handy guide for some engineers, I don't personally find it to be very useful, because it doesn't take into account the differences in volume between different instruments, any acoustic factors in the room, or the effects of microphone polar patterns. Indeed, judicious baffling of instruments and some attention to room treatment are just as important for managing spill, in my experience, as mic selection and positioning. A more sensible route, in my opinion, is to work in terms of the ratio between the level of the close mic for each instrument and the total spill level picked up for that instrument by all the other mics. You can easily test this by asking each miked musician or group of musicians to play in turn with all the mics open — if muting that instrument's close mic reduces the overall mix level by around 9dB you should be pretty well in the clear.

Clearly, directional mics can make the task of reducing spill levels easier, because rejection nulls can be aimed towards neighbouring instruments — the deep side nulls of figure-of-eight mics can really come

Variable Phase Devices

The comb filtering that occurs due to phase cancellation can often be a desirable effect, and while a simple delay can be used to 'correct' time-of-arrival distances between (for example) signals from two different microphones recording the same source, it doesn't give you control over which frequencies are being aligned.

Hardware units such as the Neve Portico 5016, the Little Labs IBP, the Audient Mico and Radial's Phazer all enable you to make variable phase adjustments. They use a phase-shifting circuit, which enables the user basically to decide which frequencies add constructively and which still cancel — allowing the subjective 'focus' of a sound to be optimised, or the sound quality from the combination of DI and miked signals to be tweaked in a satisfying way. Similar tonal shaping options are offered by software such as Tritone Digital's Phase Tone, and can be used to tweak the sound during the mixing stage — though, of course, it is better to get the sound right during recording.

Some devices, including the Neve Portico 5016, the Radial Phazer and Audient Mico, and software like Tritone Digital's Phase Tone, allow you to decide which frequencies add constructively to your sound and which phase-cancel.

into their own here. However, all directional mics (especially less expensive models) colour off-axis sound to some extent, which on occasion causes more problems than the directivity solves, because a lesser amount of nasty-sounding spill can prove more difficult to deal with (even though it causes less phase cancellation) than a greater amount of comparatively uncoloured spill. This principle is at the heart of a useful technique (described by Mike Stavrou in his fascinating book *Mixing With Your Mind*), that involves miking up each instrument to get it sounding the best you can on its own, and then repositioning that instrument and its mic together (without changing their relative positions) to achieve a balanced tonality for any spill.

Pretty much whatever you do, though, you'll still get spill of one kind or another on most of the mics. This means that even if you get each close mic sounding fine for the instrument it's nominally assigned to, it's likely that some of these lovingly-finessed mic signals will suffer from adverse phase cancellation when all the mics are mixed together. In this kind of situation, the first thing to try is reversing the polarity of various different combinations of mics, perhaps starting with the most compromised-sounding close mics. In this situation there are no 'correct' polarity settings, so you should look for the combination that offers the best tonal balance across all the instruments of the ensemble. It's vital to remember, though, that inverting the polarity of any single mic will not just change the sound of the instrument it's pointing at, so keep a wary ear out for tonal changes to any instrument, especially those in close proximity to the mic in question.

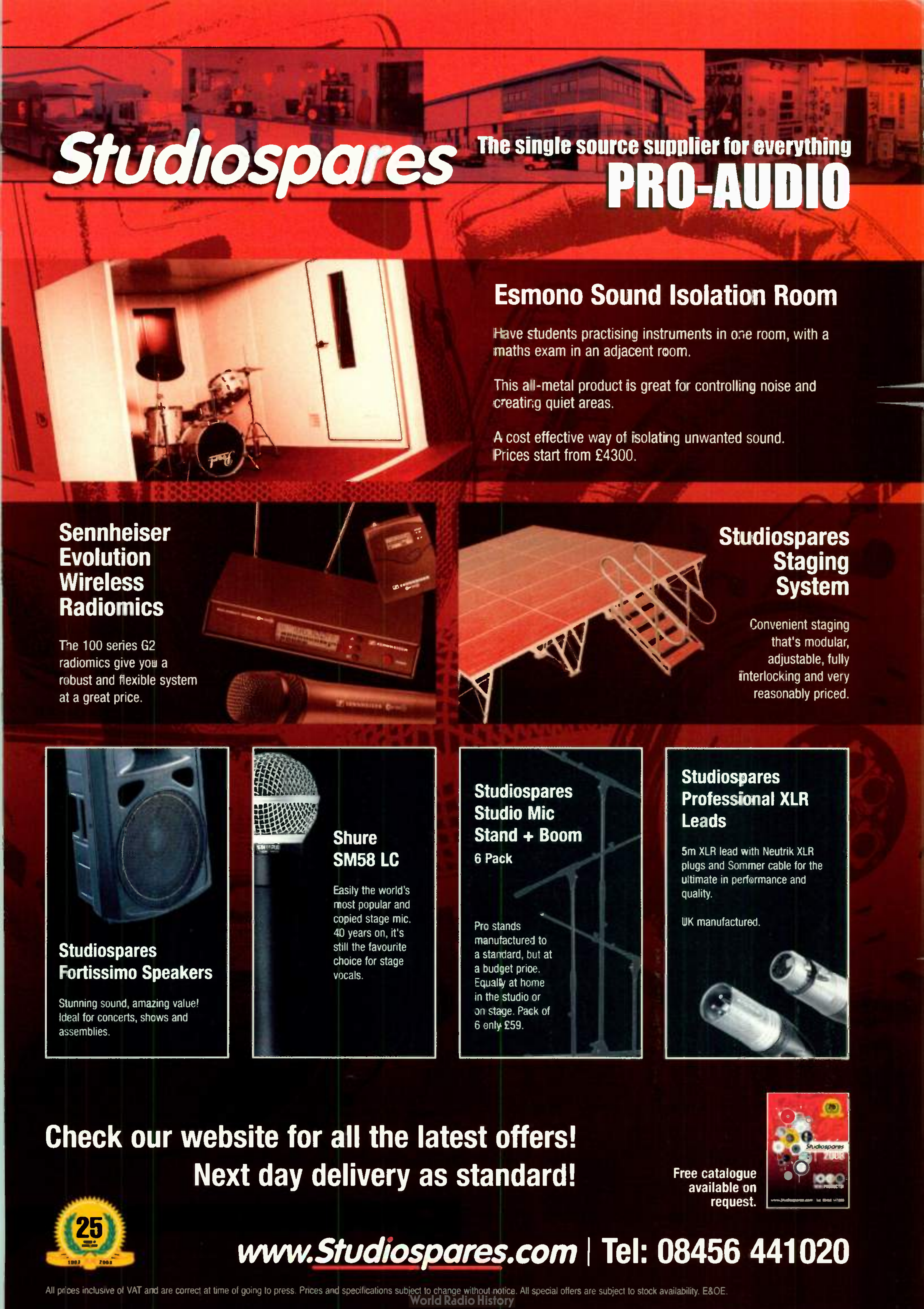

When miking a snare from above and below, the two signals will be out of phase, so use the polarity-invert switch on your desk (or in your DAW) on one of them.

As long as you're getting a good sound on each close mic, and have been careful with the timbre and level of any spill, you should be able to find a combination of polarity switches that will give you a good ensemble sound. If the polarity switches don't do the job for a particular instrument, you probably need to work harder at adjusting the quality of that instrument's spill on some of the other mics. Quickly muting a few likely contenders can help isolate which of them are contributing the most problematic spill.

A carefully multi-miked ensemble recording can allow for a great deal of mixing control over the instrumental balance, but it should be pretty obvious by now that setting everything up can be complicated and time-consuming. However, if the ensemble is already pretty well internally balanced as it is, you can make things a bit easier for yourself by trying to capture that balance via a main stereo mic rig, and then using close mics only to support and adjust the balance as required. This allows you to reduce the mix contribution from each of the close mics, or even fade them up only one or two at a time when needed, and makes the exact nature of the comb filtering between them slightly less critical to the overall sound. The phase cancellation between the close mics and the main stereo pair then becomes the predominant focus of attention.

Set Phase To Stunning

For many home studio owners, the word 'phase' in relation to mic technique carries with it a certain whiff of mystery; equal parts magic and menace. Although phase is indeed capable of transforming duffers into diamonds and vice versa, I hope I've been able to demonstrate that the issue is comparatively straightforward to understand, and that dealing with it properly will help you get better results every time you put up a mic to record. SOS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EWQL Gypsy, Voices Of Passion & Stormdrum 2

Sample Libraries For Mac & PC

We catch up with the latest instalments in the eagerly awaited 'Play' series of sample libraries.

Dave Stewart

In last month's *SOS* I waxed lyrical about EastWest's Beatles-homage sound library, Fab Four, and described the enjoyable ear-bashing I'd received courtesy of Quantum Leap's well 'eavy Ministry Of Rock. Both libraries run exclusively on EastWest's Play audio engine, which works in plug-in and stand-alone mode on Mac and PC. Accompanying Fab Four and MOR are three more Play-formatted sample collections created by EastWest's partner company Quantum Leap: QL Gypsy, QL Voices Of Passion and QL Stormdrum 2. These libraries were first announced in the spring of 2007, but would-be buyers had to wait until February 2008 to get their hands on Stormdrum 2. At the time of writing, a sixth Play title, QL Pianos, is still under construction.

When In Roma

QL Gypsy (shouldn't that be 'QL Person Of Romany Extraction'?) brings you the instruments and percussion associated with gypsy music. Given the ancient origins, complex migratory patterns and wide geographic dispersal of that particular ethnic group, you might suppose this library would have to cover an awful lot of musical territory, but Gypsy avoids excessive globe-trotting by focusing on a relatively limited instrumentation drawn from the gypsy musical traditions of Romania, Eastern Europe and Spanish flamenco.

A bandoneon — one of the many accordion-type instruments offered by the Gypsy sample library.

If accordions are your thing, you're going to love this library. I used to vainly imagine that if push came to shove I could probably get a tune out of an accordion because it has a keyboard attached, but in the case of the bandoneon, that idea's a non-starter: this rectangular squeezebox is played with buttons that produce different pitches depending on whether the bellows are closed or open! Like certain ghastly war criminals, bandoneons originated in Germany and later appeared in Argentina, where (unlike the Nazi fugitives) they play a prominent role in tango orchestras. Thanks to an ▶

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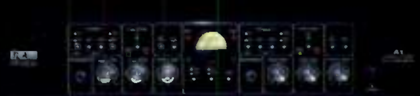

A1 Discrete Class A and Tube Dual Preamp / DI

A2 Discrete Class A and Tube Stereo Processor

A3 Discrete Class A and Tube Mono Channel Strip

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EWQL GYPSY, VOICES OF PASSION & STORMDRUM 2

accomplished sampling job, we can enjoy the bandoneon's wide range and big, stately reedy tones from the comfort zone of our MIDI keyboards.

The library has three more accordions up its sleeve: the Italian Campana model has a smaller, more friendly sound and its wheezy, warbling 'Musette' stop immediately evokes clichéd images of beret-clad Parisian onion sellers. Another Italian make, Excelsior, sounds grander and has a more percussive attack, courtesy of an optional layer of key-click samples, while the American Silvestri model has the most intimate tone of all. These accordions don't just play single notes at one dynamic; the bandoneon has long (unlooped) notes, short notes, portatos, accents and sforzandos, while the other instruments have 'air in' and 'air out' variants. A set of basic major, minor and dominant 7th chords are provided in all keys for accompaniment.

Acoustic guitar is the other main weapon in Gypsy's armoury. I enjoyed the subtle, natural dynamic response of the nylon-string classical guitar, and found that judicious use of the vibrato and legato samples livened up lines played with its no-vibrato articulations. In a jazzier vein, another acoustic guitar has a perky set of single-note multisamples played in the style of the fabulous Django Reinhardt — and these are supplemented by a comprehensive set of Django-esque chords,

which include quite complex voicings. A Spanish steel-strung guitar contributes some fine single-note 'strum' patches which, if played in the right way, can do a very decent rendition of strummed chords, while the Flamenco guitar nails the fierce strums and staccato chordal accents of that dance style. Accompanying this haughty string-flailing are flamenco dancer percussive foot noises, castanet hits and a set of monosyllabic male vocal utterances, evidently intended to encourage the dancer. Thankfully, no-one shouts 'olé'!

We now head East. Quantum Leap's Nick Phoenix is right in identifying the cimbalon as a fantastic film score element: the instrument famously performed the *The Third Man* theme in 1949, was used by John Barry in *The Ipcress File* and makes an appearance in John Williams' *Raiders Of The Lost Ark* score. The cimbalon is an Eastern European 'hammered dulcimer' (aka a large zither) played with yarn-covered beaters. Its dim, dreamy twang is terrific for melodies and arpeggios, with grace notes and tremolandos adding to the mysterious, slightly oriental atmosphere. I layered two cimbalons and detuned one slightly (with some difficulty, as the fine-tune parameter is concealed in a box called 'Current Instrument Advanced Properties'). The result was a chorused, Leslie speaker-like sound — very nice indeed, an inspirational musical timbre.

It has been said that all a Hungarian needs to get drunk is a glass of water and a Gypsy fiddler. The violinist on this project deserves a drink too, as his legato performances use nearly 7500 samples. This bears witness to the fact that the producers have utilised 'interval sampling', a technique pioneered by the Vienna Orchestral sampling company. There's no doubt in my mind that it's the best way to produce effective legatos from a sampled instrument, and here it adds a silky sheen to the violin's emotional delivery and expressive vibrato. A similar technique smoothes over the pitch slides of the very presentable solo trombone, whose short staccatos are played with great precision in classic Quantum Leap style. In addition to these romantic sounds, Gypsy makes sure you will never again have to echo British music hall entertainer Arthur Atkinson's plaintive cry of 'where's me washboard?' by generously providing samples of that peculiar domestic object. They go straight into my 'samples I will almost certainly never use' top 10.

Voices Of Passion

Ever since Lisa Gerrard did her ethnic-sounding thing in *Gladiator*, composers have been falling over themselves to hire female vocalists to add a human touch to their TV and film soundtracks. There's something immediately engaging about a solo voice, even if you don't understand a word the person is singing — and of course Lisa Gerrard, Liz Fraser of the Cocteau Twins and Miriam Stockley of 'Adiemus' fame all invented their own sung quasi-languages anyway, thus ensuring total incomprehensibility.

Quantum Leap's Voices Of Passion brings you more of the same in the shape of five solo female vocalists from different corners of the world. Naturally, I can't understand the Bulgarian singer's lyrics, but even if it turns out she's actually asking someone to call her a minicab, the sound of her voice is still

SOUND ON SOUND

EWQL Gypsy, Voices Of Passion & Stormdrum 2

pros

- Gypsy's instruments offer plenty of musical and timbral choice within their ethnic niche.
- Voices Of Passion is a great source of atmospheric female vocal performances.
- Stormdrum 2 is imaginative, varied and astonishingly powerful, and its MIDI files are a great bonus for the cinematically minded composer.

cons

- While Gypsy covers the popular bases very well, its instrumentation doesn't offer total authenticity for all gypsy-related music.
- A few phrases in VOP could do with re-tuning to fit concert pitch.
- Since everyone will be using Stormdrum 2 on their soundtracks for years to come, there's a danger the samples might become too recognisable!

summary

Once again, EWQL demonstrate a knack for filling gaps in the film-composer sound-library market. Anyone with a sense of rhythm will get a kick from Stormdrum 2 — its exciting sound design elevates it into the highest echelon of sample libraries. Gypsy and VOP are more specialised, but both contain lovely material with lots of useful variations.

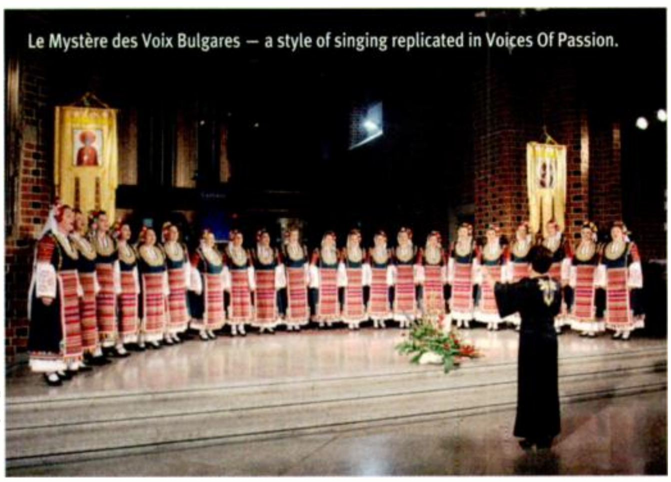
EWQL GYPSY, VOICES OF PASSION & STORMDRUM 2

mighty evocative. The vowel sounds, occasionally strident 'head' tone and elaborate multi-note ornaments will be familiar to anyone who's heard the amazing female choir on the *Le Mystère des Voix Bulgares* CDs. This is the sound of the ancient past, evolved through centuries of Thracian, Ottoman and Byzantine history: a fine collection of subtly evolving vowels, ululations, liturgical-sounding melodies and mini-phrases contrast with lovely soft oohs, ohs, eehs and ahs, all sung with great intonation. Play a string of moody string chords, pick the right vocal phrase, add some reverb, and the effect is absolutely spine-chilling.

From Bulgaria to Syria, whose vocal representative performs elaborate phrases with that unmistakable Middle Eastern melismatic delivery. The licks were sung in all 12 keys; many of them feature the characteristic Arabic scale interval of a flattened second, and their exotic, chromatic quality adds to the yearning, expressive effect. There's also some quarter-tone stuff going on which sounds out of tune to our European ears until you realise it's deliberate. (At least, I hope it is!) The South Indian vocalist also does some great improvisational stuff but a few phrases seem to have drifted sharp of concert pitch, which means you'd have to employ global pitch correction to make them fit with a backing track. It would be worth the effort, as the performances are excellent.

The Welsh singer doesn't sing in her country's language but instead performs a mini-dictionary of random English words such as 'breathe', 'dream' and 'fly' (and less cheerfully, 'death', 'drown' and 'hate'). The articulation is deliberately blurred and breathy and the words are indistinct, the intention being to provide a set of multi-purpose syllables that can be combined into quasi-phrases. Quantum Leap describe the performances as being in the 'Celtic shoegazer style', which says it all, really. If you like the singer's voice (as I did) but don't need the words, you can use her portamento slides and sustained vowel sounds, both based on legato intervals, to great effect.

Having scuttled swiftly round the globe, VOP ends up back in America with a singer who should be congratulated on the versatility of her performances, which range from soft breathy R&B 'oohs' and 'ahs' to proclamatory, near-operatic vibratos. The deliveries comprise a choice of simple and evolving vowel sounds, short ultra-breathy 'ahs', hums, crescendos and drifting-pitch sustains (good for spooky cluster chords). All in all, a very varied and colourful 7.3GB collection of expressive, finely performed and beautifully recorded solo female vocal performances, all patiently waiting their chance to make it onto the soundtrack of *Gladiator 2* ('just when you thought it was safe to go back into the arena').



Le Mystère des Voix Bulgares — a style of singing replicated in Voices Of Passion.

QL Gypsy & Stormdrum 2 Instrumentation

QL GYPSY (12GB)

Accordions
- Bandoneon (Argentina)
- Campana (Italy)
- Excelsior (Italy)
- Silvestri (USA)

Acoustic Guitars
- Classical nylon-string guitar
- 'Django' acoustic guitar
- Flamenco nylon-string guitar
- Spanish steel-string guitar

Percussion
- Bass drum
- Snare drum
- Cymbal
- Tambourine
- Castanets
- Washboard
- Flamenco dancer foot stamps

Miscellaneous
- Violin
- Trombone
- Cimbalon
- Male vocal yells

QL STORMDRUM 2 (13.1GB)

Ethnic Drums
- Congas
- Bongos
- Timbales
- Djembe
- Udu
- African bowl (kettle) drums
- Darabuka
- Darabuka with jingles
- Dumbek
- Dholak
- Odaiko (Japan)
- Nagadaiko (Japan)
- Tong zi drums (China)
- Kettle drums (China)
- Indonesian small drums
- Nepalese drum
- Roman kettle drum

Orchestral Percussion
- Bass drum
- Snare drum.
- Piatti (clashed cymbals)
- Suspended cymbals

Drum Kits
- Gretsch kit (from Ministry Of Rock library)
- Gretsch toms
- Ludwig toms
- Octaplus toms

Metals
- Triangle
- Bowl gongs
- Large bowl
- Bowed cymbal
- Chinese hand cymbals
- Finger cymbals
- Tibetan bells
- Tam tam
- Gongs (unpitched)
- Peking Opera gong
- Riq (shaker)
- Persian metal castanets
- Brake drums
- Spring drum
- Metal bridge
- Miscellaneous metal objects
- 'Whale drum'
- Waterphone

Tuned Percussion
- 'Aluminaphone'
- Angklung
- Indian cowbells
- Gongs
- Hang drum

Unpitched Percussion
- Tambourine
- Giant logdrum
- Tongue drum
- Aboriginal shaker and bone hits
- Chinese rattle drum
- Chinese wood blocks
- Danmo
- Bamboo stick hits
- 'Devil chasers' (hollow wooden sticks with grooves)
- Vietnamese shakers and rattles

Miscellaneous Percussion
- Table hits
- Miscellaneous clacks

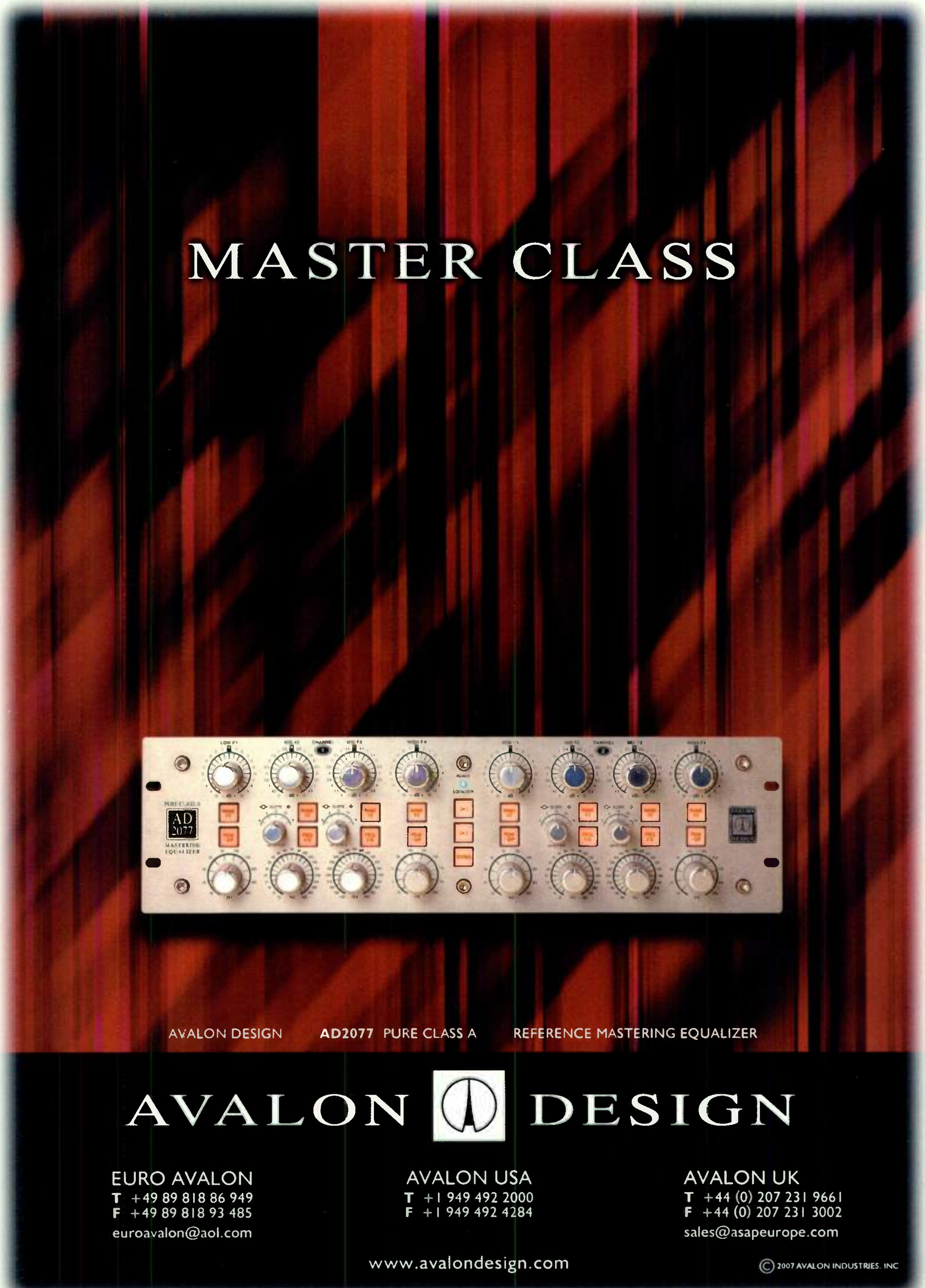
Miscellaneous Effects
- Piano staccato low notes
- Piano interior effects

Sound Design Percussion
(partial list!)
- Processed drum, percussion, synth hits and loops
- Crescendo hits
- Miscellaneous backwards 'swoosh' crescendo effects
- 'Psycho' low groans, scrapes and slithers

Sturm Und Drang

Cited modestly by its creator as "the most successful acoustic percussion library ever released", Quantum Leap's Stormdrum has won many fans since its release four years ago. This being a product aimed at the Hollywood movie industry, there had to a sequel, and sure enough we now find ourselves confronted by 'Stormdrum 2 — The Next Generation'. (I like to think the subtitle is ironic.) The brutal metallic graphics alone are enough to scare anyone half to death, so it was with trembling fingers that I tore open the box and installed the library.

What I found was a very satisfying mix of ethnic and processed percussion, presented as separate hits and also blended together in various fiendishly clever ways. The variety of sounds is enormous, as can be seen by a quick look at the instrument list above. Rather than souping up the samples with EQ and compression, the producers concentrated on capturing the natural resonance of the drums and added a very pleasing room ambience. This means that while the timbales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EWQL GYPSY, VOICES OF PASSION & STORMDRUM 2

(for example) don't clang manically like the ones in the theme tune of *I'm A Celebrity, Get Me Out Of Here*, big drums such as the giant odaiko and large, ambient dumbek really boom out impressively as nature intended. This boomy quality reaches its zenith in 'Earthquake Ensemble', a collection of low-pitched drums whose seismic rumble is capable of demolishing a row of houses. It's the kind of percussion sound that film composer Hans Zimmer is known for, with a low bass end designed to shake the walls of cinemas equipped with 5.1 surround systems.

Although culturally aware, the library is by no means purist; the producer thinks nothing of bunging a lot of reverb on a sample if it helps it achieve its effect. Consequently the massive, reverb-enhanced crescendo hits in the patch 'Rumpfs' sound absolutely devastating. But it's not all big bangs: some of the smaller hand drums sound very tuneful, making it easy to program more delicate grooves. I found some of the *nomenclature* a little unhelpful: the 'Malaysian' djembe sounds exactly like the African drum of the same name and the Indonesian 'bongos' sound more like high-pitched clay drums. I thought the intriguingly-named Roman war drum was going to raise the roof, but it turns out to have a rather pacific sound, somewhat like a softly-played timpani! The drum performances are comprehensive and varied, and it's good to hear brushes as well as sticks used on some of them.

There's a lot of very nice stuff in the metals department: the bell-like Asian bowl gongs (bowed and hit), some great large gongs (including deep Javanese-sounding specimens), the obligatory spooky waterphone, scary piano noises and even an "80-foot metal bridge". (How did they get that in the car on the way to the studio?) The stand-out instrument for me was the 'hang drum', a resonant, metal, drum-like instrument that produces tuneful pitches in the manner of a Caribbean steel drum, but with a much softer, more beautiful sound. I found myself jamming along for ages with its attractive, understated, almost gamelan-like tones (which are tuned to a D-minor scale). For those who need a drum kit, Quantum Leap have included a scaled-down version of the 'Black' Gretsch kit from Ministry Of Rock. A set of powerful tom-tom samples from different kits recorded at the MOR sessions (but not used in that library) are also included.

System Requirements

- Mac: G4 1GHz or faster, 1GB RAM, Mac OS 10.4 or higher, DVD drive.
- PC: P4 2.5GHz or faster, 1GB RAM, Windows XP SP2 or Vista, DVD drive.

Both platforms require an iLok key (not included).

Hang drums — just one of the many types of unusual percussion instruments to be found in Stormdrum 2.

Acoustic percussion is only half the story — a large part of Stormdrum 2 centres on processed 'sound design' percussion, much of it distorted, reversed and generally messed up. This kind of thing has been done before, but SD2's programmers have a talent for creating hip, contemporary noises that work well for programmed rhythm patterns, especially in conjunction with the library's giant drums. To get you in the programming mood, 106 MIDI files are included, in a variety of tempos. Each has its own multi-instrument set-up; the moods range from *Alien 2*-style military snares and bass drums to BT-esque fuzzed-up breakbeats. There are so many fantastic electronic noises in there I couldn't begin to describe them — suffice it to say that they rock, big time.

Play Away

Reviewing these wildly disparate sound libraries gave me the opportunity to get more familiar with the Play audio engine. The absence of on-screen multiple sound slots gives the erroneous impression that it's a single-instrument player, but in fact one instance of Play can handle multiple instruments operating independently on up to 16 MIDI channels. To create a multi-channel setup, you load a selection of instruments (choosing the 'Add' option rather than 'Replace'); their names are then listed in the Instrument window, and clicking on one reveals its individual screen display, where you can select the instrument's MIDI channel (which may be set to 'omni') and/or alter its volume, pan, ADSR and effects settings. It's a simple, flexible and effective system that enables you to quickly build complex setups without having to continually look at a cluttered screen.

Stormdrum 2 contains 16,000 samples, and I don't even want to think about how long it took to record the countless performances in Gypsy and Voices Of Passion. There's a sense of devotion about these projects, the long months spent recording and programming seeming to go well beyond the call of duty. As a consequence, each of the three Play titles is an artistic success. Sample libraries don't get much better than these, and any composer with an ear for sound will find much inspirational material in them. SOS

information

- £ Gypsy 270 Euros; Voices Of Passion 339 Euros; Stormdrum 2 339 Euros. Prices include VAT.
- T Soundsonline Europe +31 20 404 1687.
- W www.soundsonline-europe.com

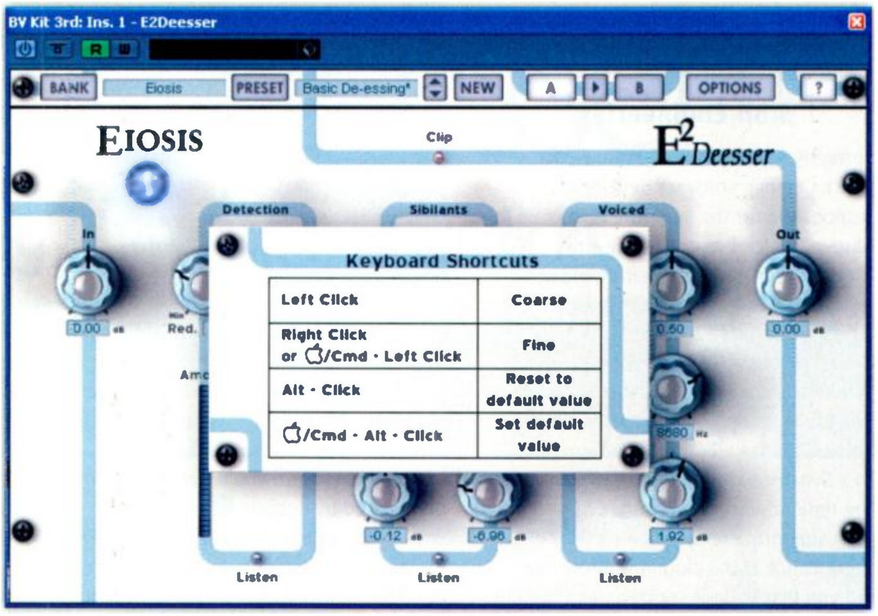
in the EQ section is causing the sibilants to disappear altogether, it's actually effective to use the Level control to bring their overall level up.

There's also a band of EQ in the Voiced section, which provides an easy way to add brightness to the rest of the signal. It uses the same algorithm as Air EQ, and sounds similarly impressive, with a choice of parametric or high shelving filter.

In Use

In action, the first thing that hits you about E² De-esser is the accuracy of its sibilant detection. Eiosis recommend using it as the first stage of a signal chain, prior to compression, and even when the level of the incoming vocal fluctuates quite a lot, the detection algorithm tracks it very well. On all the vocals I tried, it triggered reliably and consistently across a wide range of sensitivity settings, and the two Listen options make it very easy to check this.

The next thing that strikes you is how effective it is at dealing with a variety of sibilance problems. For my first test, I threw it in at the deep end, on a male lead vocal which had caused me no end of trouble. The vocal sat best in the mix if I added a high shelving boost and used fairly prominent reverb and delay, but all of these compounded the original recording's tendency to exaggerate 'S' and 'T' sounds. I had already done what I thought was a finished mix, using Waves' Renaissance De-esser, but the results had been something of a compromise. Simply substituting E² De-esser for the Waves plug-in was a revelation, even on the default setting. After some experimentation, I settled on a 5dB shelving cut at 5.8kHz, combined with a 2dB gain drop. These settings all but eliminated the harshness and spittiness of the sibilants, with almost no audible side-effects. Using the EQ in the de-esser's Voiced section to deliver a high shelving boost then made it straightforward to deliver the necessary overall crispness without undermining the sibilant reduction.



Though there are more controls than on your average de-esser, the interface is very easy to use, with some nice additional touches.

E2 De-esser faced an even sterner test on a collection of badly recorded female backing vocals I'd been sent, in which a general dullness of tone was complemented by nasty, over-prominent sibilants. Unsurprisingly, perhaps, it took rather longer to set up, but the results were still very impressive. On one particular track I ended up treating the sibilance with a brutal 12dB/octave filter turning over at 4.6kHz, and adding a 13dB level gain to compensate. The processing was audible when the track was soloed, but in context it sounded surprisingly natural, and made it possible to mix a vocal that had been borderline unusable.

Hip To E Squared

In short, with every source I tried, E² De-esser was orders of magnitude more successful than any other de-esser plug-in I had available to me. There's a comprehensive preset library, but I generally found it easier to set up by ear, a process made easier by the Listen modes and the ability to A/B two sets of settings. Because sibilants are processed separately from the rest of the audio, you can treat them as aggressively as you like without affecting the overall sound of a vocal.

The plug-in interface is generally friendly and easy to navigate, although some of the parameter names and descriptions are a little misleading, and I did wonder whether the metering could have been organised better. There's a multi-segment meter labelled Amount, which lights up to varying degrees when sibilants are encountered, but it's not entirely clear what this represents. I don't think an all-singing, all-dancing FFT display is required, but more could be done to make it clear whether the audio is flowing through the Sibilant or the Voiced path at any given moment.

Beyond that, the only functional improvement I would like is the ability to set the attack, release and look-ahead or crossfade times of the detector independently. At present, the Response dial affects all three simultaneously, in a 'coherent' way, and for most purposes, this works well. However, unlike a conventional de-esser, the amount of sibilant reduction you can achieve tends to be limited not by the sound of the processed sibilants, but by the crossfades between sibilant and voiced sounds becoming noticeable. There were occasions when I'd have liked the sibilant reduction to kick in earlier and more gradually.

All in all, though, I'm hugely impressed with this plug-in. Eiosis have applied fresh thinking to an old problem, and their solution leaves existing de-esser plug-ins in the dust. Renaissance De-esser has had a long and distinguished career, but now that E² De-esser is installed on my computer, I can't see myself going back. I don't know what Eiosis are planning next in the E² series, but if it's as good as this, I can't wait to find out. SOS

Doing The Splits

Because E² De-esser divides the incoming audio up into sibilant and voiced sections, you can copy your vocal to a duplicate track within your DAW and use two instances of the plug-in in Listen mode, so that one track produces only voiced sounds and the other only sibilants (it's a shame there's no off-line AudioSuite version for Pro Tools that would allow you to apply these edits permanently). This would allow you to, for example, apply a high-frequency boost only to the voiced part, using the EQ of your choice, or to eliminate all sibilants from your reverb send, or to use different reverbs on the two components of your vocal. The manual suggests lots more interesting applications for non-vocal sources, such as processing the attack of a conga or other hand drum independently of its sustain.

information

£ Native version 150 Euros; TDM version 300 Euros. Prices include VAT.

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TC Electronic Studio Konnekt 48

Firewire Audio Interface For Mac & PC

The Studio Konnekt 48 is an appealing package, benefiting from extensive use of TC's trademark DSP technology and a remote control — but is that enough to secure its place in the audio interface race?

Paul White

There's a bewildering choice of audio interfaces on the market, but as the requirement for more I/O channels increases, so the choice decreases. More channels means more bandwidth requirement, so most interfaces of this kind use either Firewire or some form of direct PCI connection to hook up to the host computer. The TC Electronic Studio Konnekt 48 is of the former type and offers 12 channels of analogue I/O, internal DSP effects and speaker management, and even comes with a handy remote control unit.

Compatible with both Mac and PC machines, the Konnekt 48 will work with Mac OS 10.4 or higher and Windows XP or Vista X32, and comes with its own NEAR control panel software. After you install the control panel and driver software the Konnekt 48 is ready for use, though with the version I used I had to restart after letting the software update the unit's firmware. When I first tried the unit some months ago the software was a little buggy, which is why I've held off on the review until now. I'm glad to say that the public Beta software I used for this review worked fine with my Mac, and audio playback was uninterrupted by unwelcome clicks or pops — even at buffer sizes of 128 samples or below. By the time you read this, the release version of the driver will be on the TC web site.

Comprehensive I/O

The Studio Konnekt 48 I/O section comprises 12 analogue channels — eight line inputs (switchable for -10dBv or +4dBu operation) on the rear panel, and four inputs for the TC Impact II series mic/line preamps, offering gain trim, switchable phantom power (global) and 20dB pad buttons, on the front panel. The four Neutrik combi XLR/jack (mic/line) inputs on the front of the Studio Konnekt 48 also offer high-Z guitar settings that use additional buffer circuitry to avoid loading the guitar pickups. A clever touch is that the input stage automatically senses

Alternatives

Some obvious alternatives, if you're in the market purely for an audio interface, are the recently announced Mark Of The Unicorn 828 Mk3 (look out for an *SOS* review in a forthcoming issue, and also see page 194 of this issue), the Alesis I/O26 (reviewed *SOS* June 2007), the Presonus Firestudio (reviewed December 2007) and the RME Fireface 400 (reviewed July 2007). Competing interfaces also often include software-controlled mixer facilities and occasionally DSP effects, but where the Konnekt 48 stands out from the competition is in the very high quality of its DSP effects.

when a high-impedance source is connected, at which point a relay switches in the high-Z circuitry.

There are eight analogue outputs on balanced TRS sockets and two further main outputs (1/2), featuring digitally controlled analogue level control, on XLRs. In addition to this, two sets of ADAT/TOS ports add eight further channels of ADAT I/O (two sets are needed for 88.2kHz or 96kHz operation in SMux mode). Then there's the S/PDIF stereo digital I/O — so there's no shortage of physical connectivity.

You should be aware that at 'normal' sample rates you can't use both ADAT ports at once to double the channel count, although you can configure one as a stereo Toslink port and use that in addition to eight ADAT channels at 44.1kHz or 48kHz. If you don't need ADAT interfacing, both optical ports can be configured as stereo Toslink ports, so nothing is redundant. As part of my testing during the review, I plugged in a Behringer ADA8000 eight-channel mic preamp, which interfaces via ADAT, and set it to slave from the Studio Konnekt 48. This worked with no fuss, giving me eight more ins and outs.

A pair of standard MIDI I/O ports are provided on the rear panel, and in terms of computer connectivity the Konnekt 48 sports two Firewire ports, so a further Firewire device may be daisy-chained if bandwidth makes this a practical option. Word clock connections are available on the usual BNCs to faciliate synchronisation. If the external clock signal drops out for any reason (and this includes any digital input being used as a sync source) an Advanced Clock Recovery system steps in and generates a free-wheeling clock signal at the correct sample rate to keep the system running until a valid clock source is picked up again. Also onboard is TC's 'JET' Jitter Elimination Technology. to clean up the incoming sync clock when other devices are used as the clock master.

An LED-ladder meter section on the front panel monitors the 12 analogue channels as stereo pairs, as well as indicating activity in the ADAT ports, the Toslink ports and the S/PDIF ports. A further LED confirms that the Firewire link is working.

You also get two independent headphone ports that can have different sources and levels, set within the NEAR mixer software control panel. It's even possible to route two independent signals to the left and right sides of your headphones, so if the singer likes to work with one phone off you can silence that side, to avoid spill. Alternatively, you could have the track in one ear and a click in the other. If this all seems impressive for a 1U I/O box, the Konnekt 48 has further surprises in store for you.

DSP On The TC

TC Electronic have a history of building DSP-powered effects, both as hardware units and as plug-ins fuelled by their Powercore platform, so they've taken the logical step of adding some DSP-driven features hosted by the two on-board DSP chips. These provide on-board mixing, and monitor/headphone submixing and speaker management for multi-channel work. On top of that you get the really high-quality Fabrik R Studio digital reverb, based on TC's System 4000 hardware algorithms, and TC's Fabrik C Studio mastering channel strip. This offers four-band EQ plus multi-band compression and limiting derived from the algorithms used in their flagship System 6000 platform, so you're getting the real deal here. What's more, you can run up to four of them. Additionally, TC have included their ResFilter plug-in, the Assimilator 'fingerprint EQ' and a utility plug-in that allows external hardware devices to be connected and routed within the mixer in the same way as conventional plug-ins.

As a mixer, the Studio Konnekt 48 offers a 24:8 format, and is capable of routing 30 streams of audio to the connected DAW (24 inputs plus sends and outputs from the DSP effects) and receiving 28 streams from the DAW at the same time. It uses 48-bit, double-precision summing, with 56-bit processing for the mix engine, so it may well be possible to gain better audio quality by mixing channels and groups from your DAW in the Studio Konnekt 48 and then routing the stereo mix back to your computer, rather than mixing everything in the DAW itself — depending on which DAW you use, of course. (If you prefer to mix 'inside the box', the Studio Konnekt 48 control panel still makes it easy to arrange zero-latency monitoring with effects for tracking.) The mixer includes extremely flexible routing options; a really neat touch is that the hardware senses which connections are being used, so when you

SOUND ON SOUND

TC Electronic Studio Konnekt 48
£1049

pros
- Good audio quality.
- Included remote control.
- Can function as a mixer.
- Built-in Powercore-quality DSP effects plug-ins.

cons
- None, since the early driver problems were fixed, though some may find the hardware metering a little on the compact side.

summary
A great proposition if you need a Firewire interface with multiple I/O connections, ADAT expandability, mixing capabilities and really good DSP effects that can also be called up as plug-ins within your DAW.

open the mixer control panel you can opt to view only channels that are actually connected.

The use of DSP enables the Studio Konnekt 48 to function as a controller for surround speaker systems (even offering an adjustable crossover frequency for the subwoofer), which in itself can save a fortune in surround monitor controllers. The speaker management section employs technology from Dynaudio's AIR series DSP-controlled monitor speakers, and in addition to controlling overall level it can handle individual speaker level and delay settings, and bass management for any sub you have connected. It also has the ability to store and recall three separate speaker setups, which can include headphones, if you wish. In theory, then, you could even define a speaker setup comprising only headphones and a subwoofer, though my guess is that few people will find a need to do this! As mentioned earlier, the two main outputs use digitally controlled analogue gain control, so there's no loss of resolution when you turn your monitors down.

NEAR Software

I tend to like a simple life, so I'm generally a bit sceptical of any interface that comes with its own software mixer or effects. They all claim they do it to allow near zero-latency monitoring, but using buffer sizes of 128 and below I've never really found latency to be a problem, and having more elements in the signal path can be confusing. However, the NEAR mixer is pretty straightforward and allows you to combine any of the live inputs with the DAW outputs, while adding effects if required, so if you want to set up a monitor mix where the singer has just the right amount of reverb in the cans, it is easily done. If you don't want to use the mixer functions at all, however, you don't have to. Simply mute or turn down all the channels except for the DAW output, solo the DAW output, or un-tick the Direct Monitor tick-box at the bottom left on the Mixer page, and you can treat the Studio Konnekt 48 as a rather nice 'dumb' interface with half a Powercore's-worth of effects plug-ins and a remote volume control. NEAR operated perfectly during my tests, although its controls tend to lag a little way behind your mouse movements — something that I'm told TC are trying to improve. The remote control is a great asset, though (see next section), as often-used functions, such as monitor level and talkback, are right where you need them, and you can also control a surround mix if you happen to be working in surround. The software even includes a guitar tuner, which can also be activated from the remote control.

The Konnekt 48's remote control unit: a simple idea but a very useful one.

Remote Control

Audio interfaces are often tucked away in racks where you can't reach them easily. The Studio Konnekt 48 solves this problem with a neat little

Chip Off The Old Block

The Konnekt 48 is the latest addition to TC's line of Konnekt Firewire audio interfaces based around the versatile Dice II chip. This chip is a spin-off from TC Electronic technology (TC Applied Technology) and is also available to the manufacturers of competing products (I know Alesis, Presonus and SSL already use it). Although not all of the chip's power is used here, the Konnekt 48 includes substantial DSP capability (apparently half the power of a Powercore Firewire unit derived from two DSP chips), allowing it to be used as a mix engine and effects processor for setting up zero-latency monitor mixes with effects, or for other mixing and routing purposes.

Multiple Konnekt devices can be used togther: you can use up to two Konnekt 48s at the same time, two Konnekt Lives together with a Konnekt 48, a Digital Konnekt plus a Konnekt 24D and a Konnekt 48 or four Konnekt 8s. All communicate with the same NEAR software control panel for setting up mixes or monitoring. Sample rates of up to 192kHz are supported, except at the ADAT ports, which have a maximum capability of 96kHz in SMux mode.


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Session Man

Chris Denman: Recording XFM Radio Sessions

Working straight to DAT, with a minimum of equipment, Chris Denman records more bands in a year than most engineers manage in a lifetime.

Sam Inglis

One day, the world may hear a lot more of a four-piece indie band from West London called Dega Breaks. However, it's safe to say that on their first visit to the XFM studios they're a pretty well-kept secret. I've never heard of them, and more to the point, neither has XFM's resident session producer, Chris Denman. Yet it's up to him to listen to their debut single, figure out the production tricks that make up their sound, load in their gear, mic them up and record four of their songs — all in the space of three hours. And after lunch he'll be doing it all over again, perhaps for another bunch of newly signed hopefuls, or perhaps this time for rock royalty like the Cure or Queens of the Stone Age.

XFM is London's alternative music radio station, and as part of its licence remit is committed to playing a lot of live music. Denman is the man whose job it is to put that commitment into action, and in 2007 alone he recorded 341 different bands.

Everyone who finds themselves lugging their equipment to the fourth floor gets the same treatment. There's simply no time to do things any other way. "We do one pre-recorded session at 12, then an hour lunch, then our next band comes in at four. Each session's an hour setup, so we're done and dusted by half-two, loaded out by three, I have my lunch, four — next band in. That's how tight it is, so you have to get it bang on, there and then."

Two-track Mind

The combination of time and budget constraints means that all these sessions are done using the barest minimum of equipment. "It is, basically, me, on a set of headphones out of the desk, mixing straight to a DAT tape. No multitracking involved. It comes down to the age-old art of two-tracking, which is a skill in itself. I don't have any outboard, so it's just the desk. Originally we had a Yamaha O2R, and we upgraded to the DM2000 in 2002. Nothing has actually been changed on it. I've still got version 1, still got the old internal effects."

All of which makes Denman's ability to replicate a band's sound on the basis of a 10-second flick through their recorded output even more impressive. In Dega Breaks' case, that sound is an edgy indie-rock production that owes much to the Killers, but previous sessions have covered every conceivable style, from dusty blues to cutting-edge electro, from balls-to-the-wall metal to delicate folk.

"I've touched on quite a lot of different production techniques, because I've had to. About a year ago, there was a trend for quite a lot of slap-back on vocals, stuff like that, and drums are my favourite thing, because they will basically lead a mix. You

People Skills

Unlike a producer or engineer on an album session, or someone who's touring with a band, Chris Denman has hardly any time to get to know the people he's recording, so he has no choice but to make snap decisions about how to get the best from them. "I've become very good at seeing the dynamics of a band, like who leads the band, and you work with those people to help move along a session. I also help to load in the bands' gear. It's part of the actual process of meeting the band and crew. If I'm helping crew with the load-in, they don't think I'm an arsehole producer, and that's part of it.

"I will also talk things through with the band — how did they feel doing the album? Because sometimes you'll find they've actually let things go that they didn't like: "I didn't think this was right, I kind of wanted it a bit more like this". And you end up going 'OK, I'll take that on board'. The great thing about being at XFM is that I'll actually see a signed band about twice, maybe three times a year, because of the release of singles. They'll come in about three or four times, and it's a lot easier because I've met them before. You build up a relationship, and you know what they sound like."

Of course, none of this is much use if the bands themselves aren't capable of playing live. There's always an element of risk here, especially given XFM's policy of giving exposure to new bands, but Chris is generally impressed with what he finds. "Because albums aren't selling, a lot of bands know that live is a key thing to getting their music over and making a little bit of money as well. Even the younger bands are going out and playing a lot more, so when they do sessions like this, they're actually a lot tighter."

have your room mics, and it depends on how you're compressing your room mics, depends on how much you're mixing your close mics into your room mics. I close-mic all the drums — snare top, snare bottom, kick, rack toms, floor toms, two overheads. I never mic up the hat, because unless you're really recording an album and you want to use it in a certain production technique, you don't need to.

"We'll all set up, and I'll mic up, and we'll run through drum sounds. So the drummer will play, and I'll get him to actually play the tracks — I don't like any drum techs playing, because they play totally differently from the drummer, and the drummer will have his own style and his own dynamics that come through. So I'll get that set up, get the bass and guitar sounds right, fit it all in, get the vocal sounds right, and the only thing you normally end up playing around with — the only thing you *can* play around with, really — is basically reverbs coming in and out. Maybe if they're coming into a chorus I'll wind in more delay, stuff like that.

"If you're doing covers, or if it's acoustic or stripped-back, then you'll go with something different. But otherwise you'll naturally go towards the album, because that's how they want their finished product. So if you go towards that sound, they're very happy, they're happy to play, so they do a great take. It's basically a big bit of karma you've got to deal with to make the session go well. I'm working with one hour setup, two hours mixdown. I've got to push them as a producer in getting that right. And if you go towards that, everyone's on the right path and everyone's happy. We have such a high turnover that I can't be sitting back going 'Maybe we should try that, maybe we should try this.'

"We don't have front-of-house engineers come in, because they will bring in their two reverbs that they used on tour, and I can pull up more reverbs and get closer to an album mix, even though it's a live session, a lot more quickly than they can. Obviously, this is a studio session, we're not in a venue, and when front-of-house hear stuff, they will let things go, but I'm too anal, I'll rein them in."

Tones To DI For

All the sessions are recorded in a converted radio studio at XFM's headquarters in Leicester Square. A small, soundproofed live area is available for drums, but everyone else in the band plays in the control room. Originally, guitar amps were dumped into the corridor outside, but this caused serious problems with bleed into the adjacent radio studio, and Chris's preferred approach now is to DI guitars and basses. "Because everything is muted in the room, it's funny, because the only thing you'll hear is a vocal and the guitar strings going!"

Two DI'ing options are available, thanks to endorsements from Line 6 and Sequis: a selection of Pod digital amp modellers, and a Motherload dummy load/speaker simulator. Guitarists who insist on using their own amps can be directed to the latter, ▶

Chris uses his Line 6 Floor Pod XT to mimic most bands' recorded guitar tones; using the Gearbox utility on his Mac to program the Line 6 hardware, he has built up a huge collection of signature sounds.

SSL XLogic VHD Pre

Four-channel Mic Preamp

Paul White

Some people love the sound of SSL's modern consoles, while others prefer the 'dirt' of their older models. The VHD Pre aims to offer you the best of both worlds...

SSL's rack mount spin-offs from their high-end console technology are proving extremely popular with the more discerning end of the private studio market, and this is a niche in which the rather lengthily named SSL XLogic Alpha VHD Pre sits very comfortably. As well as four highly specified mic preamps, you get channel-independent line (Pad) and instrument level switching, so that you can use it to get just about any form of signal into your recording system — so if you added one (or more) of these to a competent DAW system fitted with good quality converters, you should have a seriously capable recording chain...

Variable Harmonic Drive

The basic preamps are very clean and quiet, but SSL have also added in their VHD circuit. Though this may sound like an antisocial disease, it is in fact a very clever Variable Harmonic Drive circuit that introduces subtle (and, at higher settings, less subtle), character-creating distortions, in much the same way as a tube audio circuit does when pushed hard. The harder you drive a tube the more harmonic distortion you add — and VHD does the same by using FETs, with more input gain bringing more warmth. In keeping with tube behaviour, it's even OK to get the red Pad level light flashing if you want a very noticeably overdriven effect.

The VHD system was originally developed for the company's flagship Duality console, but it soon became clear that it was worth adding to other products. As the input gain is increased, the VHD circuit starts to add second- and third-harmonic distortion and the VHD control allows the user to adjust the balance of the two harmonic components. Anti-clockwise adds mainly second harmonic while clockwise adds mainly third harmonic. The even harmonics tend to sound more like tube distortion, whereas the odd harmonics have more of a solid-state crunch to them. The range of effects on offer varies from almost clinically clean to noticeably trashy and when used with restraint, the effect on vocals can be very sweet and flattering. There's plenty of output gain range and a creditable amount of headroom, so you can maintain a useful output level at either very gentle or very high input drive levels.

Overview

The VHD Pre has a very clean-cut appearance, partly down to the lack of obvious metering (which isn't a problem in practice, as the Pad button lights up red if you're in danger of overcooking the input level). There are input and output gain controls for each channel, the VHD knob, and three illuminated buttons that are styled after the fashion of partly-sucked Glacier mints. One of these activates the phantom power, one a 20dB pad, and the other switches the input to high impedance, for use with electric guitars and basses that are fitted with passive pickups. However, I was surprised to find that there's no low-cut filter, and I feel that's a serious omission.

Everything is old-school analogue, and there's no audio interface or digital converter built in. Power comes in on the rear panel via a five-pin DIN socket that connects to the included universal power adaptor, which automatically adapts to the local mains voltage. You'll also find XLR mic/line inputs and XLR outputs for each channel, but the outputs are not doubled up on jacks, as is often the case with other products aimed at the project studio market. Instrument input jacks are located on the front panel, and these override the XLR inputs, allowing the latter to be left permanently connected if required.

I have a couple of concerns over the connections, although these shouldn't be a problem if you're the careful type. One is

SOUND ON SOUND

SSL VHD Pre £999

pros
- Tonally versatile, from clean to flattering.
- Sensibly priced, given its pedigree.

cons
- No low-cut switch.
- Very rudimentary metering.

summary
The VHD circuit elevates the Alpha VHD pre from 'well specified but unremarkable' to tonally very flexible. Used with care, the VHD system really does sound convincing.

Inputs and outputs are all on XLRs, while power comes in via a five-pin DIN socket.

that because the power comes in on a five-pin DIN, you could conceivably plug the output from the PSU into a MIDI socket by mistake. The other is that the mic and line inputs are effectively the same thing but with a pad for the line signals, so it is conceivable that you could accidently switch phantom power onto a line level source and possibly do it some damage.

Oddly enough, for a professional product like this one, there's no useful technical spec other than the size and weight of the unit and so on. There are no noise, distortion or other performance-related specs in the literature that comes in the box. I can only assume that SSL have taken the stance once adopted by Rolls Royce for their cars, where the horsepower and top speed was specified in the handbook as 'Adequate'.

Solid & Logical?

Tested with a variety of studio mics, the preamps came across as very neutral, providing that the input gain was kept below halfway and the output level control used to make up any necessary gain. When the VHD Pre is used in this way there's no obvious character, but then a well-designed amplifier should only make signals bigger, not change the way they sound. The VHD effect comes in only at higher input gain levels, where you can hear a subtle thickening and brightening of louder sounds — though if you set the drive level so that the red clip light flashes occasionally on peaks, the effect is far less subtle and may be more applicable to electric guitar and bass or hip-hop drums than to most vocals. I found that driving the input until the red LED started to flicker and then backing it off so the light didn't come on at all worked pretty well for male vocals, and I was able to fine-tune the character of the distortion by adjusting the VHD control and backing off the input gain where the effect was too pronounced. Anti-clockwise sounded warm and somewhat tube-like, while clockwise was brighter and grittier. By adjusting the VHD control and the input gain control I could coax some very nice sounds out of this preamp, but given the lack of any meaningful metering to indicate the amount of harmonic distortion being added, I found that I had to make all adjustments by ear.

I found the Alpha VHD very competent, but other than the harmonic distortion control there's nothing that really makes it stand out when compared with other respectable preamps in the same price bracket. In fact, in a direct shoot-out with the Aphex 207D that I was also testing at the time, I found that I slightly preferred the crisper sound of the Aphex unit on my own voice — although, to be fair, with careful use of the VHD control I was able to get pretty close. Such preferences are, of course, subjective, however, and the beauty of the VHD control is that you can emulate a range of 'flavoured' preamps as well as having a clean one. I chose the Aphex preamp for comparison as it has a similar (slightly lower) cost per channel — and because I had one to hand — but the VHD Pre also compared well with the preamps in my MOTU interface and even with my Universal Audio Solo 110.

Alternatives

There are few directly comparable four-channel alternatives, but on a cost-per-channel basis the following preamps would be worth auditioning: the Aphex 207D; the Focusrite 828; and the SPL Gold Mic.

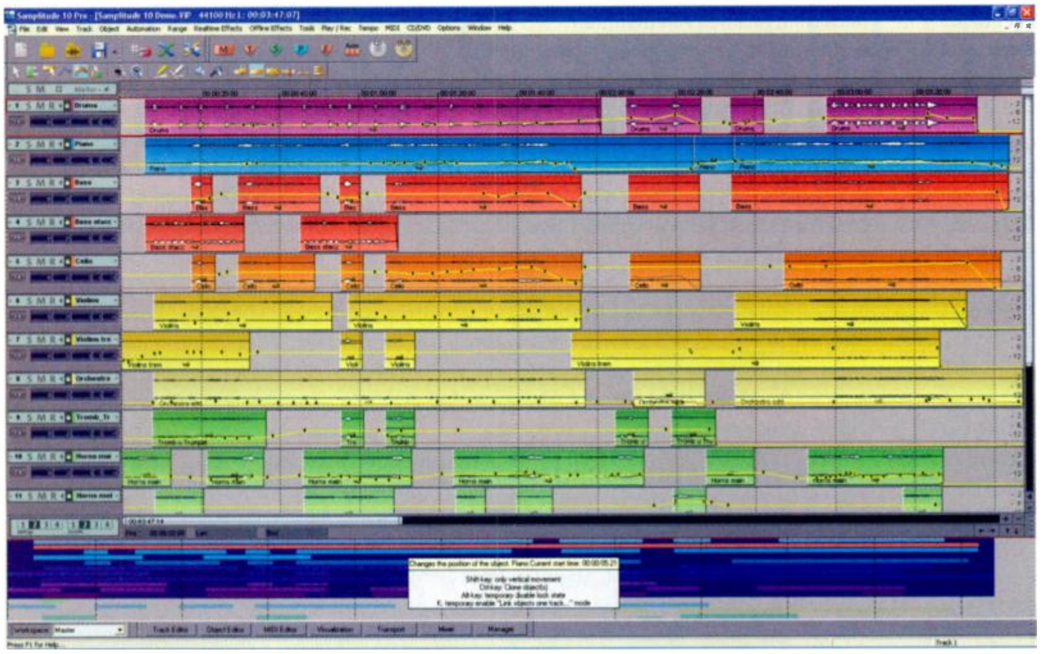
Conclusion

Having four mic preamps in a 1U space is definitely welcome in today's crowded studios, and the VHD system could offer a viable alternative to owning more mics just to capture the right vocal quality for any given singer. The SSL name doesn't do the client credibility any harm either.

Although there's strong competition in this market, the VHD Pre offers good value, it is very easy to use (though you need to be careful when adjusting the VHD drive) and it offers a wide range of useful preamp characters. In fact, I've nothing negative to say about it other than that I'd have liked to see a low-cut filter included. All in all, this is a very flexible preamp that delivers a very convincing sound and would be well worth auditioning. SOS

information

- £ £999 including VAT.
- T Sound Technology +44(0)1462 480800.
- E info@soundtech.co.uk
- W www.soundtech.co.uk
- W www.solid-state-logic.com



The Overview, at the bottom of the screen, is an excellent aid to project navigation.

in a holding pattern for whole seconds. I'm sure this is easily fixed: it's not as if that action is competing with any other and needs any real computational decisions.

One feature that has slightly changed content but, happily, not its irrationality is the gloriously random (and slightly bonkers) index at the back of the manual. No mention will be found there of Am-munition dynamics processing, Cleaning & Restoration Suite (hint: look under 'New'), Resampling HQ algorithms, or other such useful items. Instead, under 'A' we find "Access to the Internet must be permitted"; under 'H', "How it works" ("it" turns out to be aux bussing); under 'I', "Information regarding Samplitude network installation". My favourites are under 'T', which covers both "The difference between loading and importing audio files" and "There is no permanent memory on the standard CodeMeter stick". There are also, incidentally, 14 individual entries for ROBOTA, all referring to the same four pages. But this isn't really a complaint; such quirkiness now feels like part of Samplitude's individual charm (like the ketchup bottle on the SADiE toolbar for 'write back to source file').

Automation

The most powerful new feature in Samplitude 10 is the advanced automation that is now available at track, object and master level. At track level — the most comprehensive of the options — you can automate the volume, the pan, aux sends, track EQ, and all of the parameters of the excellent suite of Magix plug-ins that have been provided as standard since Samplitude 9 (see Sam Inglis's review in *SOS* January 2007). Track volume and pan automation is pretty common in most digital audio workstations these days — how could we create a mix of even moderate complexity without it? — but the new availability of full and efficient automation for the normal channel-strip and other Magix plug-ins opens up a whole host of creative and corrective possibilities.

In the early days of multitrack recordings, mixes were generally fairly 'static': the main advantage of mixing was felt to reside simply in making post-recording balance choices possible, so once a mix was agreed it stayed pretty much the same throughout the song. But then more dynamic mixing techniques were developed and 'flying fader' automation was invented to ease the task, and eventually upped the creative ante — for those who could afford it. I have fond memories (well, memories) of working with non-automated mixing desks in situations where I had to write out and rehearse fader moves, with every member of the band being given responsibility for various mix parameters (though never the level fader of their own instrument!) for live dynamic mixdowns. Of course, with no automation, there was no recall either, so an error in any single pass required a completely new attempt. Within such constraints, the creative element of dynamic mixing was severely limited (although, of course, it didn't feel like it at the time): fading sources in or out, up or down, and rough manipulation of aux send levels was just about the best that could be hoped for. So when automation became more widely available, creative juices began to run more freely. This is the kind of place where art and technology sit so well together: when what is initially developed as an added convenience eventually becomes a creative tool in its own right.

The reason I'm labouring this point a little is because until you have tried working with this level of automation you may not know just how creatively interesting mixing can be. Of course, such a level has been available for a long while, for those with the budgets to cover it, and even in some lower-budget DAWs, but the particular combination of well thought-out automation and the high-quality plug-ins in Samplitude seems to me to provide a whole new level of creativity for the cash.

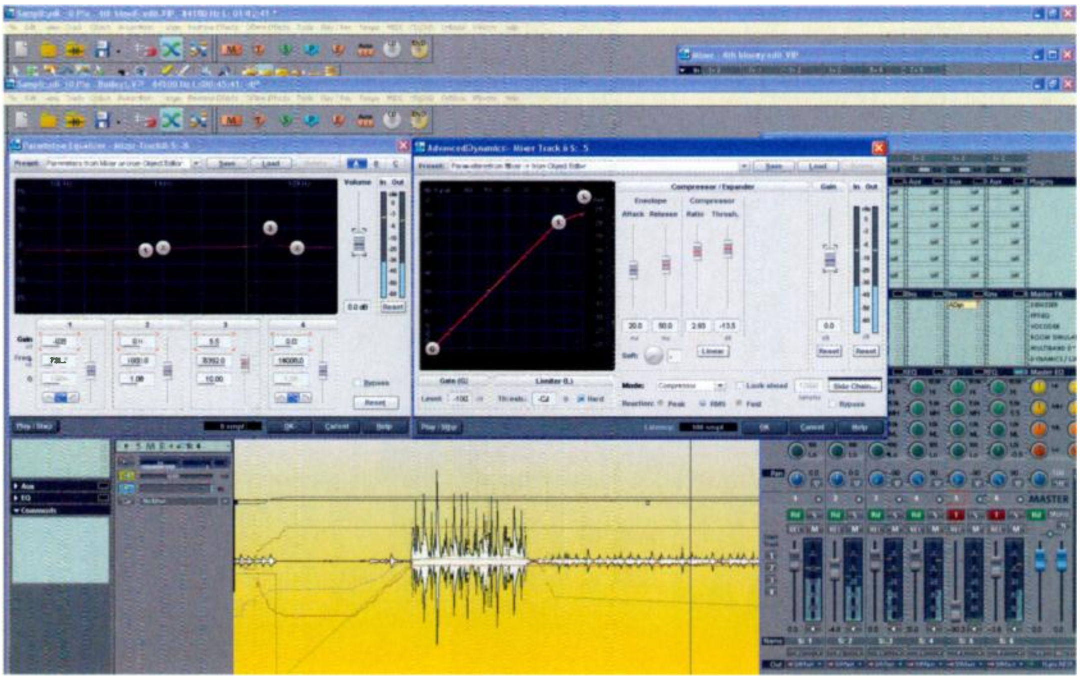
Power To The People

Here's a real-life example. I was mixing some songs for an all-acoustic folk band with twin lead vocals — male and female; there were no purist constraints on the mixing, so I was free to use standard techniques of EQ and compression. I was setting the mix parameters of the vocals, and I had automated the relative volumes so that when one of the voices was joined by the other, the first was dipped slightly, relative to its original level, to make 'room' for its companion. It all worked fine, and generally I would have been quite happy with the result. But then I started playing with the auto functions on track dynamics and track EQ, and found that really small adjustments to both during twin passages — basically increasing the role of the compressor and dipping clashing frequencies that created a mild harshness — not only gave me more easily mixable vocals, it created a more powerful effect.

Now, such a manoeuvre would actually have been possible in Samplitude 9: I could have created new objects at the crucial points, changed the object-level dynamic and EQ settings, and then crossfaded between them, with a mirror operation at the other end of those objects where the parameters were all returned to normal. But that's obviously fairly unwieldy and quite time-consuming, especially given that I'd have to experiment with the exact amount of change that had to be faded in and out, and with the fade parameters themselves. With the new automation in Samplitude

10, by contrast, it took less time to make these moves than it has for me to write this sentence. I simply set the automation recording mode to Touch, opened the plug-in dialogue for the Am-track compressor, pressed play, and made my mouse moves at the appropriate moments (in this case I experimented with lowering the threshold and/or increasing the ratio). Once that section was done, I changed the automation mode to Read and listened back to the result.

The absolute ease of this kind of operation leads fairly swiftly to other ideas about how to use this new power. What happens if we increase the chorus effect on one of the acoustic guitars just before the next verse? Do we create a more intimate effect if we momentarily dip the female vocals' aux send to the Variverb during her final solo passages? What whacko (but, of course, creatively entirely justified) results do we get if we sweep through a few parameters of Filtox during that violin solo? Naturally, we know that power corrupts and absolute power corrupts absolutely, but once you're past the showing-off-to-your-friends stage, this level of easily applied subtle enhancement becomes quite normal. As with volume automation, you will need it to be available, but after a while you no more feel a great desire to include such 'enhancements' all the time than you feel the need rapidly to ride the faders at inappropriate moments.



New automation features at last support Samplitude's own plug-ins as well as third-party processors.

Fixing It In The Mix

In addition to this creative use, automation opens up corrective and what we might call pre-emptive problem-solving possibilities as well. The following example comes from an interesting and unique session that arose at short notice — but with excellent timing — just as Samplitude 10 arrived for review.

The project was to record, edit, mix and master a complete three-song EP for the jazz singer Frances Shelley, using a small band of musicians who, through good fortune and coincidence, just happened to find themselves in the same part of the country for a single day. The venue for the recording was Butley Priory, in a small, stone-built main room which had a sort of circular domed ceiling, and the idea was to record as much live work as possible, with the backing musicians more or less improvising their accompaniment from some written-out skeletal structures of Frances's songs. This organic creation of the arrangements worked surprisingly well. The instrumentation was just Frances' vocals plus acoustic guitar, violin, cello and African percussion, which together created a quite lovely blend of melody, texture, and harmony — but it was also a combination that presented a real challenge to record and mix to professional levels. The main recording problem was obviously mic bleed from the percussion, especially the larger instruments, whose low tones sometimes filled the room. The players had to be close enough to one another to allow clear sight-lines for the non-verbal communication that improvisation demands, and the mics had to be close enough to keep the room reflections under control (that domed ceiling created some interesting reflection patterns), but then they also had to be far enough away to allow the capture of the natural acoustic timbres of each instrument. That's not a happy mix of requirements.

The recording ideal for natural instruments (in classical music, jazz, or any genre with audiophile ambitions) has always been to capture everything as truthfully as possible, using minimal miking and careful placement, and to keep post-production processing to a minimum. Well, that's the official hair-shirt version. But the reality quickly dawns that unless you have a superb recording acoustic to begin with, and are given adequate setup and experimentation time, all such recordings require some compromises. And the sad fact is that record-company politics and higher costs often mean that room quality and engineering time are the first things to be cut. So, in my humble opinion (speaking as a sometime specialist in small-scale classical recordings), the biggest boon in the past decade has been the development of professional tools that I can use after the recording to minimise the imperfections which I know those compromises have created.

If, as is all too common, we can use detailed editing successfully to create the illusion of perfect instrumental technique, I now see no reason not to factor in the use of perhaps fairly substantial post-recording processing to take the heat off the engineer faced with imperfect acoustics and/or lack of time. Of course, for most non-naturalistic recordings this has been a normal way of working for years, but the fact is that this way could only become possible for the naturalistic engineer once processing — specifically EQ, compressors and reverb — of quality sufficient to make it 'invisible' became available.

I mention this to introduce the fact that although it wasn't a matter of expense, I knew that the recording we wanted to make was going to create issues that would require post-recording correction, so I took a small gamble and set up for the recording with that specifically in mind. My recording decisions then responded not to the rough mix that I actually heard during the session, but to a 'virtual' mix that was being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Am-munition is a powerful and complex mastering dynamics processor.

'processed' in my imagination. Of course, this way of working can then be applied all the way down the line: if you know that you have sufficient quality tools at your disposal (and control over the decisions!) at the next stage, not only can you record with the mixing in mind, but you can mix with the mastering in mind. This isn't so much fixing in the mix as keeping a single creative line going through what were generally thought of before as three separate processes. So the subsequent challenge for Samplitude 10 was then whether or not it could actualise the virtual and bring off that processing at all, and if so, in a way that would allow the artificiality of it all to be concealed.

The most simple and straightforward example to illustrate the corrective features was that of the far-too-omnipresent low end of the percussion I've mentioned before. Not only was this in the room generally, but given the mic distance for the individual instruments (whose natural timbral characteristics ruled out close-miking) it was also in all the mics that were open. Often, when mixing such a project, the first thing to do would be to fade down those mics when they were not actually in operation, but here that was not an option, because the placement of the mics and the resulting mic bleed were the result of deliberate decisions taken to create a nice overall acoustic space. The percussionist controlled his dynamics to suit the sections of the music, playing with great subtlety during the vocal or solo instrumental passages, and really letting his sounds ring out in ensemble sections. The problem then was to allow the mix to reflect this, and the answer was obviously to automate the track EQ. There are other ways to achieve a similar result, but given the ease with which problematic frequencies, once identified, and well under the fundamental of all the other instruments except the cello, could be dialled in and out, and the ease of listening that easily automated features allow the mix engineer, this was clearly the technique of choice.

Am-munition

Magix describe Am-munition as "an extremely versatile, dynamic tool for editing [*eh?*] groups or signal sums, especially in the domain of mastering. It has separate units like compression, filtering, side-chain, limiter and clipping." These, they go on to say, allow for "effective enrichment of the programme material without causing bothersome artifacts, a high reachable volume, and 'analogue' behaviour with an individual sound signature." They then add, ominously, that "Due to these [*separate units*] and other details, Am-munition appears inconvenient and complex compared to other traditional dynamic tools." Oh yes.

Probably the best way to think about Am-munition is as a device that shares the dynamic manipulation load between compressors, limiters and soft-clipping devices. It can operate in either stereo or M/S mode and comprises two side-chainable opto-compressors, followed by a two-stage limiter, which is itself followed by a two-unit clipping stage, the first unit of which allows for two-way frequency-dependent clipping.

That didn't really help all that much, actually, did it?

The easiest way to think about Am-munition is as a compression/limiting device that doesn't actually need you to understand its innards in order to begin to use it effectively. Knowing more about its actual modes of operation will naturally lead to more efficient and effective use, but as whole articles could be written about how to use it (and will be: look out for

These are just some of the internal options available within Am-munition.

a forthcoming *SOS* technique column on this processor), it seems to me that the best way to learn is to start with one of the 44 presets provided and tweak them to taste. The whole learning curve can actually start quite gently, due to the excellent metering provided. This tells you just how much compression you're adding, just how much limiting and even how much clip management is going on. There's also a hugely useful gain-adjustable bypass button, which enables you to match the level of the processed and unprocessed signal and so hear the effect of the manipulation without being misled by the apparent advantages of mere greater gain.

My own listening tests were entirely positive. Having worked my way diligently through the flow chart provided in the help file, I still began by simply running a variety of mixes through the device, choosing a Samplitude preset that had something like the right name, and then adjusting the main parameters to see what happened. Having worked out the general procedure, and the way compression 'slope' is nicely different in relative effect from 'ratio', I then began to thoroughly enjoy the experiments — even finishing a couple of projects along the way. In my humble opinion, there is nothing in all of explored space that can match the Pendulum OCL2 for transparent compression duties with acoustic material, but it seems to me that Am-munition benefits from one of the best opto-compressor emulations I have heard so far, especially at the gentle end of the settings, with the warmth and musicality for which this form of compression is renowned. I was less taken with the sounds of the plug-in pushing hard into the later stages of limiting and clipping, but the music I generally work with has little tolerance for clipping, no matter how well managed, and neither do I. Tastes, indeed perceptions, vary widely here, though, and if absolute level is sometimes (still?) at a premium, at least Am-munition enables you to choose and manage your sonic compromises.

Conclusion

What is beginning to push Samplitude ahead of the large, unruly pack of DAWs that now offer complete in-the-box solutions for recording, mixing and mastering engineers is the usable quality of what actually comes in their boxes. Magix's decision to automate their plug-ins, and then add the excellent Am-munition to their roster, makes a strong case for Samplitude being quite the best DAW of its class. My grump about the non-inclusion of the new Cleaning & Restoration Suite in Samplitude Pro is only because that one odd omission seems to break the promise of being able to buy a single software package that, in the right hands, can do everything required. In absolute terms, of course, the individual elements have competition from the very highest-quality stand-alone developments (both hardware and software), against which they do not compare, but even if you have the budget to employ those tools, I'd still suggest you buy Samplitude for the basics of capture and mix control and then move further with the extra money. For the rest of us, given that those basics now come in at such a high quality, we can buy Samplitude for nearly everything else as well and be amazed at the money we've saved. SOS

information

- **£** Samplitude Professional £787.63: Samplitude Master £235.35: Samplitude £393.81. Upgrades £201.59, £79.70 and £161.28. Prices include VAT.
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slightly offset to one side. In conjunction with retuning the batter head, this got us a much more usable sound, with a decent snap to it, as well as a good body weight, though more time spent tuning the kit would certainly pay off. We still needed to EQ fairly radically to arrive at an acceptable sound, and as well as boosting in the expected 70-90Hz range for punch and the 4kHz range for click, we also needed to cut back at 200-300Hz to reduce boxiness. Many kick-drum mics have this type of curve built in, to reduce the need for lots of extra EQ, but the Samson model clearly didn't lean far enough in that direction.

Perhaps the most dramatic improvement you can make to a recorded kit sound is to gate the toms. If you solo the tom tracks they hum and resonate all the time, even though you may only hit them twice in the song! We gated both toms, the kick and the snare, using the fastest possible attack time and one millisecond of look-ahead, so as not to kill off any of the attack. You need to adjust the release time of the gates so you don't kill the drum sound too abruptly, but you can afford to make it slightly shorter than is natural, as adding in the overheads usually disguises this. If the kick drum lacks consistency, a compressor with an attack time of 15ms or so and a release time of around 60ms will beef it up a little. This can also help to emphasise the initial click.

I often find myself rolling some of the low end off overhead mics, but the Samson C02s seemed to pick up mainly mid frequencies and high end, so little or no EQ was needed. Pushing all the faders up revealed a less boxy drum sound, but we had to roll off quite a lot of low end from the toms (using a 350Hz shelving low-pass filter) to stop them from sounding 'lumpy'. Our extra absorbers had lessened the unpleasant impact of the room sound, so one effective way to put something more attractive back is to find a very short, lively ambience reverb program and add a little of this, either to the whole kit or just to the overhead mics. You can then add more conventional plate or room reverbs to individual drums as required. In this case we found something suitable courtesy of Sonar's convolution reverb; all the synthetic reverb plug-ins that Stu owned seemed to have too slow a build-up time.

Vocals

Our final challenge was to improve the sound of Stu's recorded vocals, which he'd been doing in the live room, with a couple of blocks of furniture foam stuck behind the mic to form a kind of DIY Reflexion Filter. Today was Stu's lucky day, though, as Sonic Distribution had kindly given us a genuine SE Reflexion Filter, so after abusing the mounting hardware to make it balance better on a normal mic stand, we put up Stu's AKG tube mic and made some test recordings. The combination of the Reflexion filter and the extra Auralex foam in the room really helped, and we got the best results with my spoken word test when I had my back to the foam on the inside of the door. Hanging a duvet over a boom mic stand set up in a T-shape behind the singer (which Stu said he would definitely try) would dry up the sound even more, so he shouldn't have any further problems with his vocal recording.

Stu seemed pleased with the improvements and was very happy to have a real Reflexion Filter in place of his home-made version. The drum recordings were definitely moving in the right direction and we had a simple strategy for avoiding boxy vocals. It may be that the live room could be improved further, by substituting two or three of the drop-in suspended ceiling panels with foam slabs above the drum kit, as mentioned earlier, but that was beyond what we could achieve, given the materials we'd taken with us, so that's one for Stu to decide. And so, after polishing off his vanilla cream Hob Nobs and making a start on the chocolate ones, we had our final cup of coffee and set out for home. SOS

Reader Reaction

Stu Evans: "When I first built the studio, I focused on making the room as soundproof as I could. It was later that I found it hard to achieve consistent sound and soon realised that I needed to put in more time post-recording.

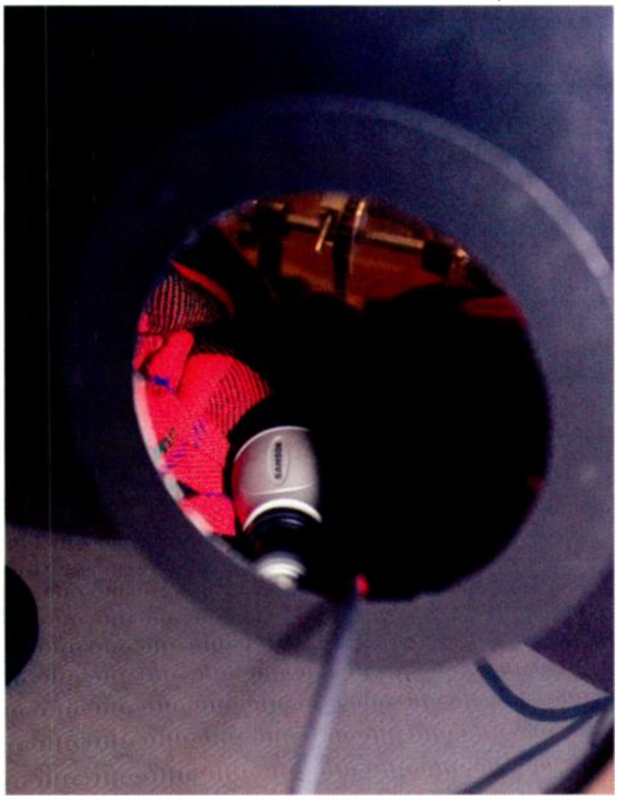
"The difference in the control room is striking. I instantly wanted to go over all the recordings I had made and remove all the EQ I had spent hours adding and even building into my templates. With the moving of the monitors and the changing of the monitor settings, which I had just left to 'normal', I now not only have more space on the desk but the sound has also gained some real definition.

"I never liked the bass drum sound, even though it's a good kit. I had not tuned it for some time and it was a fair point that a decent tune-up makes a whole lot of difference.

"After we tried the vocal setup in the live room with my AKG Solid Tube I was relieved that Paul and Hugh did not laugh at my home made Reflexion Filter, instead presenting me with a real one. I was very impressed with the results, and especially Paul's bedtime-story voice (a wasted talent). I am definitely going to add the foam to the live-room ceiling and see how that goes.

"I'd like to say a big thank you to *Sound on Sound*, to SE for that nice Reflexion Filter, and especially to Paul and Hugh for a good day and spending the time to go over my setup. A great result!"

Stu Evans in his revamped studio.



To give more snap to the 'beach ball' kick sound, we retuned the batter head of the kick drum. We also took the microphone off the short stand it was usually mounted on and placed the mic inside the drum, about halfway between the heads and slightly to one side, so that it picked up more of the beater sound.

Samplebase Satellite Pro

Software Sampler

Satellite Pro may look like just another soft sampler, but it has a unique sample-library philosophy and a particularly attractive price tag...

Alan Tubbs

Every month seems to bring a new crop of sample players to the sonic field, and it's getting harder and harder to choose the right software.

Although Samplebase Satellite is similar to other soft samplers in many ways, it does have a number of features that set it apart. Firstly, Samplebase is an Ilio company — Ilio's samples have had rave reviews in *SOS*, as well as in the working world, and a superb sample library is a huge benefit for any sampler. Secondly, although Satellite might not be unique as a sample player, it does everything you could expect it to do, and then some. Perhaps the most interesting thing about Satellite though, is the sample library that comes with it — or, rather, doesn't come with it. You see, instead of coming with a massive sound library included, like much of the competition, the Samplebase library comes in the form of 'Soundblocks' — small collections of samples (typically between 20 and 200MB) which can be downloaded from their web site for a small fee — usually between $19 and $49. This allows you to build up a sample collection based on your own precise sonic needs and keeps the cost of the player down. Satellite comes in two flavours: a free 'preset' player and a highly editable Pro version that costs just $149.

The main page makes it easy to see your setup, mix your channels and control the available synth settings. Satellite Pro will let you reassign the controller knobs to any available parameter from a drop-down list.

We Have Ignition

Satellite works stand-alone or as a VST or AU plug-in and, at its most basic level, is a sample player. You load in a sound, adjust the envelope and filter settings, if you like, and play. You can stack and/or split multisamples and use granular synthesis to retune the sample and/or time-stretch it. Satellite will also load loops, including 'sliced' audio (REX or RX2 files). Finally, you can mix and match loops in a Multi, assign them to keys and then play them together.

I installed both the Free and Pro versions of Satellite for the PC. Both versions installed quickly on my PC and without a hitch. Satellite is advertised as being compatible with Mac OS 10.4 and Windows XP, as well as earlier PC operating systems, but I managed to run it on a Vista machine with no real problems.

Samplebase include a free 'Ignition' Soundblock to get you going. It's more of a teaser than anything else, but enough to see how the software functions. Soundblocks are at the top of the sound hierarchy and include the raw samples along with information about keyboard assignments and layering, synthesizer and effects settings, looping and transposition information and the assignment of patches across the 16 MIDI and virtual mixer channels in Multis. In case you were wondering, you don't get access to the raw samples themselves, since they are locked up within the Soundblock.

Samplebase is a completely on-line entity, and you get an on-line account that contains

SOUND ON SOUND

Samplebase Satellite Pro $149

pros
- A great library is available.
- Excellent loop editing and playback.
- Good fun to play.

cons
- Library isn't included.
- No contextual help.

summary

Satellite Pro gives access to a world-class library at a good price, and includes a well-rounded sample and loop editor with plenty of power for rolling your own performance Multis.

all of your purchased Soundblocks and software. Other than the possibility of men in black helicopters grabbing keystrokes, the only downside of doing business on-line is the fact that if the Internet does go down, you're stuffed. I've had to pop onto the Samplebase site many times for this review and couldn't get a connection a couple of times, but for normal usage this shouldn't be a problem. The only time you really have to access your account is when adding more Soundblocks, or, rather irritatingly, if you want to view the Help menu. Without an immediate Internet connection on my music computer, I was reduced to saving the manual and clicking on the PDF file. There's also no contextual help available. That said, the manual itself is in depth, while the on-line tutorials are well done, especially for the novice.

Blocks Of Sound

What really sets Satellite Pro apart from everything else is the way the library is provided. Say your magnum opus needs nice cellos, or a movie score could use some Eastern European voices but you're on a tight budget: Soundblocks cost a fraction of what you'd pay for full libraries like Vienna and are a very cost-effective way of building a sample library if you know what you need. You can listen to demos of the on-line libraries just to make sure. While the Ilio samples are of a known quality, the rest of the Soundblocks I tested are also very good.

Soundblocks come in two different types: Construction Kits and Instruments — both of which are fairly self-explanatory. There are far too many Soundblocks to discuss here, but I must say the Electrohouse Soundblock made me want to set up a disco ball in the studio, while Garden Grooves begged me to turn down the lights.

It isn't hard to pull in more Patches and play over the beat or re-cut the loops to your own specifications. Nor is it rocket science to bring in your personal elements (either synths or loops) to expand the song into one of your own. If making your own Soundblock from scratch is monotonous work, all the tools are there to create a custom song 'in the box'.

If you don't want to be a one-man band, the instrument loops can be fun as well as useful. There are four or five riffs from the Memphis Horns included with the free Ignition Soundblock; I loaded them up and started playing, and it seemed I was in Frank Sinatra's *The Man With The Golden Arm*. The Red Hot Blues Guitar Soundblock was just as cool. You can build a song out of the four-bar riffs, or just throw a few of the riffs into an original composition: great fun to play with.

Getting Familiar

Satellite Free, as mentioned earlier, is the cut-down version of Satellite Pro. At this point it will only load Soundblocks, but, as long as you're happy with this and don't need to delve too deeply into programming, it works fine. The Pro version will play back Soundblocks and also any third-party WAV, AIFF and REX/RX2 files, although be aware that it will truncate rather than dither any 24-bit files, so you'll need to prepare such files accordingly before loading them in.

The Satellite screen is divided into four areas. The upper-left quadrant provides global screen selection, as well as loading and saving features, while the Producer and Soundblock icons section shows you what's loaded at a glance.

What displays in the upper-right section is determined by the three buttons at the top of the panel, which are labelled Control, Info and FX Info. The first, Control, displays eight control knobs. These have fixed assignments in the Free version but can be reassigned via ▶

drop-down menus in Satellite Pro. Typically, the top four knobs control the amplifier ADSR envelope, while the four below deal with the filters and volume.

The Info button replaces the Control section with information about the currently loaded Soundblock, including the type of patch and any loop information, such as the bpm speed or key. Patches come in several different flavours: Instruments, which can be basic synth voices or looping Instruments with integral MIDI sequences; Sliced Recycle-style Loops; and normal Loops and Phrases. Satellite can determine the tempo of the loop (if any) and suitably stretch the file to match the current song tempo.

The FX Info button brings up the effects screen. There are four effect slots: two channel sends and two FX buses. The effects themselves are quite extensive and include the usual suspects, from delays, chorus, flangers, phasers, rotary effects and reverbs to filters (including a wah-wah), distortion and bit-crushers. There are also channel-type effects like EQ, compression, limiting and auto gates/panners for the mixer. There are 119 presets and nine open patch slots, but Satellite Pro also lets you save FX settings within Multis. The effects are quite usable, though they do seem to be geared towards electronic music. You won't want to toss your high-end software or outboard overboard, but for adding a little reverb to acoustic samples, or effects to your synth sound, they work fine.

The bottom two thirds of the Main page are occupied by the Multi and Patch Selection module on the left and another switchable pane on the right. The Selection module lets you explore the loaded Soundblock's Multis and associated Patches using the selector arrows. There are 16 slots for the Multis' Patches, each corresponding to a Mixer channel. The mixer is a virtual audio mixer, rather than a MIDI mixer, and each Patch can have its own MIDI channel or share one with another mixer channel's Patch, which makes it easy to stack and layer sounds.

The mixer itself is a fairly straightforward affair, as you can see from the screen on the first page. However, there are three further views, namely Details, Browse and List, which reveal controls for some additional functions.

Details is where you can arrange key-mapping, set tuning and adjust the gain, polyphony and transpostion of patches. It also allows you to determine whether a sample will be stretched and if so, whether in pitch and/or in time — and you can then adjust the 'grain' size, which has eight different settings, going from nice and smooth to seriously lo-fi.

The Browse and List pages let you find and load in individual Multis or Patches from your hard drive and then save them as personalised Multis. Once you've loaded in your Multis and/or patches, the List page displays them and you can rename, delete, and generally muck about with them to make your own Soundblocks. It sounds a bit convoluted to use the three pages to get your Multis and Patches up and organised, but in practice it works well. There is simply no easy way to deal with hundreds of available possibilities and the three pages help keep you focused on the job at hand.

The Pro Advantage

Things start to get really interesting when you turn to the pages that are completely unavailable in the free version of Satellite — and I was surprised to find just how deep the programming possibilities go. The Patch page replaces the main page with more synth controls for mutating your samples: pitch control, filtering, envelopes and LFOs are all here, and in fairly standard forms, but you do have five-stage envelopes instead of the usual ADSRs. The extra stage is a Rise/Fall fader that adds a time variant after the Sustain. For example, the amp envelope fader can either raise the volume back up to full before Release or to zero over time. While not as fancy as multi-stage breakpoint envelopes, this element adds a nice twist.

The filters sound good but do get peaky quickly — if you don't watch out for resonance you can blow a speaker. However, there is envelope and LFO control over both cutoff and resonance, which is always nice and overlooked too often. Unless you are into serious modulation routing (there is no modulation matrix), you should be able to do just about anything you'd want to do: after all, this is sample-based synthesis, not modular virtual analogue.

The Effects page is fairly simple, allowing you to select and edit effects. There are only six parameters to control and you can't change assignments, so once again, tweakheads will have to look elsewhere. The knobs seem to put the most important parameters under your fingertips, but more control is always nice. Once you do get an effect you like, you can save it and use the FX List page (which operates like the main List page) to shuffle and rename all your effects.

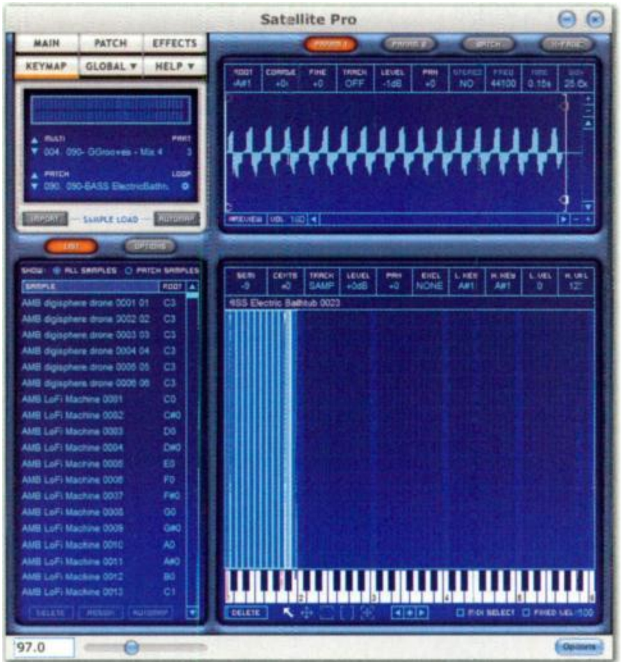
The last Satellite Pro-only Page is the Keymap page, where you import samples, edit them and assign them across the keyboard. The Waveform section in the upper right lets you edit the pitch, volume, pan and start and end points of the selected sample. This process doesn't change the original sample on your hard drive, only how it works in the Soundblock within Satellite.

Once you have the sample whipped into shape you can assign it to the keyboard for key and/or velocity splitting within the keymap zone display at the bottom of the screen. There is a lot going on here — it's a really nice feature and almost worth the price of admission on its own. While Satellite might not be a tweaker's paradise elsewhere, you can get down and dirty here. One note: my Sonar DAW software wouldn't play well with this page and kept locking up, so I simply used the stand-alone version of Satellite Pro for my editing in this area.

Conclusion

Despite putting a world-class sample library at your virtual fingertips, Satellite faces a lot of competition as a sample player. As an integrated loop-player, however, it edges into a different league. It's not quite

The quick and easy interface makes it a breeze to set up a Patch's synthesis settings. If a parameter doesn't have a knob, button or fader you can click on the parameter itself and scroll.



The Keymap page: load in Soundblock samples or (in the Pro version of Satellite) your own samples, and edit them to your heart's content...

the same thing as Ableton Live or Project 5, since it doesn't record, but you can use it within a DAW to make up for the missing elements.

It's easy to massage all your loops and synth samples into Multis for an entire song, and some of the construction kit Soundblocks show just how powerful this ability is. I downloaded both MIDIhead and Patchen's kits and had lots of fun playing through the 10 to 15 Multi 'songs'. Each one of these Multis uses about five loops mapped to adjacent keys, and builds from a basic drum/bass rhythm by adding leads or licks, breaks and change-ups, or just by dropping lines out. Each loop will play for a few bars, and if you can count to four you can mix and match them. The key is to define and refine your presets before playing, rather than loading samples and loops once you're getting into a performance. If you sort that out beforehand, Satellite Pro is very fun and performance oriented.

To sum up, if you're in the market for a sampler, I think Satellite Pro has a lot to offer. While it might not excel in any single area, the whole adds up to more than the sum of the parts: the synthesizer sounds good, the granular sample stretching can get a little grainier than you might want at extremes but is great with reasonable settings, sample editing facilities are well covered, and the price is certainly right, even though the software doesn't come with a large, pre-installed library. Indeed, the last could be a positive point, as the Soundblocks concept should appeal to anyone on a budget or simply looking to save the hard disk space that a modern multi-gigabyte library will take up. Finally, and perhaps most importantly, this software is also lots of fun! SOS

information

S Satellite Pro $149, Soundblocks from $19 to $49.
W www.samplebase.com

Mackie Control Pro

Mackie have updated their popular control-surface system. We find out what's new.

Paul White

The original Mackie Control, which evolved from the earlier Logic Control, will be familiar to many DAW users, but it has recently been replaced by a MkII version called the Mackie Control Pro. The same modules are available as before — the Control Universal Pro master unit, the Control Extender Pro eight-fader expander module and the Control C4 Pro 32-knob controller — but although the functionality and control layout are exactly the same as before, there are some differences worth looking at.

Silver Lining

The most obvious change is that the new units are silver rather than Mackie grey, and have a more steeply raked display section for better visibility. All the buttons and knobs have been redesigned along with the casing, and the front-to-back dimension of the units is now an inch or so less than before, so you get a bit more room on your desk for your computer keyboard. I particularly like the new more 'weighted' jog/shuttle wheel and the straight unit sides that let multiple units sit together in a far more visually appealing way than the rounded sides of the earlier models.

Perhaps the biggest practical difference is that you no longer need a separate MIDI interface to connect the units. The Control Universal Pro, the Control Expander Pro and the C4 Pro still have MIDI ports and can be connected the 'old' way if you prefer, but the Control Universal Pro also has a USB port, allowing it to be connected directly to your Mac or PC without requiring you to load any additional drivers. Three sets of MIDI In/Out ports have been added to the rear panel of the new Universal Pro, and these may be used to link to up to three expander units. In the unlikely event that you want to use more than three expanders, you'll need a separate MIDI interface. As before, each expander needs its own MIDI port — but if you don't need all three MIDI ports on the Mackie Control Universal Pro for expanders, they can be used to connect other MIDI equipment.

It appears that the motorised fader quality-control has been tightened up further, to promote better reliability and to avoid fader-calibration error messages. The external universal power supplies are also a little smaller, although they have the same output voltage and current rating as the originals.

The front-panel screening is still based on Apple's Logic software, with included Lexan (tough plastic) overlays for Pro Tools, Nuendo/Cubase, Digital Performer, Sonar and Tracktion. There are three different modes of operation, the first of which is Mackie Control Universal, offering direct support for Tracktion, Ableton Live, Sonar, Samplitude and Sequoia,

SOUND ON SOUND

Mackie Control Pro

pros
- Ease of connection.
- All the original features.
- Improved display and control visibility.
- Supports most commonly used DAWs.

cons
- An expanded system takes up rather a lot of desk space.

summary
A practical and welcome update to a time-tested control-surface standard.

Alternatives

The Tascam controllers that were the most obvious alternatives to the Mackie Control have now been discontinued, but the slightly more costly Euphonix Artist series of control surfaces or the CM MotorMate offer impressive facilities in a somewhat smaller footprint, albeit taking a slightly different approach. There's also the new Master Control combined controller and audio interface from Alesis. At the lower-budget end, you may like to consider the range of multi-fader/knob controllers from Behringer or the excellent single-fader controllers from Frontier Design and Presonus.

Digital Performer, Reason, SAW Studio, Nuendo/Cubase and Adobe Audition. The second mode is a dedicated Logic controller mode, and the third offers HUI compatibility for use with all variants of Pro Tools. To change modes on both the main unit and the expanders you need to power each of them up while holding down the Select buttons on the first two channels. Your choice is then remembered until you change it.

In Control

After I first connected the system, I powered up each module and chose Logic as my operating protocol. The Mackie Control Universal Pro was recognised instantly but I just couldn't get the expanders to show up. After some head-scratching, it turned out that my MIDI cables (you have to provide your own) were too fat to properly fit through the holes in the Mackie Control rear panels, so I solved this by slimming down the plastic outer shell of the offending plugs with a sharp knife, to expose more of the metal part. This did the trick and allowed me to push the plugs fully home, which rectified the problem.

When connected via MIDI, you can power up the units in the order in which you'd like them to show up in your DAW, but when the master unit is connected via USB it obviously has to be switched on first, so that its MIDI ports will work. In Logic you can simply drag the icons of the controllers into the correct order in the Control Surfaces Preferences setup page, and this arrangment will be remembered once you quit Logic. Other DAWs have their own way of doing things, but the end result is much the same, enabling you to set up the units in any physical arrangement you like and to have that correspond with the way the units are seen by your DAW.

In all other respects, the operation of the units is exactly as in the MkI versions and differs slightly from DAW to DAW. The C4 expander is particularly useful, especially where a separate controller is needed to control plug-ins, as its 32 knobs and four display strips make it easy to keep track of what's being adjusted. Alternatively, you could have its knobs controlling the levels of 32 audio channels, or you can even configure it for tool selection or other frequently performed tasks. The C4 comes bundled with the C4 Commander software, as before, which facilitates 'drag and drop' C4 control mapping to your favourite MIDI hardware, plus you get a free version of Mackie's Tracktion 2.1 sequencing software.

Conclusion

These updated units are more or less the same price as the originals, and not needing a separate MIDI interface is definitely a bonus. Even though the competition has had a few years to catch up, the Mackie Control Pro is probably still the most elegant choice if you want an expandable DAW controller system at this price point. SOS

information

- **£** Control Universal Pro £999; Control Extender Pro £629; Control C4 Pro £869. Prices include VAT.
- **T** Mackie UK +44 (0)1494 557398.
- **E** uk@mackie.com
- **W** www.mackie.com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Pete Townshend writes...

I use Digital Performer, as you may know, and after following Robin Bigwood's directions for MIDI networking [*in Sound On Sound March 2007 and on-line at www.soundonsound.com/sos/mar07/articles/dpworkshop_0307.htm*] my life has changed! Robin, you're a star. As for audio networking, you probably know that Abyssoft, who offer Teleport, now offer Soundfly, which works well with Soundflower, so it gives a fix you might test for one of your next pieces.

Thanks for your ideas and wonderfully clear presentations.

Pete Townshend

Editor In Chief Paul White replies: Thanks for the kind words, Pete. We're all enthusiasts here and keen to share anything we might learn. Thanks also for the tip about Soundfly, which I'm sure Robin will explore further. It's heartening to know that *Sound On Sound* is a useful resource even for experienced professionals such as yourself.

Glitch-proof Vista

I'd like to comment on the 'Vista For Musicians' article in *SOS* June 2007. On page 166, the Multimedia Class Scheduler Service is mentioned, "which should provide bomb-proof audio performance".

In my experience this is over-optimistic. I'm the developer of MultitrackStudio (www.multitrackstudio.com), which supports WaveRT and MMCSS. I've tried hard to demonstrate the blessings of MMCSS, but this isn't easy. The problem is that it only improves glitching caused by low-priority threads. Notorious problem-makers such as Aero painting (or the tool-tip fade effect in Windows XP), however, aren't low priority, so MMCSS doesn't improve this. Even once Aero is completely disabled I haven't been able to demonstrate any benefits. I think MMCSS won't solve anything for people who have audio problems, it will only further improve systems that are already stable.

Giel Bremmers, Bremmers Audio Design

PC Music Specialist Martin Walker replies: Many thanks for the feedback Giel! I deliberately included the word 'should' in the phrase "should provide bomb-proof performance", since over the years I've heard many claims for both software and hardware that aren't borne out in practice. It's only when software developers like you code such features into your products that you find out whether or not such features are of actual benefit.

Microsoft have also gained a reputation over the years for setting down ground rules for Windows developers and then breaking them with other software in their own product range, so it doesn't really surprise me that the potential advantages of MMCSS have been seemingly scuppered by giving their own Aero eye-candy a higher priority than streaming audio.

The many tests that have now been done across a range of PCs by DAW builders have already proved that the Aero interface holds back ultimate audio performance, so most of us looking to run the maximum number of soft synths/plug-ins will disable this anyway. However, it looks as if those musicians looking forward to bomb-proof audio streaming under any circumstances may be disappointed.

Improvements to music tech education

In response to Chris Mayes-Wright's Sounding Off article on music education in the February 2008 issue [*on-line at www.soundonsound.com/sos/feb08/articles/soundingoff_0208.htm*], you may be

The Windows Multimedia Class Scheduler Service (MMCSS) is designed to give priority to multimedia applications, such as Giel Bremmers' MultitrackStudio (shown here), when requesting CPU resources. However, even when MMCSS is implemented correctly, aspects of technology such as Windows Vista's Aero still have higher priority for access to the CPU.

interested to know that there is a body within the pro audio industry, and has been for a while, which supports and advise educational establishments right up to HE level. JAMES (Joint Audio Media Educational Services) was launched around 18 months ago by the educational arms of the APRS (Association of Professional Recording Services) and the MPG (Music Producers Guild). The partnership was designed to supply increased support to our existing course Accreditation Scheme and associated services such as seminars and masterclasses. JAMES also represents the industry's educational needs to government agencies and wider sectors such as the Sector Skills Councils and awarding bodies.

Most significantly, the JAMES Working Group is made up of industry professionals — engineers, producers, studio owners, musicians — people who have worked (and survived!) many years in the recording industry and have a very realistic view of what is required of a person entering into the industry today.

While your article related generally to Secondary Education, your overall comments on provision at grass-roots level, ill-spent budgets and limited technological knowledge due to 'lack of time' all apply right across the educational sector and are areas where JAMES has identified a real need for support, assistance and advice. This is why we have launched the JAMES Educator's Forum (JEF) which will act as a massive support resource to all educators at all levels, whether Secondary, FE or HE. We want it to open up communication between the industry and education and encourage dialogue about equipment, technology, production, courses, careers... in fact, all things audio!

To further support this initiative we are in the process of setting up a network of Regional Centres to enable local industry support for educational activities.

Our recognised course Accreditation facility is now available to FE courses, and with the recent addition of UK Screen to our strategic partnership we have extended the coverage to post-production courses as well. Our master-classes and seminars are consistently popular with students and faculties alike. LIPA (Liverpool Institute for Performing Arts), Glamorgan and Leeds Metropolitan University are recent establishments that have requested a JAMES industry panel to share their views and experiences with students.

More bodies are realising that JAMES really has something valuable to offer to education, our most recent development being discussions with Skillset (Sector Skills Council for the Audio Visual Industries) on working closer with them to provide accreditation and support.

For more general JAMES information, have a look at www.jamesonline.org.uk or if you wish to take advantage of the JAMES Educator's Forum, email us at interested@jefonline.org.uk.

Tony Andrews

News Editor Chris Mayes-Wright replies: Thanks for your comments. I'm aware of the good work that JAMES does in the FE and HE sectors, and it's great to hear about new initiatives such as the JEF and your forthcoming Regional Centres. We'll certainly keep our ears to the ground for all future JAMES-related news! SOS

Playback
Readers' Music Reviewed
Sponsored by breed-media.co.uk

Hours And Minutes
demo

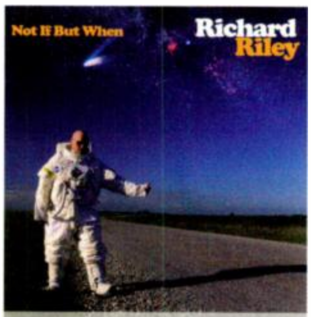
A probation officer by trade, Sam The Scholar has paired up with Afro Rizzy, one of the ne'er-do-wells in his charge, to make music. This is probably a very good way of achieving all the things that the probation service is supposed to achieve, so naturally it's a sackable offence on Sam's part. Consequently, Hours And Minutes are forced to conceal their true identities. (Of course, they may have made all this up to conceal the fact that they're actually accountants. I would, if I were a rapping accountant.)

Musically, Sam seems to hail from a guitar-oriented background, while Rizzy is a talented MC with plenty to say. Rock-rap hybrids don't always work, but this one does. The combination of urgent rapping and relentless riffing has a focus and energy that goes a long way towards overcoming the demo's technical shortcomings.

Most of the latter are to do with bass levels, which are skewed enough to suggest monitoring problems in Sam's studio. Not only are the mixes bass-light, with bass instruments mixed too low, but snare and tom sounds lack any of the fatness that can make you feel as well as hear a beat.

Sam Inglis

W www.myspace.com/hoursandminutesmusic



Richard Riley
Not If But When

Hey Sam! This one sounds like a real record. There's live strings, brass, drums, nice-sounding vocals, well-recorded guitars, and proper songs with structure and everything. Richard Riley is pretty good.

Not If But When is a collection of 10 charming and quirky songs written and produced by Richard, which features a bunch of his mates, so it seems. According to the nice letter he included with the CD (and vinyl LP, which was unexpected), he uses mainly hardware, as well as some pretty nice guitars, mics and amps, to create his blend of cosy-sounding and completely inoffensive songs. These songs wrap around you like one of those giant fluffy blankets with the silk trim that you get in hotels — you know the ones I mean, they keep them in the cupboard with the spare pillow.

As for criticism, there's not much to say, although it sometimes sounds as though the drums, which are very well played, are a few tens of milliseconds behind the guitars, which could be the result of sync'ing multiple devices. This brings a certain slackness to some of the tracks, but who's to say that this isn't an artistic decision? Another thing that is a personal annoyance is the lyrical content on some of the songs. It's been said here before that 'ooh yeah', 'uh-huh', and 'woah' are not words, and don't really need to be uttered after every line of every verse. For my liking, the first song, 'Evie', is blighted by too many of these embellishments, and it's made even more cringeworthy by the almost Travis-like repetition of the name Evie, sung in such a way that the entire song sounds like a tribute to a certain auction web site. Apart from this, I really like the album, and might even keep a couple of the tracks on my iTunes.

Chris Mayes-Wright

W www.richardriley.co.uk

Cutscene
demo

If you were on the lookout for the cutting edge of urban music, you probably wouldn't expect to find it in Preston. (Come to think of it, I'm not sure what you would expect to find in Preston, although Wikipedia tells me that it has a very comprehensive bus network.) That fair city is, however, the home of Chris J and Alan M: two producers who, as Cutscene, "suggest that you big up your chest for the North West".

Since no MCs are credited, I'm guessing that Chris and Alan have taken on vocal duties themselves. Either way, the rapping is definitely the best feature of this demo: ferociously intense yet intelligible, natural and unforced.

The duo also have some neat production ideas, although sampling *Monty Python's Life Of Brian* is not one of them. The combination of out-of-tune Spanish guitar and Nine Inch Nails synths on

'Induction' is very effective, and the sound is abrasive and confrontational throughout. There are, though, ways in which it could perhaps be made more effective with some different mix and arrangement choices. For instance, there's something suspect about the overall balance on the first track, as though a 'smile' EQ has been pushed too far, and the contrast between the searingly bright acoustic guitar and the dull-sounding beats leaves the latter feeling small. *Sam Inglis*

W www.myspace.com/cutscenerap

Country & Blues
1) Beatles Without John
2) Don't Give Up On Love
3) She Is
4) American Idol
5) Restless Heart
6) The Party (Tough)
7) Senior High
8) After You
9) Same Of Saturday Night
10) Falling
11) Haunting Me (Tough)
12) Here Again - Duet (Tough)

Pop & Rock
13) Things You Said
14) Kate (Tough)
15) South Street (Tough)
16) Find Your Way
17) Sweet Addiction (Tough)

Dave Tough Productions
demo

Diving into a big box of demos can be a test for the will, and unfortunately some packages have a tendency to go straight back in without making it into the CD player. Seventy-four minutes of New Age meditation music? Not today, thanks. And, if I'm honest, probably not tomorrow.

With a dry track listing that divides 18 songs into 'Country & Blues' and 'Pop & Rock', Dave Tough's demo CD is among those that requires some resolve from the listener. He's clearly serious about the business of being a songwriter, though, and thankfully, this has led him to hire some excellent musicians and vocalists. They're put to work on material that is, at best, really good. Few of these cuts would be out of place on a modern country station in the States, even in their demo state.

However, the business of songwriting is a fiercely competitive one, and there are times where Dave's output feels a little too generic to stand out. His songs are never less than supremely competent, but only a few, such as the opening 'Beatles Without John', have the sort of attention-grabbing lyrical or musical quirks that could lift them above the crowd. *Sam Inglis*

W www.davetough.com

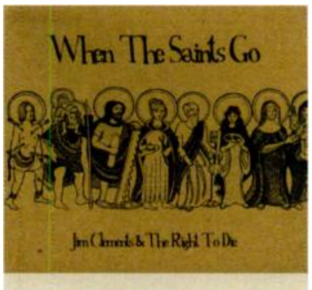
Lookout Joe
demo

If their press release is to be believed, Lookout Joe are simultaneously "emerging from the up-and-coming Coventry scene" and "jumping straight out of university". What's more, they are all "classically trained graduates", and it shows. Like the Ben Folds Five, there are three of them, and they spurn the common-as-muck guitar altogether, instead placing the grand piano at the centre of their sound.

Lookout Joe's technical ability is beyond reproach (unlike their use of brick-wall processing on the master bus), and their enthusiasm is infectious. Whichever of Ed, Ben and Drew is the singer (it doesn't say) has a fine voice, and they're not afraid of variety, with ska, jazz, blues and Latin music all rearing their heads at various points.

Despite the band's obvious talents, though, there's something naggingly unconvincing about their demo. Lookout Joe's stylistic dabblings have something of a dilettante quality, and although the band seem very proud of the fact that "Ed's Latin American travels have a strong influence on the lyrics", the end results have a tendency to sound a bit like a gap-year student's Livejournal entries set to music. *Sam Inglis*

W www.myspace.com/lookoutjoemusic



Jim Clements & The Right To Die
When The Saints Go

It appears that James Blunt has been moonlighting as a man called Jim Clements, and has formed a band called Jim Clements & The Right To Die. How his bandmates didn't notice is a miracle, since Blunt is one of the most annoying men alive.

Anyway, we're limited to how many times we can use James Blunt's name in each issue, for sanity more than anything else, and if we draw too much attention to Blunt's second income, Her Majesty's Revenue & Customs may be alerted, and we wouldn't wish that upon anyone. So on with the review.

Despite the band having the word 'Die' in their name, they aren't a bunch of grubby goths, more a rabble of folky types, with a line-up of guitar, bass, vocals, drums, keyboards and violin. *When The Saints Go* is a concept album of sorts; the songs are all quite 'folk-rock' in arrangement, sometimes verging on the Balkan gypsy style, but all distinctly weird, with James Blunt (sorry, Jim Clements) singing. Oh, and yes, there is a version of 'When The Saints Go Marching In'. It sounds like a ringtone.

On the technical side, I'm not keen on the timbre of the violin — it's clearly electric, and sounds very un-violin-like, except when it's at the very back of the mix, when it seems to blend reasonably well with the other instruments. The piano sound is also fairly weak when it's being played on its own or with little else in the mix. On the positive side, the drums are good; there's quite a lot of marching snare kind of stuff, and the backing vocals work well whenever they're heard. Importantly, Jim/James' guitar sounds sweet and spacious, and in the last song, 'Mayfly', he sounds less Blunt-ish, which has to be a good thing. *Chris Mayes-Wright*

W www.jimclementsandtherighttodie.com

Giovanna Olvera & The Sands
demo

Continuing the geographical theme, we're told that Giovanna Olvera's music has "presence in the Midlands". It also has presence in the treble region, and plenty of it. Something has gone awry with a mix where acoustic guitars and hi-hats sizzle like pork chops falling into the sun, while there's a black hole where the bass guitar should be. Giovanna and her band are all very capable performers, but their music really doesn't suit this sort of over-bright, fatiguing production.

The tuneful, jazz-inflected pop-soul on offer is a little reminiscent of artists like Corinne Bailey Rae, and if only the gentle sunniness of the music was reflected in the mix, it'd be a very pleasant listen. There is, at least, a glimpse of what might have been on the third song, where the offending drum kit and acoustic guitars are dropped in favour of hand percussion and mellow electric chords. I'd love to tell you what it's called, but despite sending three closely typed pages of A4 bumph, her management don't seem to have found space for a track listing. *Sam Inglis*

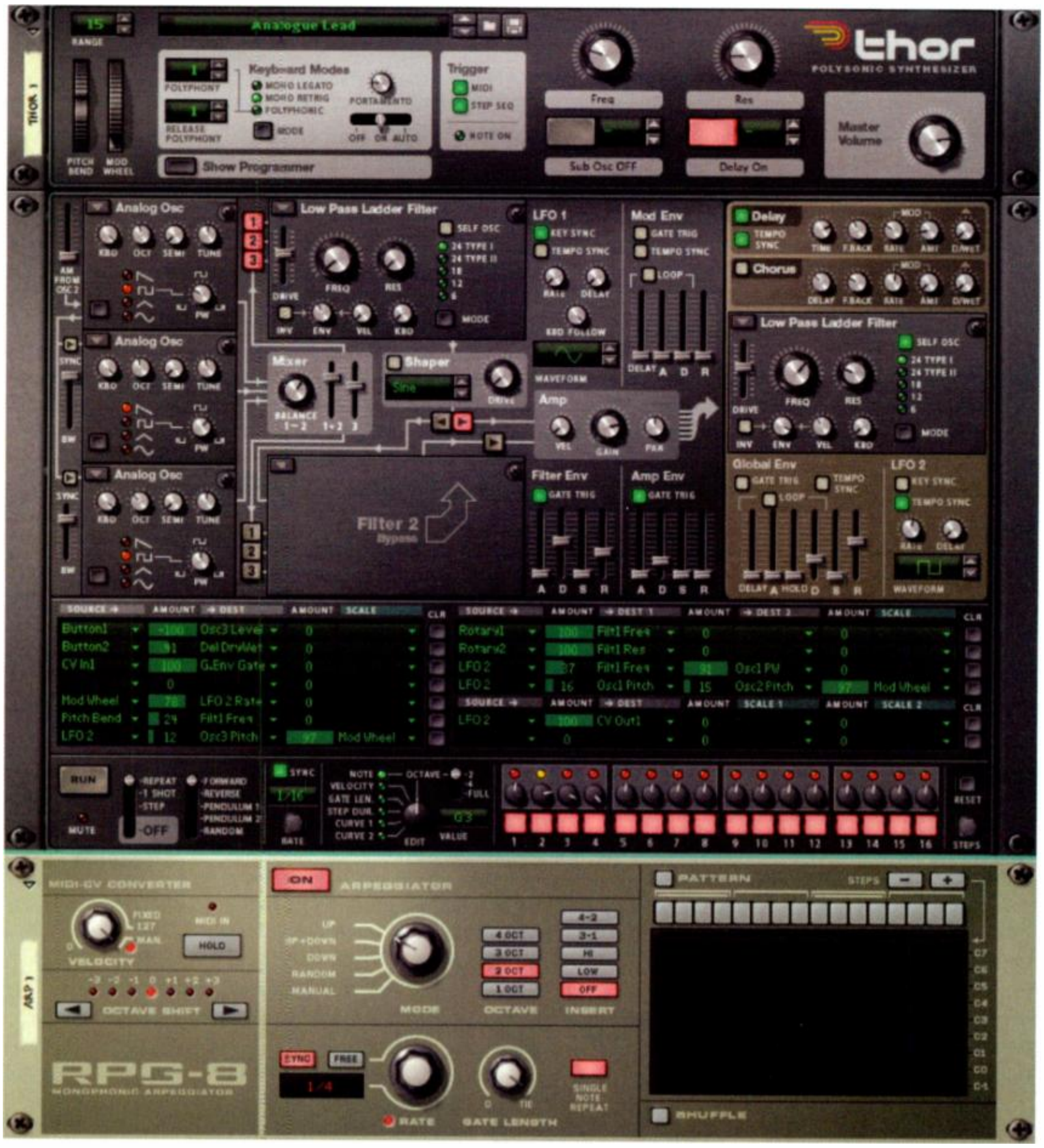
W www.giovannaolvera

Delving beneath the surface of Reason 4's new RPG8 arpeggiator reveals a treasure trove of rhythmic modulations and variations.

Advanced Arpeggiation In Reason 4

Simon Price

For years, an arpeggiator module has been one of the features most frequently requested by Reason users. No surprise, then, that when the Props finally responded they delivered something that goes well beyond the basics. As is often the case in Reason, you can use the RPG8 at two levels: you can let it connect itself to an instrument directly, and just start playing and adjusting the front panel controls. Then, for the tweakheads, there are many less-obvious possibilities for experimenting with advanced settings and creative CV connections. The main purpose of this article is to look at some of the latter, but as this is the first time we've covered the RPG8, let's start with a quick look at its main features.



This is what we'll be working towards: the final patch, with the RPG8 triggering notes and Thor's own sequencer, generating an 'arp within an arp'.

RPG 101

If you select an instrument in the Reason rack, and then add an arpeggiator (Create / RPG8 Monophonic Arpeggiator) the RPG8 will be auto-cabled to the instrument. The RPG8 will also automatically become the live MIDI device in the sequencer, so will immediately start receiving input from your keyboard. The arpeggiator translates your played MIDI notes into an arpeggiated performance, which is sent to the connected instrument by means of CV and Gate signals. As well as the Gate and CV signals, you'll see that there are also Pitch and Mod Wheel connections between the RPG8 and its associated instrument. This is simply to pass through these controller sources, because your MIDI keyboard is no longer pointed directly at the instrument.

The main panel controls are mostly quite easy to understand, especially if you experiment with them to see what their effects are. The central Arpeggiator panel houses familiar controls found on most arpeggiators. The Mode knob selects the order in which order the held notes are played: Up, Up and Down, Down, Random, or Manual (which plays the notes in the order in which they were first played). The Octave buttons let you choose how many octaves the sequence will be extended to.

The Insert buttons change the pattern of played notes. 'Off' results in traditional arpeggiation, with the notes cycled in the order set by the Mode knob. 'Low' produces a sequence that alternates between each held note and the lowest note held; 'High' is the opposite of this; and '3-1' and '4-1' create sequences that transpose each cycle, following a 'three (or four) steps forward, one step back' pattern that rises or falls depending on the Mode. These last two are particularly good when the Random Mode is selected, as the result is an ever-changing sequence.

The Rate control sets the speed of arpeggiation, which can be sync'ed to the song tempo, or switched to Free mode. Gate Length sets how long each note in the sequence is held.

Pattern Recognition

While the Mode and Insert sections determine the melody of the arpeggiated sequence, the Pattern section lets you vary its rhythm. The display in this section, which looks like a miniature, multi-octave version of the Matrix device, is always active, showing a visual representation of the sequence that is playing. To use the Pattern function you need to switch it on with the small grey button at the top of the section. The 16 step buttons and pattern length controls create a rhythmic pattern of On or Off steps. The important thing here is that the note sequence is independent of the Pattern; the pattern adds rests cyclically, but doesn't restart the note sequence each time the pattern loops.

RPG8 Trickery

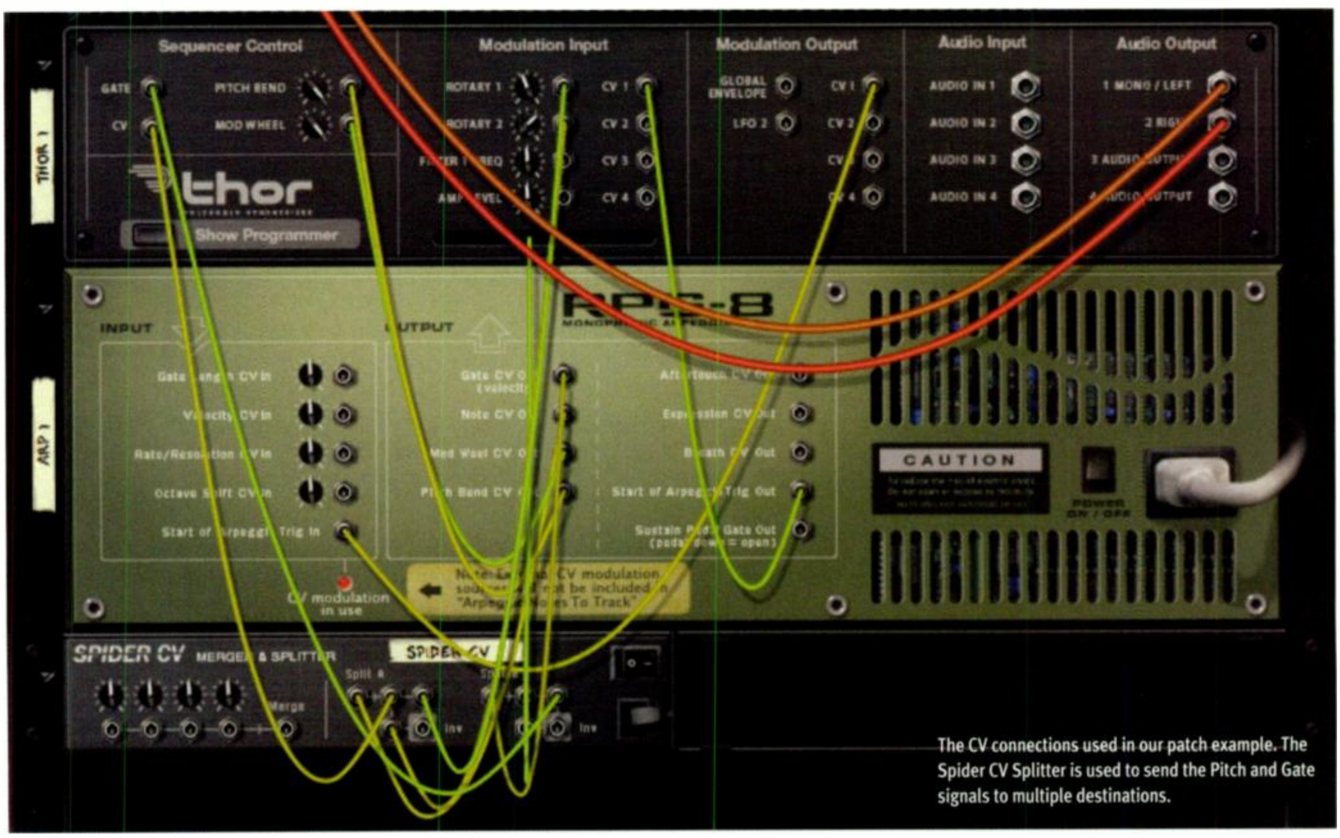
As I said, the best way to grasp the basic arpeggiator functions is to play with them rather than read about them, so let's move on to some more unusual tricks you can try with the RPG8. First, let's get an RPG8 running with the new Thor synth. Simply create a Thor, then create an RPG8, and everything will be set up for you. In the following examples, I'm using a Thor patch called Analogue Lead, which can be found in the Synth Lead folder in the Reason Factory Sound Bank.

The simplest trick with the RPG8 is to send the Pitch CV to other destinations on the connected instrument. To start doing this you'll need to split the CV signal, using a Spider CV Merger and Splitter (as in the screen below). The auto-cabling system splits the CV and Gate signals for you, assuming this is what you want. For now, we're only interested in CV, which appears on Split 'A'. As we've seen in previous articles, Reason's CV system is all normalised, so CV signals are interchangeable between different parameters, including note values. Thus if we send the Note CV to the filter we can control the full range of the filter from MIDI notes. In the picture below I've actually connected the CV to Thor's Rotary 1 input, as this control is mapped to the filter over a predetermined range set by the patch.

Now, the filter frequency will change with the notes as the arpeggiator plays. Try altering the CV input trim and using a large octave range for the arpeggiation, to create a more dramatic result. But couldn't this have been achieved with Thor's Modulation Matrix? Well, yes, up to a point, but if you try it you'll ▶

Modulation Only?

Once you've experimented with using RPG8's CV outputs as modulation sources, it's a natural step to consider using them purely for modulating an instrument, without using the notes generated at the same time. For example, you might want to use a synth polyphonically, playing pad chords for example, while still using an RPG8 to modulate the synth based on the notes you are playing. This presents the same technical challenge we've encountered before with vocoding, because you're trying to connect your MIDI keyboard to two places at once. The solution is the same: put both devices into a Combinator, which can receive the notes and pass them on to both the arpeggiator and the synth at the same time. You'll need to make sure that note CV is not connected between the RPG8 and the sound source, and that Receive Notes is checked for both devices in the Combi Programmer panel.



The CV connections used in our patch example. The Spider CV Splitter is used to send the Pitch and Gate signals to multiple destinations.

The adapted Thor patch, with Filter 3 triggered by the arpeggiator, and LFO 2 triggering restarts of the arpeggio sequence.

find it has a limited range and different characteristic to using the CV method. This way is also quicker and saves Modulation Matrix slots, which we'll need shortly.

Before moving on, take another CV split of the note output to Thor's Rotary 2 input. The Resonance now changes with the notes of the arpeggio, too, with pleasing results.

Trigger Happy CV

Next, we'll add some rhythmic interest using a Gate signal. Connect the 'Start of Arpeggio Trig Out' port on the back of the RPG8 to the CV1 input on Thor. This output sends a Gate signal every time the arpeggio starts a new cycle. As we've connected the signal to one of Thor's general-purpose control inputs, we'll need to use the Modulation Matrix to route the Gate within Thor. Open up Thor's Programmer Panel and locate the first empty slot in the Matrix. In the Source field, choose CV Input / 1. For the destination select Global Env / Gate, and set the Amount to 100. Now, add a Low-pass Ladder Filter in the Filter 3 slot, and set it up as it is in the screen above. Each time the arpeggio restarts, the Global Filter's envelope will be triggered. This instantly creates more movement, as both the individual notes and the overall melodic loop have independent filters.

A lot of sonic variation is available in this patch. Use the Global Env Hold, Decay and Release sliders to alter the movement within the arpeggio. Vary the RPG8's Octave range and the number of keys held down to change the length of the sequence and how often the filter envelope triggers. Varying the filter frequency, resonance and envelope amounts for Filters 1 and 2 all create different results that interact with one another.

One thing this patch highlights is that arpeggio sequences vary in length, depending on how many notes are held and the number of octaves set for the range. Sometimes it's preferable to have an arpeggio with a regular cycle, regardless of these factors. In the above example, this would create a regular, rhythmic pulsing as Filter 2 is triggered. It's easy to force an RPG8 to do this, by re-triggering the arpeggio from an external source. In the picture on page 175 you can see a connection from the CV1 output of Thor to the 'Start of Arpeggio In' port on RPG8's rear panel. I've used the Modulation Matrix to send LFO 2 to the CV 1 output. Finally, I've set the LFO to a pulse wave, and sync'ed it to tempo. Now, the LFO can re-trigger the arpeggio at any division of the tempo. In this example I've used the LFO from the instrument connected to the RPG8, but there's nothing to stop you using any other gate signal, such as drum triggers from a Redrum, to re-trigger the arpeggio in other rhythmic ways.

Patterns Within Patterns

The last trick this month is to use both the RPG8 and Thor's step sequencer to create a double pattern effect. The RPG8 will play an arpeggio at a moderate speed, and the step sequencer will play multiple notes for each note triggered by the arp. First you need to connect the RPG8's Gate output to Thor's sequencer. The Gate should already have been split via the CV splitter when you first connected it. Simply take another of the split outputs from 'Split B' on the back of the CV splitter and connect it to the 'Gate In (Trig)' port on the back of Thor. Now set up the front panels as in the screen on the first page.

The RPG8 has a slow rate (quarter notes). Each note triggers Thor's step sequencer, which is set to 16th notes, and has its first four steps adjusted to play a simple rising sequence. The resulting sequence is a combination of arpeggio and step sequence — like an arpeggiated arpeggio! For the best results, set a long arp gate length so that all the step sequence notes are triggered, and reduce the LFO 2 rate so that the arpeggio has a chance to play before being reset. You will also need to switch on the Step Seq button in the Trigger section of Thor's main panel.

Hopefully the examples here show that the RPG8 offers unique modulation possibilities that could not be achieved with Thor's internal sequencer or a Matrix step sequencer. This is largely because the arpeggiator generates step sequences in real time that change every time you play different notes and chords. What's more, the modulation is directly related to the note sequence that is being generated, creating engaging sequences with bags of natural movement. SOS

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Building Bass Sounds With Prologue

There's more to Prologue than its presets — and it doesn't take long to create some excellent and unique patches.

John Walden

Some powerful VST synthesizers are now included in Cubase, and although they contain various great presets it's well worth delving deeper to create your own patches. This month I'll explain how you can start to make custom bass sounds using the Prologue synth.

Prologue To Prologue

Before I start, it's worth reading the comprehensive technical information about Prologue in the Plug-In Reference PDF — and the signal-flow diagram at the end of the VST Instruments section is particularly helpful.

Prologue is a virtual subtractive synthesizer that has three oscillators, each of which offer the same selection of waveforms — with the common sawtooth, parabolic, square, triangle and sine waves supplemented by some more unusual options. The balance between the three is set via the Osc 1, Osc 2 and Osc 3 rotary controls, and setting any of these controls to zero mutes the output of that oscillator.

Oscillator 1 can be considered the 'master' oscillator that determines the fundamental pitch for the overall sound. Oscillators 1 and 2 both feature wave modulation, and Oscillators 2 and 3 offer frequency modulation. This isn't the place to go into the intricacies of various synthesis technologies (see the 'Waving At The Freqs' box below, or for a comprehensive explanation try Paul Wiffen's 'Synth School' series starting in *SOS* May 1997, or Gordon Reid's epic series 'Synth Secrets' that started in *SOS* May 1999). Wave modulation and frequency modulation both modify the fundamental waveform being used, allowing you to create a greater range of sounds.

The outputs from the oscillators are combined prior to processing through Prologue's single filter section, which offers a good selection of low-pass, band-pass, high-pass and notch filter types, and a number of further controls that include the usual frequency cutoff and resonance (labelled 'Emphasis' in Prologue). Further sound-sculpting can then be performed using the LFO, envelope and effects sections.

The main Prologue user interface.

Good Vibrations

So which of Prologue's waveform types work best for bass sounds? Trawling through the bass presets shows that sawtooth and square waveforms are often used as a starting point. The mini displays for each waveform type show why: while they contain lots of harmonics, both square and sawtooth waveforms are dominated by the fundamental frequency of the note being played (compare these with the Neg Slope 9 waveform, for example) — which means a healthy dose of bottom end on low-pitched bass notes.

Two presets provide good examples. 'Thinlinebass', based only on a single oscillator set to a sawtooth wave, is one of the simpler bass sounds, but it has been fattened up using wave modulation and gentle distortion. Turning off the wave modulation (using the button beside the waveform display) and the distortion (via the EFX window), allows you to hear just the waveform and filter working. In this case, the distortion was contributing significant additional harmonics to the sound, but the mid range and top end can also be increased by opening up the filter cutoff slightly.

For something a little meatier, look at the 'Tribal Bass' preset, which combines two sawtooth waveforms but, aside from detuning

Waving At The Freqs

Prologue's oscillators offer both waveform modulation and frequency modulation to modify the fundamental sound of the waveforms used. In the case of waveform modulation, a phase-shifted version of the oscillator's output is added to itself, and this can produce a richer-sounding, more complex waveform. If the waveform modulation is itself modulated by an LFO (as can be done via Prologue's LFO window), then the resulting oscillator output will vary with time, producing changes in the sound — often described as giving the sound 'movement' and generally making it seem more interesting.

In contrast, in frequency modulation (FM), the frequency of one oscillator is changed (modified) by the frequency of another. In Prologue, Oscillator 1 is used to modify the frequency of Oscillators 2 and 3, and Oscillator 2 can also be used to modify the frequency of Oscillator 3. Again, the process can result in a more varied and complex range of sounds from the oscillator being modulated.

(both coarse and fine) and the filter, uses almost none of Prologue's other processing options. The detuning provides a fatness to the sound: the coarse detuning gives the sound energy at both low (via Osc 1) and high (Osc 2) frequencies, while the fine detuning provides an almost chorus-like effect. In this case, the pitch of Osc 1 is also controlled by the LFO, resulting in the slight 'wah' element of the sound. Other good preset examples based on sawtooth and square waves include 'Pulse Bass', 'Ultra Fat Bass' and 'Deep Wish'.

While the sawtooth and square waveforms provide excellent starting points for bass patches, a number of other waveforms are also suitable. If you select the parabolic, triangle and sine waveforms, you can see that they contain relatively few harmonics. If used in their raw forms, they tend to produce sounds with little 'brightness'. Other good waveforms for bass include Formants 1, 2 and 3, the Vocal Vowel 'O' and 'U', Partial Not Prime, and the higher-numbered Slope waveforms. Check out the 'E-Bass' preset for an example of what these other waveforms can offer. This patch combines Reso-Pulse 1 (Osc 1) and Formant 3 (Osc 2) waveforms. The latter provides the roundness, while the former is tuned down by an octave, giving the whole sound more 'growl'.

Other waveforms (such as some of the Reso Pulse and Neg Slope types) are less suitable for bass patches, as they're dominated by higher harmonics — though in combination with other waveforms they can help the sound cut through by adding some top-end definition.

Prologue offers a wide range of waveform types, many of which provide excellent starting points for bass sounds.

The 'Tribal Bass' preset uses detuning between Osc 1 and Osc 2 (as indicated by the 'coarse' and 'fine' settings), to achieve a fatter, chorus-like effect.

All Mod Cons

Having configured Prologue with your waveform (or waveforms) of choice, there are plenty of ways to enhance your sound. Both the wave modulation (as used in some of the presets listed above) and frequency modulation can be used to modify your basic ▶

When frequency modulation is selected, the Coarse and Fine tuning controls adjust the frequency ratio (in this case between Osc 1 and Osc 2).

waveform selections. The degree of wave modulation to Osc 1 or 2 is set via the Wave Mod knobs, and generally serves to thicken up the basic sound.

FM synthesis is a more complex beast (read parts 12 and 13 of 'Synth Secrets' if you need convincing of this), and random experimentation is likely to create something as musical as scraping chalk across a blackboard. However, the 'FM Based' preset shows that Prologue's FM capabilities can be useful in creating bass tones. This patch combines two sawtooth waveforms with Freq Mod engaged on Osc 2 (so the frequency of Osc 2 is modulated by the frequency of Osc 1). If the output of Osc 1 is turned down, the almost hollow or bell-like sound generated by Osc 2 can be heard. The Ratio control adjusts the amount of frequency modulation. With FM engaged, the functions of the Coarse and Fine controls change, so they adjust the frequency ratio of Osc 2 relative to Osc 1. Adjust both to the lowest settings and then compare the sound of Osc 2 with Osc 1: the further you adjust these controls from this point, the more dramatic the change in Osc 2's output.

Ring modulation and noise generation provide yet more options prior to the signal reaching the filter. There's little I can add to what is in the Plug-In Reference PDF, but the noise generator is probably of most use when creating percussive sounds, while the ring modulator is often used for bell-like sounds (as demonstrated by the 'Bell Mondo' preset).

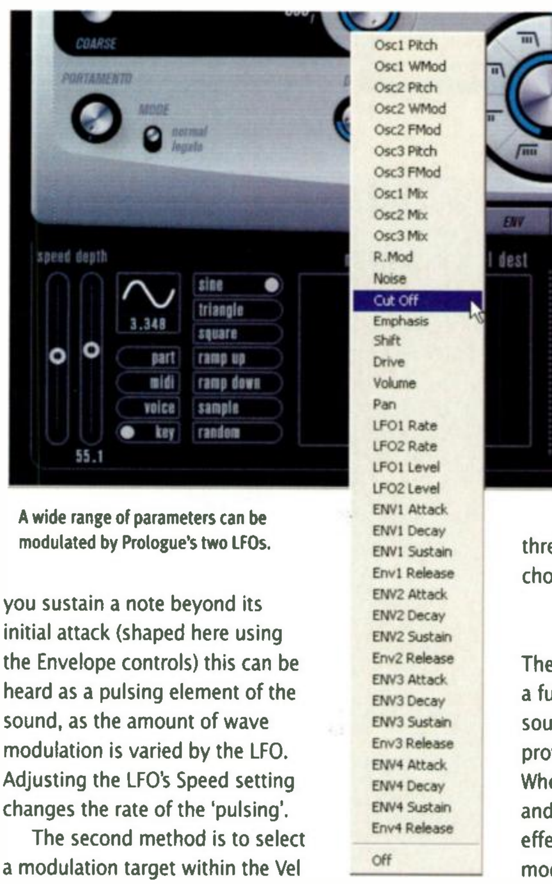
Colour Me Bass

Of course, Prologue's sound-shaping functionality goes beyond the basic choice and combination of oscillator waveforms. The filter is obviously a crucial tool here, and it sounds excellent. Again, its controls are well described in the Plug-In Reference PDF, so I won't retread that ground: for bass sounds, it is sufficient to note that the majority of the preset patches feature one of the low-pass filter types, and raising the filter cutoff will produce a brighter bass tone.

There are still further tools at your disposal and the controls within the LFO, Env, Event and EFX windows offer plenty of scope for adding more expression to your patches. By way of example, let me explain how you can use the LFOs to add extra colour and interest to your sound.

Prologue has two identical LFOs, each with a choice of waveforms and the ability to change both the speed and the way the trigger of the LFO syncs to notes played. The LFOs are used to modulate (either subtly or more dramatically) some of Prologue's other controls. For example, you might set the LFO to vary the filter cutoff, the output level of one of the oscillators or the amount of wave modulation applied to an oscillator.

There are two ways to do this. First, clicking within the Mod Dest box brings up a list of all the possible targets for the LFO. When a parameter is selected here, the LFO simply modifies the chosen parameter over time, based upon the LFO waveform selected. For smooth, gradual shifts, the sine and triangle waveforms work best. The number beside the LFO destination controls the amount (depth) of modulation applied by the LFO. A simple example of this is supplied by the 'Deep Wish' preset, which applies LFO modulation to the Osc 2 Wave Mod level. If



A wide range of parameters can be modulated by Prologue's two LFOs.

you sustain a note beyond its initial attack (shaped here using the Envelope controls) this can be heard as a pulsing element of the sound, as the amount of wave modulation is varied by the LFO. Adjusting the LFO's Speed setting changes the rate of the 'pulsing'.

The second method is to select a modulation target within the Vel Dest box: the same list of targets is available, but the degree of modulation varies dynamically with note velocity. This can be used to add considerable expression when playing. As an example, try removing the Mod Dest entry in the 'Deep Wish' preset mentioned above and adding the same Osc 2 Wave Mod selection in the Vel Dest box instead. Set the amount to 99 and then play

News

- **VST3 Plug-ins: Leading German software synthesizer and virtual effect manufacturers VirSyn have announced the first VST3 update of a third-party plug-in. With the update to version 1.1, VirSyn's Matrix Vocoder is now available as a native VST3 version for Mac OSX 10.4 and Windows XP from the VirSyn web site at www.virsyn.de/en/E_Home/e_home.html.**
- **Cubase 4.1.2: The catchily-titled Cubase 4.1.2.851 update was recently released, making Cubase 4 officially compatible with Mac OSX 10.5. The update also includes bug fixes and other minor improvements, not least of which is support for Euphonix's new Artist series controllers (reviewed elsewhere in this issue). A similar update is available for Nuendo and Cubase Studio 4. Visit www.steinberg.net/1044_1.html for further information.**

notes of varying velocity. The degree of wave modulation applied to Osc 2 will vary with your playing.

The LFOs open up a huge range of possibilities for creative sound-shaping. Modulating the filter parameters is a great way to add interest — for example, raising the cutoff frequency to give a brighter sound based on the velocity of a note, or increasing the drive to produce greater distortion on louder notes — but there is plenty of scope with other parameters too. If you're looking for something more complex, multiple destinations can be specified for each LFO, and the 'Brett Bass' preset provides a good example: LFO1 is used to apply pitch-shift to all three oscillators, bringing a rich, chorus-like quality to the sound.

Epilogue

The Envelope controls provide a further means of sculpting your sounds, while the Event settings provide real-time control via the Mod Wheel, note velocity and aftertouch — and all this can be topped off via effects such as distortion, delay and modulation in the EFX window (or, of course, via plug-ins inserted in Prologue's output channel).

All this only scratches the surface of what is possible with Prologue — even for producing bass sounds. It is equally adept at pads, percussive sounds, leads and synthetic strings or electric piano sounds, so perhaps we'll return to some of these in future. Until then, happy programming... SOS

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Customising SFZ Files

Making More of Sonar's Synth File Format

Understanding how to program SFZ files opens up lots of ways to better exploit what Sonar's bundled soft synths have to offer. Don't panic: it's easier than you might think...

Craig Anderton

Sonar 7 is positively bristling with soft synths, including z3ta+ (the subject of last issue's workshop article), the TTS1 GM module, DreamStation, Pentagon I, PSYN II, Roland GrooveSynth, and the underrated Cyclone. And there's another group of soft synths, including DropZone, RXP, SFZ, Session Drummer 2 and LE versions of Dimension and Rapture, that all have something in common: they can load SFZ-format files. Should you care? Yes, you should, and here's why.

The SFZ File Format

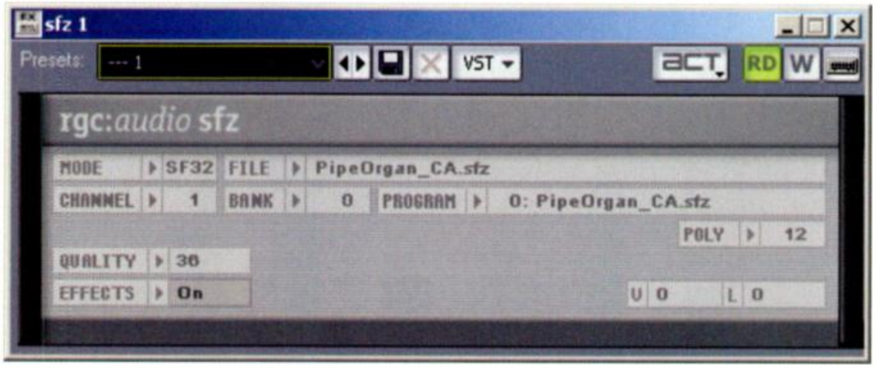
The SFZ file format is not unlike the concept of Soundfonts, where you can load a ready-to-go multisampled sound — not just the samples — as one file. We touched on SFZ files in the Sonar workshop in *SOS* April 2007 (www.soundonsound.com/sos/apr07/articles/sonarworkshop_0407.htm), which described Session Drummer 2 and how to create your own drum kits by collecting various samples, then assigning them to pads with SFZ-format files. However, the SFZ concept goes much further than that simple example, so we'll explore the subject further here.

Unlike Soundfonts, which are monolithic files, the SFZ file-format has two components: a group of samples, and a text file that 'points' to these samples and defines what to do with them. The text file describes, for example, a sample's root key and key range. But it can also define the velocity range over which the sample should play, filtering and envelope characteristics, whether notes should play based on particular controller values, looping, level, pan, effects, and many more parameters.

However, note that not all SFZ-compatible instruments respond to all these commands; if you try to load an SFZ file with commands that an instrument doesn't recognise (possibly due to an SFZ v2 definition file being loaded into an SFZ v1-compatible instrument), the program will generate an error log in the form of a text message. Fortunately, nothing crashes, and the worst that can happen is that the file won't load until you eliminate (or fix, in the case of a typo or syntax error) the problematic command.

It's worth mentioning that the SFZ spec is license-free, even for commercial applications. For example, if you want to sell a set of SFZ-compatible multisamples for use in the Cakewalk synths, you needn't pay any kind of fee or royalty.

Why bother learning about SFZ files? Well, there are three main reasons. Firstly, if you like to create your own sounds you can make far more sophisticated ones for SFZ-compatible instruments if you know how the SFZ format works. The files you create will also load into other SFZ-aware instruments (particularly if you limit yourself to using commands from the version 1.0 SFZ spec). Secondly, by editing SFZ files you can overcome some of the limitations of the LE versions of Rapture and Dimension included in Sonar 7. It's been pointed out that you can't adjust tuning in the LE versions, which can be a real problem if, say, you've recorded a piano track where the piano was in tune with itself, but not tuned to concert pitch and you then want to add an overdub. If you know your way around SFZ, you can edit the tuning of the SFZ file that's loaded into the instrument, and get around the problem that way. Finally, SFZ files facilitate cross-host collaboration. The SFZ Player included in Sonar 7 is a VST plug-in that works with any VST-compatible host, and is available as a free download from www.project5.com/products/instruments/sfz_player/default.asp. The interface isn't fancy, but it will play back SFZ files and it also offers built-in effects and several modes for loading sounds (including streaming from disk).



The SFZ Player is a free download that works in any VST-compatible host, not just Sonar.

As to why this is important, suppose you're using Sonar, a friend is using Ableton Live, and you want to collaborate on a part based on some samples you've grabbed. String those samples together into an SFZ file, have your friend download the player, send the SFZ file to your friend, and you can swap parts back and forth. You can even use really big samples, because the SFZ player supports compressed Ogg Vorbis files. So you can create a compressed, 'draft' version of the SFZ file, then substitute a full version with WAV files when it's mixdown time.

Creating Your First SFZ File

Creating an SFZ file is not unlike writing software code, but don't panic: it's easier

All the opcodes (commands) for the 1.0 version of the SFZ spec are listed and described on Cakewalk's web site.

Opcode List

The following is a description of all valid opcodes for the sfz format version 1.0:

Opcode	Description	Type	Default	Range
Sample Definition				
sample	This opcode defines which sample file the region will play. The value of this opcode is the filename of the sample file, including the extension. The filename must be stored in the same folder where the definition file is, or specified relatively to it. If the sample file is not found, the player will ignore the whole region contents. Long names and names with blank spaces and other special characters (excepting the = character) are allowed in the sample definition. The sample will play unchanged when a note equal to the **pitch_keycenter** opcode value is played. If **pitch_keycenter** is not defined for the region, sample will play unchanged on note 60 (middle C). Examples: sample=guitar_c4_ff.wav sample=dog kick.ogg sample=out of tune trombone (redundant).wav sample=staccatto_snare.ogg	string (filename)	n/a	n/a
Input Controls				
lochan **hichan**	If incoming notes have a MIDI channel between **lochan** and **hichan**, the region will play. Examples: lochan=1 hichan=5	integer	lochan=1 hichan=16	1 to 16

lokey
hikey
key
	If a note equal to or higher than **lokey** AND equal to or lower than **hikey** is played, the region will play.
	
	lokey and **hikey** can be entered in either MIDI note numbers (0 to 127) or in MIDI note names (C-1 to G9)
	
	The **key** opcode sets **lokey**, **hikey** and **pitch_keycenter** to the same note.
	
	Examples:
	lokey=60 // middle C
	hikey=63 // middle D#
	lokey=c4 // middle C
	hikey=d#4 // middle D#
	hikey=eb4 // middle Eb (D#)
		integer	lokey=0, hikey=127	0 to 127
				C-1 to G9

than writing music! Despite the many commands, you don't need to learn all of them, and the syntax is pretty straightforward. Although you can 'reverse-engineer' existing SFZ files to figure out the syntax, it's helpful to have a list of the available commands, and you can find one at www.cakewalk.com/DevXchange/sfz.asp, or check out Appendix A in the book *Cakewalk Synthesizers* by Simon Cann (published by Thomson Course Technology).

As an example of how the SFZ protocol can dress up a sample, suppose you've sampled a guitar power chord in D and extracted a wavetable from it — a short segment, with loop points added in an audio editor (we'll call the sample GuitWavetable_D1.WAV). It won't sound like much by itself, but let's create an SFZ file from it, and load it into SFZ Player.

Arguably the two most crucial SFZ concepts are Region and Group. Region defines a particular waveform's characteristics, while Group defines the characteristics of a group of regions. For example, a typical Region command would be to define a sample's key range, while a typical Group command might add an attack time to all samples in an SFZ multisample. Another important element is the Comment. You can include comments in the definition file simply by adding a couple of slashes in front of the comment, on the same line; the slashes tell SFZ to ignore the rest of what's on the line.

Here's a suggested procedure for getting started with SFZ files.

1. Create a folder for the samples you plan to use. I called mine GuitarWavetables.
2. Drag the sample(s) you want to use into the folder. In this example, I used only one sample, to avoid complications.
3. Open a text editor such as Notepad (the simpler the better — you don't need formatting adding extraneous characters to the underlying text file). If you use a word processor like Word, save the file as plain MS-DOS text.
4. Add some comments to identify the SFZ file, such as:

```
// SFZ Definition File
// Simple Guitar Wavetable File
```

5. Let's turn this wavetable into a region that spans the full range of the keyboard. To do this we need to add a line that specifies the root key and the key range, and tells the file where to find the sample. Here's the syntax:

```
<region> pitch_keycenter=D1
lokey=C0 hikey=C8
sample=GuitWavetable_D1.WAV
```

Pitch_keycenter is the root key, lokey is the lowest key the sample should cover, hikey is the highest key it should cover, and sample defines the sample's name. As the definition file and sample are in the same folder, there's no need to specify the folder that holds the sample. If the definition file is outside the folder, change the 'sample=' line to include the folder, as follows:

```
sample=GuitarWavetables\
GuitWavetable_D1.WAV
```

6. Save this text file, under the file name you want to use (for example, 'GuitarPowerChordWave'), in the GuitarWavetables folder. You could save it anywhere, but if you do it this way and you move the folder, the text definition file and samples move together.
7. Open an SFZ-compatible instrument — say, Dimension LE. Click in the Load Multisample window that says 'Empty', then navigate to the desired SFZ file.
Double-click on it, and now you should hear it when you play Dimension. If you don't, there might be a typo in your text file; check any error message for clues as to what's wrong.

Going Further

OK, we can play back a waveform... big deal. Let's make things more interesting by loading two versions of the same waveform and detuning them slightly. This involves adding a tune= description; we'll tune one down five cents and the other up five cents. Here's how the file looks:

```
<region> pitch_keycenter=D1
lokey=C0 hikey=C8 tune=-5
sample=GuitWavetable_D1.WAV
<region> pitch_keycenter=D1
lokey=C0 hikey=C8 tune=5
sample=GuitWavetable_D1.WAV
```

Now let's pan one of the waveforms towards the right and the other towards the left. This involves adding a descriptor of 'pan=', where the value must be between -100 and 100.

Next up, we'll add one more version of the waveform in the centre of the stereo image, but dropped down an octave to give a big bass sound. We basically add a line like the ones above, but omit tune= and add a 'transpose=-12' command.

Loading the SFZ file now loads all three waveforms, panned as desired, with the middle waveform dropped down an octave. But it sounds a little buzzy for a bass, so ▶

CUSTOMISING SFZ FILES

let's add some filtering, with a decay envelope. This is a good time for the <group> function, as we can apply the same filtering to all three oscillators with just one line. And here is that line, which should be placed at the top of the file:

<group> fil_type=lpf_2p cutoff=300
ampeg_decay=5 ampeg_sustain=0
fileg_decay=.5 fileg_sustain=0
fileg_depth=3600

Here's what each function means:

fil_type=lpf_2p
This indicates that the filter type is a low-pass filter with two poles.
cutoff=300
Filter cutoff in Hertz.
ampeg_decay=5
The amplitude envelope generator has a decay of five seconds.
ampeg_sustain=0
The amplitude envelope generator has a sustain of zero percent.
fileg_decay=.5
The filter envelope generator has a decay of 0.5 seconds.
fileg_sustain=0
The filter envelope generator has a sustain of zero percent.
fileg_depth=3600
The filter envelope generator depth is 3600 cents (three octaves).

As you work with SFZ files, you'll find they're pretty tolerant. For example, the sample names can have spaces and include any special characters other than '=', and you can insert blank lines between lines in the SFZ definition text file. But one inviolable rule is that there can't be a space on either side of the '=' sign.

Overcoming LE Limitations

Rapture LE and Dimension LE are useful additions to Sonar 7, but obviously they have limitations compared to the full versions. For example, with Dimension LE you can edit two stages of DSP, a filter, and some global FX — and no other parameters such as tuning, transpose and envelope attack. However, if the sound you want to load into either of these LE versions is based on an SFZ file, you can modify it well beyond what you can do with the instruments themselves. (Note that these instruments often load simple WAV or other file types instead of the more complex SFZ types; in this case, editing becomes more difficult because you have to first turn the WAV file into an SFZ file, and if you're going to put that much effort into programming, you might want to just upgrade to the full versions that have increased editability.)

Let's look at a Dimension patch, 'Hammond Jazz 3'. This loads an SFZ file called 'Hammond jazz.sfz', so it's ripe for editing. We'll take that Hammond sound and turn it into a pipe organ by creating two additional layers, one an octave above and one an octave below the original. We'll pan the octave-higher and main layers right and left respectively, with the lower octave in the middle. Then we'll tweak attack and release times, as well as adding some EQ.

To find the SFZ file, go to C:\Program Files\Cakewalk\Dimension LE\Multisamples\Organs and open 'Hammond Jazz.sfz' in Notepad. Here's what it looks like:

<region> sample=Hammond Jazz\HBj1slC_2H-S.wav key=c3 hikey=f3
<region> sample=Hammond Jazz\HBj1slC_3H-S.wav key=c4 hikey=f4
<region> sample=Hammond Jazz\HBj1slC_4H-S.wav key=c5 hikey=f5
<region> sample=Hammond Jazz\HBj1slD_5H-S.wav key=d6 hikey=f6
<region> sample=Hammond Jazz\HBj1slC_6H-S.wav key=c7 hikey=f7
<region> sample=Hammond Jazz\HBj1slF#1H-S.wav key=f#2 hikey=b2 lokey=c1
<region> sample=Hammond Jazz\HBj1slF#2H-S.wav key=f#3 hikey=b3
<region> sample=Hammond Jazz\HBj1slF#3H-S.wav key=f#4 hikey=b4
<region> sample=Hammond Jazz\HBj1slF#4H-S.wav key=f#5 hikey=c#6
<region> sample=Hammond Jazz\HBj1slF#5H-S.wav key=f#6 hikey=b6

This shows that the SFZ definition file basically points to 10 samples, with root keys at various octaves of C or F#, and spreads them across the keyboard as a traditional multisample. Note that it doesn't use the 'pitch_center=' statement, for two reasons: first, because Dimension LE doesn't recognise it; and second, because the 'key=' statement sets the root key, low key, and high key to the same

Click in the Load Multisample field in Dimension or Rapture and a Load Multisample browser will appear; navigate to what you want to load. The Garritan Pocket Orchestra samples for Dimension LE are a rich source of SFZ files.

value. You can add modifiers to this, like 'lokey=' and 'hikey=' statements, as needed.

Before this block of <region> statements, add a <group> statement as follows, to modify all of these regions:

```
<group> ampeg_attack=0.2
ampeg_release=2 pan=-100
eq1_freq=4000 eq1_bw=2
eq1_gain=20
```

These parameters add an amplifier EG attack of 0.2 seconds, amplifier EG release time of two seconds, pan full left, and one stage of EQ (with a frequency of 4kHz, a two-octave bandwidth, and 20dB gain).

Now we'll add another region an octave lower, and put a similar <group> statement before it. We'll simply use one of the existing samples and, to minimise memory consumption, as this sample plays more of a supporting role, we'll just stretch it across the full keyboard range.

```
<group> ampeg_attack=0.2
ampeg_release=1 transpose=-12
eq1_freq=2000 eq1_bw=4
eq1_gain=20
<region> sample=Hammond
Jazz\HBj1slC_4H-S.wav key=c5
lokey=c0 hikey=c8
```

The group statement is very similar to the previous one, except that the sample has been transposed down 12 semitones, the pan statement is omitted, so the sample pans to the centre, and the EQ's centre frequency is 2kHz instead of 4kHz. The sample's root key is C5, and the sample is stretched down to C0 and up to C8.

Next, we'll add the final new region, which is an octave higher. Again, we'll put a <group> statement in front of it.

```
<group> ampeg_attack=0.2
ampeg_release=1 transpose=12
pan=100
<region> sample=Hammond
Jazz\HBj1slC_4H-S.wav key=c5
lokey=c0 hikey=c8
```

The group statement adds the familiar attack and release, but transposes up 12 semitones and pans full right. The region statement takes the same sample used for the octave-lower sound and stretches it across the full keyboard.

I should add that although we've made a lot of changes to the SFZ file, it's still being processed by Dimension LE's 'Hammond Jazz 3' patch. As a result, if you take this SFZ file and load it into SFZ player, it won't sound the same because it won't be using the various Dimension parameters that are part of the 'Hammond Jazz 3' patch.

Are We There Yet?

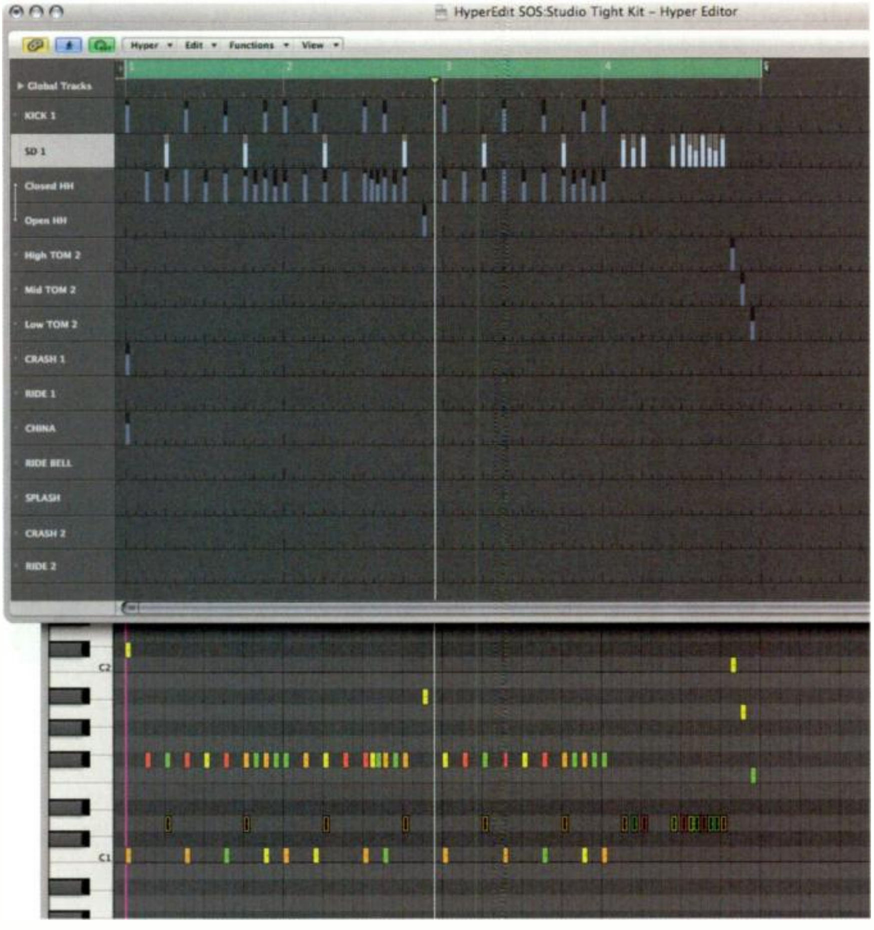
Explaining all this may make creating SFZ files seem complex, but really it isn't, if you have the opcodes in front of you, and the process soon becomes second nature.

For example, I found an SFZ bass patch that produced a great clav-like sound when I transposed it up a couple of octaves, but the attack sounded cartoon-like when transposed up that high. So I just used the 'offset=' command to start playback of the samples past the attack. And while I was at it, I added a very short attack time to cover up the click caused by starting part-way through the sample, and a decay time to give a more percussive envelope. The editing took a couple of minutes at most; I saved the SFZ file so I could use this particular multisample again.

Creating SFZ files might not replace your favourite leisure activity, but it's a powerful protocol that's pretty easy to use. Modify some files to do your bidding and you'll be hooked! SOS

Hyper Editing
Logic's Best-kept Secret

Logic's Hyper Editor is a powerful way of creating and manipulating MIDI data, but it's often overlooked. We give you the low-down in this advanced workshop, and suggest some ways of using the Hyper Editor in your music.



Logic's Hyper Editor (top) shows you far more information than the Piano Roll (bottom), so is very handy for programming drums. In the Hyper Editor, each drum is listed down the left-hand side, with event information displayed along the conventional timeline.

John Moores

Logic has always been an excellent MIDI-programming and editing package, offering several different editors which allow you to create and modify MIDI data. Each offers a view of the data in a different form and some editors offer particular advantages in certain circumstances. Quite often it's useful to work with a combination of editors simultaneously, perhaps using the Piano Roll (formerly known as the Matrix Editor) as a general-purpose MIDI editor, with the Event List for fine-tuning note lengths and the like. But by far the most specialised (and most often ignored) editor in Logic's arsenal is the Hyper Editor.

Basically, it is a display that can show all sorts of MIDI data in separate lanes, called Event Definitions, allowing you to see, create and edit different types of data in one window. It follows the timeline in the same way as the Arrange page does, and displays MIDI events as vertical beams, where the height of the beam represents the value of the event. The special benefit of the Hyper Editor is that each lane can have its own timing or quantisation grid, and this grid can be changed at any time to allow for different quantisation to be applied to new events without affecting the data that is already there.

To make life easier, the lanes can be organised into so-called Hyper Sets, and you can quickly create your own, to display the information the way you'd like. By default, there are two Hyper Sets pre-defined: MIDI Controls (to display parameters such as MIDI CC, Pitch Bend and Aftertouch) and GM Drum Kit, where each lane is mapped to a particular MIDI note, according to the GM standard drum map.

To open a new Hyper Editor, you simply navigate to the Window menu and select Hyper Editor from the list of options. The default keyboard shortcut is Apple+5.

Hyper Editor For Drum Programming

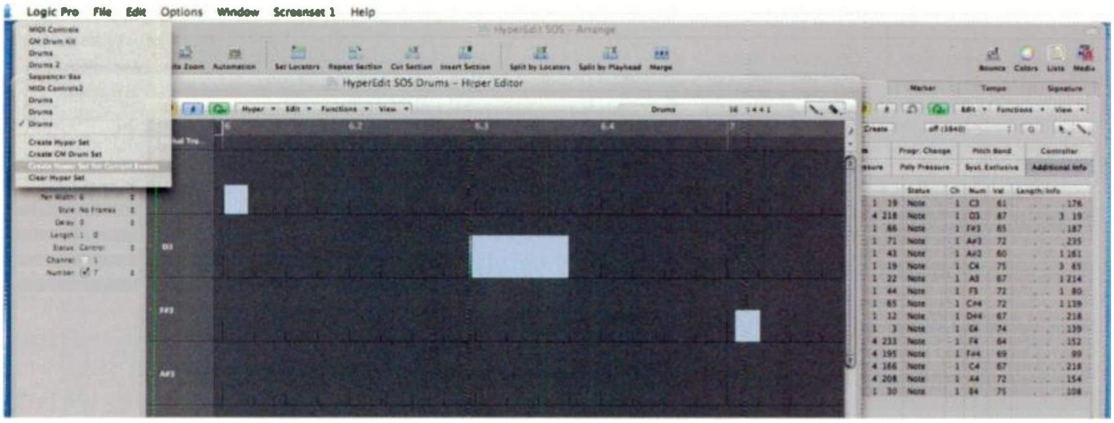
One of the most useful applications of the Hyper Editor is programming drums. But why not use Ultrabeat and its built-in step sequencer? Well, Ultrabeat's step resolution is a global setting, therefore you can't mix up 1/8, 1/16 and 1/16 triplets in a hi-hat part, for example, and neither can you view the sequences of multiple voices at the same time. But you can in the Hyper Editor.

With a Hyper Set laid out to display individual notes in whatever software you're using — Ultrabeat, EXS24 or even a third-party virtual drummer — it's a breeze to program realistic-sounding drums (or machine-like ones if that's what you're after).

The quickest and easiest method for making your own drum Hyper Set is to use or modify the existing GM Drum Kit set, found under the Hyper menu of the Hyper Editor pane. Once open, simply select the event definitions that you don't want in your Hyper Set (shift-click to select more than one lane) and use the Delete Event Definition function, also found in the Hyper menu.

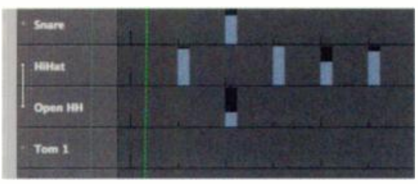
You can also customise the the order of the lanes to your liking, by dragging them up or down the list. Personally, I like my toms to go from high to low (top to bottom) rather than the default, which goes low to high! Also, you might want to duplicate some tracks, as it's sometimes handy to have two lanes set to the same note but with different grid settings, making it easy to create flams and replicate other stick techniques.

Sometimes, you'll need to create your own Hyper Sets from scratch, so let's look at



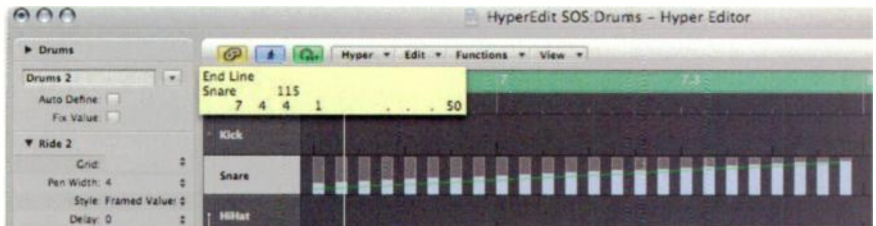
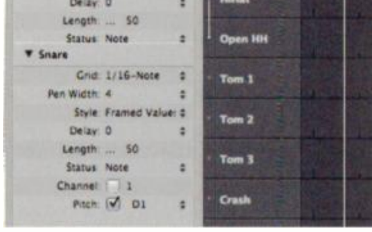
The Create Hyper Set for Current Events command allows you to quickly tailor your Hyper Editor to show only the information that's relevent to your track.

a neat way to define your own. First, in the Arrange window, record in the notes you wish to map out. It helps to play them in the order you want them to appear in the new Hyper Set (though you can move lanes up and down in the Editor later if you need to). In the Event List, select all (Apple+A), and then open the Hyper Editor. In the Inspector area (Hyper Set menu), choose Create Hyper Set for Current Events. This makes event definitions for every note that you recorded (as shown above). In the Hyper menu, choose Select All Event Definitions, then turn to the parameters down the left-hand



Hi-hat mode is designed to ensure that open and closed hi-hat samples don't play at the same time.

side of the Hyper Editor (see the box below for guidelines on the parameters). Set the Pen Width and Style and adjust the note length to around 50 ticks; most drum samples play as 'one-shot', so there's no point in using long notes. I set the default Grid to 1/8 notes, but you can change this at any time. Name each Event Definition, by clicking on the name in the left-hand display. If you have a hi-hat in your virtual kit, you may find the Hi-hat mode useful. This only

Machine-gun-like effects, such as those found in dance music drum parts, can be created using the line tool.

allows one note in the 'group' to be played at the same time: you can't have an open and closed hi-hat happening at the same time on a real hi-hat! Use the small dots on the left of each lane to group tracks using Hi-hat mode.

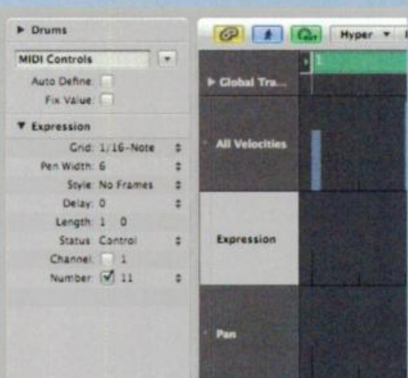
A slightly more long-winded method to configuring your Hyper Set is to use a mapped instrument object in the Environment. This process then automatically names the Hyper Set lanes according to the names in the mapped instrument (and vice-versa), which is very handy, but you may find it's just as easy to stick to Hyper-editing.

Now you can start programming! Using Logic 8's tool priority menus, which are located in the top-right of the Arrange page, I usually set up the main tool as a pencil, with the Command-click tool as the eraser, to make the whole process faster.

You can see by looking at the screenshot on the first page of this article how much clearer it is to see what's going on in Hyper Edit compared to the Piano Roll. It's also very easy to check that you don't have more than four things happening at the same time, which is especially important for drummer realism, unless you have a virtual octopus behind the kit! If less organic drumming is ▶

Event Definition Parameter Guide

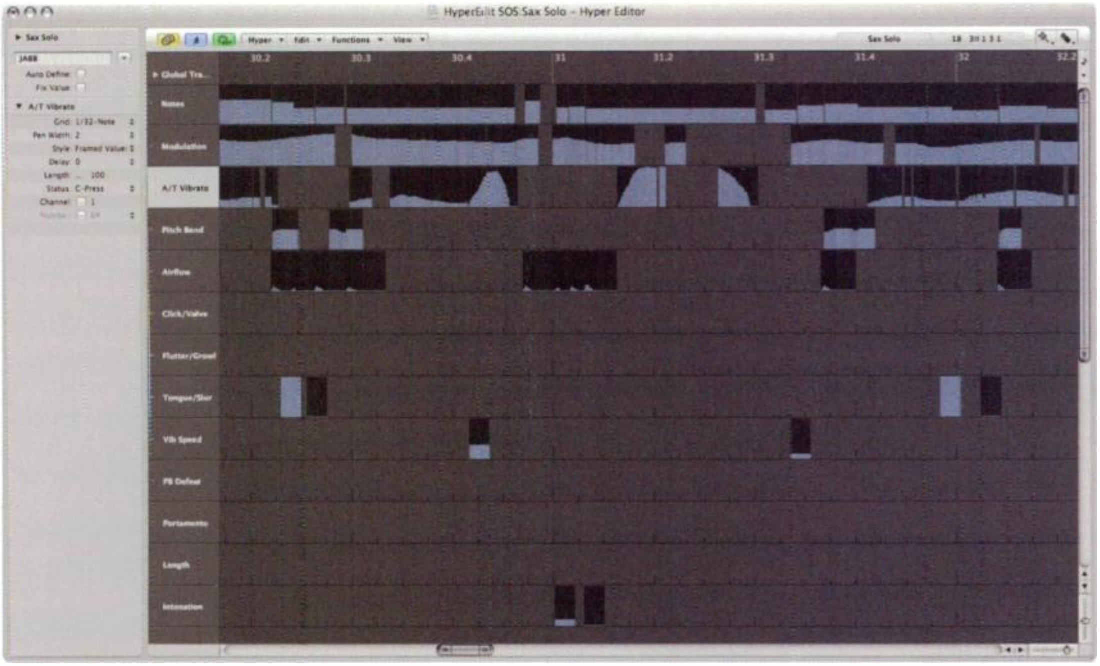
The Event Definition for each lane can be viewed in the Inspector area on the left hand side of the Hyper Editor. Remember that the height of the beam in the lane sets the value of the event, and individual event definitions can be set up by hand, but Logic offers several much faster ways of doing this automatically.



- **Grid: this is where you set or change the quantisation grid**
- **Pen Width: the beams can be displayed at a range of widths, or set to show the length of note events.**
- **Style: The two settings here (No Frames and Framed Values) enable you to view the beams in different ways.**
- **Delay: It's possible to time-shift (delay or advance) MIDI events, which is useful for adjusting the feel of a groove or ensuring that CC messages are sent before the note events.**
- **Length: This sets the length of notes created within the Hyper Editor. The default is 1/16 note, but it can be useful to set this to a shorter value.**

The last three parameters are the most important, where you define the exact type of data to be displayed in the lane. The parameters are context sensitive, so selecting one will affect the options for the others.

- **Status: choose from a menu the type of data: Fader, Notes, CC messages and so on.**
- **Channel: select this if you wish to be specific about which MIDI channel is to be displayed.**
- **Number: selects which particular parameter is being edited (for example, if you are editing MIDI CC events, selecting number 1 will set up the event definition for Modulation).**



Creating a Hyper Set for virtual instruments can help to speed up the programming process. This one is for Garritan's Jazz and Big Band.

your thing, you can use the Line tool to edit events for that 'machine gun' snare fill.

Setting Up Hyper Sets For MIDI Controls

Since events created in the Hyper Editor are automatically set to a timing grid, there are loads of useful and creative applications for MIDI control messages, pitch-bend messages and so on. You could use 'instant' pitch-bend messages to simulate a hammer-on or pull-off on a guitar part or a bass guitar part, or to create a more realistic guitar vibrato (which, of course, only goes up in pitch). An 'all notes' lane is handy here, especially when it displays the correct note length, but you may want to call up the Event List so that you can fine-tune the data. Being able to see all the required MIDI data on the screen makes it much easier to add events at the right place.

Rhythmic gated effects are also a breeze. But since MIDI CC 7 and 10 (volume and pan respectively) are reserved by the main channel strip controls, it's best to use MIDI CC 11 (expression) to automate volume effects, so that you can still use the main volume fader for adjusting levels whilst mixing. Using some of the swing grid settings here can add interest to otherwise mundane parts.

Some third-party instrument plug-ins, such as Garritan Jazz and Big Band and Personal Orchestra, use particular MIDI Controller numbers to control the articulations. I would recommend making a Hyper Set especially for these. Again, recording some automation data, then using the 'Create Hyper Set for Current Events' function is an easy way to do this.

Internal Hyper Control

Of course, you can use the Hyper Editor to control Logic's instruments and plug-ins, although the process is a little more complicated. The individual parameters of a plug-in can be accessed using what Logic calls Fader messages, but working out which one to use can be challenging. If you only want to program a few parameters, it's easiest to create some track-based automation and then use the Move Current Track Automation Data to Region function to create the basic MIDI Fader data. But if you want a more complete set up, you'll need to create Fader events in the Hyper Editor. To set up a lane to control Fader messages, choose 'Fader' from the status menu.

Once you've done this, you have to navigate a rather cryptic routing system to access plug-ins on a channel, whereby the Hyper Editor Channel parameter corresponds to the plug-in slot within the channel strip. The source object (the virtual instrument, in most cases) is found on channel 2, while the first Insert is called up by selecting slot 3, and so on. The Number parameter selects the specific control within the plug-in. Conveniently, the Info area of the Event List displays the name of the function, and it updates to correctly identify the controls if you change to a new plug-in, so it's an idea to have this open. Also, it's easier if you view the plug-in in Controls view, so you can see the parameter names.

Once you've created the Fader events in the Event List, it's relatively simple to make a Hyper Set for them. You could use one of the functions mentioned earlier to make a complete set, or check the Auto Define box, then click the desired Fader events in the Event List. Just remember to switch it off when you've done this, or you could end up with some spurious results! Within the Event Definition in the Hyper Editor, you'll then need to tick the Channel box, and set it to '2', as it's here that the input object resides. Finally, name each lane.

With your Hyper Set ready, you can now get creative, and start programming whatever parameters you like, with a rhythmic accuracy that would be very challenging to do any other way in Logic. If you want to program insert-effect plug-ins in the same way, redefine the event definition's channel parameter to the corresponding insert slot and start all over again.

As always, remember to keep all your Sets saved, so that you can easily revert back to them. SOS

Forthcoming Logic Events

Apple Solution Experts Digital Village are hosting a series of product demonstrations featuring the eight-core Mac Pro, Logic Studio, Euphonix Artist Series controllers and Apogee audio interfaces. The 21 sessions will take place in each of their eight locations within the UK. For more information and to register, visit www.dv247.com/logic.

- **DV Retail Warehouse, Romford: Saturday 5th April, 12pm, 2pm and 4pm.**
- **West London: Wednesday 9th April, 2pm, 4pm and 6:30pm.**
- **Bristol: Thursday 10th April, 4pm and 6pm.**
- **Cambridge: Saturday 12th April, 12pm, 2pm and 4pm.**
- **Clapham / Gateway School of Recording: Thursday 17th April, 2pm, 4pm and 7pm.**
- **North London: Saturday 19th April, 12pm, 2pm and 4pm.**
- **Birmingham: Monday 21st April, 4pm and 6:30pm.**
- **Southampton: Friday 25th April, 2pm.**

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The Joy Of Six

Digital Performer 6 New Feature Preview

As this pre-release screenshot shows, DP6's user interface, brighter and lighter than in previous versions, does away with non-standard or redundant features like the window title-bar buttons and the Control Panel's editing window buttons. Despite the changes, experienced users should feel completely at home.

The days are getting longer and Spring is definitely springing, but with so many exciting product announcements recently — including Digital Performer 6 — DP users might well feel like it's Christmas all over again.

Robin Bigwood

If you saw last month's NAMM round-up in *Sound On Sound*, or you follow any of the DP-based online discussion forums, you can't fail to have noticed the news about DP6, the latest version of Digital Performer. MOTU had given nothing away prior to the NAMM show, and certainly nothing to suggest the scale of the update that DP6 appears to represent.

Six Appeal

According to a January press release from MOTU, these are the main features of DP6:

- A heavily revised user interface, including vertically resizable tracks in the Tracks Overview window.
- Track comping that sounds as though it will trounce Logic 8's abilities in this area, including a new Take Tool, a multiple take view, and a flexible architecture whereby takes actually have the same status and facilities as full-blown tracks.
- Pre-rendering of virtual instrument tracks, with the promise that software instrument processing is done ahead of time, allowing any Mac to support many more notes of polyphony than it could achieve if the instruments were run in real time. This also means that instruments can be included in a Bounce To Disk with no need to record or freeze their outputs to an audio track.
- The option to work with Broadcast WAV or AIFF as a native audio format, rather than the increasingly antiquated Sound Designer II (though support for this continues, as you'd expect). Additionally, DP6 isn't fussed about whether multi-channel files are interleaved or not.
- CD burning (and disc image creation) built into the Bounce To Disk function, with Markers representing track IDs.
- ProVerb, a new convolution reverb that allows real-time tweaking and automation of parameters, drag-and-drop impulse response importing, and a dynamic mix feature that ducks wet level as the level of the audio input increases — intriguing!
- MasterWorks Leveler, a MAS emulation of a Teletronix LA-2A limiter — or, actually, four switchable LA2As of different vintages and specs.
- A plug-in management system, administered from Preferences, allowing plug-in sets to be created and the AU examination process to be re-run on demand. The possibility of running AUs that fail the examination process will also be offered.
- Improved Audio Unit performance, promising sample-accurate third-party virtual instrument timing, full audio side-chaining, and support for MIDI-aware Audio Units such as the Virus Control plug-in for the Access Virus TI synth.

- XML file interchange with Final Cut Pro, so that its edit lists can be viewed in DP6 and any subsequent changes highlighted in the time ruler. When DP is running concurrently with Final Cut Pro on a single Mac, entire projects can be exported into Final Cut Pro.

In a word, wow! DP6 appears to directly address the weaker areas of previous versions without trying to reinvent the wheel, as it were. The lack of a good bundled reverb and compressor plug-in had always been unfortunate, but the new ones seem to make up for that and then some. Decent track comping couldn't have come a moment too soon (especially since I'd tried and failed about three times to write a good Performer workshop article about best practice in DP5 and previous versions!). The enhanced audio file support and CD burning is something that everyone will benefit from, and it's great to see the Virus TI supported in DP at last, especially as similar hardware/software tie-ins seem to be all the rage these days. However, for me (and many other users, I suspect) the two real 'biggies' in DP6 are instrument pre-rendering and the user interface changes.

The Host With The Most?

Looking first at instrument hosting, it's true that in some circles DP4 and DP5 had acquired a reputation as being inefficient virtual instrument hosts — although in reality only Logic 7 could be demonstrated to be noticeably more efficient, and even then not by that much. While none of us have seen it in action yet, of course, DP6's instrument pre-rendering might just silence those DP detractors for ever. The idea is simple: DP6 does the maths associated with playing MIDI tracks from virtual instruments when it has processor cycles to spare, and during playback regurgitates the audio it has already worked out, rather than running the instrument in real time. The implications of this system, if it works well, are massive. You could load up heavyweight synths — the likes of Arturia's emulations and Rob Papen's modern classics — and run your Mac within an inch of its life while you actually jam and then record with them. But after the MIDI track is in place, processor hit during playback would fall to a level much more in keeping with the playback of a simple audio track. You could run dozens of synths, even on an older Mac, possibly making distributed processing or outsourcing of instrument hosting completely unnecessary. If this turns out to be the case, a lot of people will be very happy indeed.

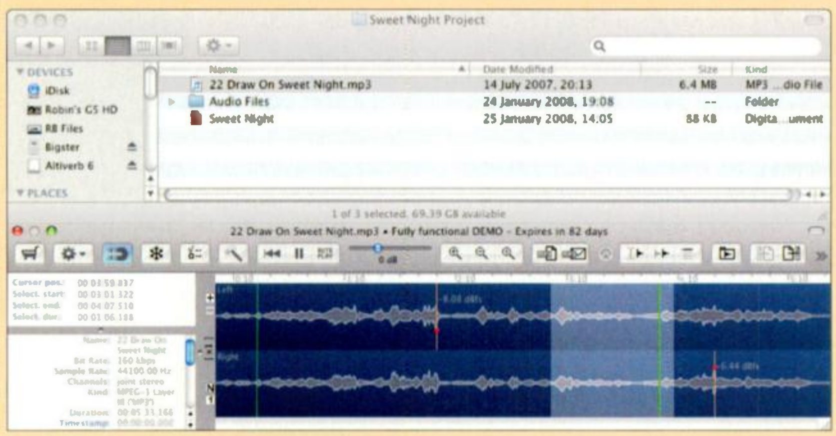
Interface Lift

Turning now to DP6's user interface, I feel MOTU were bang-on in deciding to do a redesign at this point in time. The interface quirks of previous versions, including their unusual title-bar buttons and mini-menus, were non-standard in the days of Mac OS 8, and to press on with them in the era of Leopard — and a climate in which the vast majority of applications are conforming to OS X's bright, clear and functional design — would surely have represented obstinacy bordering on commercial suicide. Consequently, DP6 looks to be a much brighter place to hang out, and has shaken off the quirky elements and the predominantly grey theme, while retaining the 'feel' and all the key features of previous versions. The zoomable Tracks Overview is just the tip of the iceberg, and there are many more important changes, at least judging by the pre-release version that was demo'd at the NAMM show. The Control Panel is completely redesigned and has lost the rather redundant pop-out panels, but has gained far better time-position displays, along with a permanently visible indication of recording sample-rate and bit-depth. The Mixing Board is noticeably more 21st century, and appears to show track input and output assignment much more prominently. Window title bars now contain nothing but OS X's standard 'traffic lights' — and at this stage we can only guess where the mini-menu and title-bar button functionality has gone instead. The Track Selector list has been 'unified', so there's now only one, which updates as you switch between different editing windows. Consolidated Window cells can now have tabs, allowing multiple windows to be active at any time and selected with a single click. There are brand new 'Inspector palettes', which can live in the Consolidated Window or float separately above other windows, displaying Event, Snap, Cursor and Selection information that continuously updates as you work. There's also live window resizing — a small but significant extra enhancement

Audio Ease Goodies

In recent weeks and months, Audio Ease have been busy posting updates to their DP-friendly plug-ins and bundles. Altiverb is updated to version 6.1.1 and offers a Leopard-compatible installer, amongst other fixes. The Rocket Science bundle goes to version 3.5.2 and gets similar improvements, while the Nautilus 2.5.2 bundle has a number of improvements to the great granular synthesis plug-in, Riverrun. You can now drag soundbites to its waveform view rather than having to 'sample' them live every time, and the waveform buffer is now saved along with a DP project.

Also, Audio Ease have finally released their 59 Euro audio file-browsing helper application, Snapper. It's an application that runs in the background and only pops up a window when you select an audio file in a Mac Finder window (or in iTunes). Then it displays the waveform for the file, along with an indication of peak-level values in each channel, and allows you to preview the audio with a simple tap of the space bar. However, some little (and very handy) editing facilities are also included: after dragging over part of the waveform you can export the selection in a variety of file formats, create a new Mail message with the audio attached as an MP3, or drag it to the desktop or another Finder window. As yet, some additional facilities afforded to Pro Tools users — pasting selections at the playback cursor or into the region bin — aren't implemented for DP or any other DAW, but I hope at some point that we might at least be able to drag audio selections from Snapper to audio tracks in DP directly. So Snapper certainly has a very streamlined feature set, but it's very nicely implemented and I've found myself using it a lot already. It will inevitably be compared with AudioFinder ($69.95 from www.icedaudio.com). Both applications integrate with OS X in a similar way, to preview audio files, but AudioFinder goes much further, functioning as a full-blown audio editor and sound-management application. (I should also point out that it does allow Finder audio selections to be dragged straight to DP.) Still, there's room for both in the market; AudioFinder wins in flexibility and power, whilst Snapper's strength lies in phenomenal ease of use and simplicity of purpose. As both are available as functional demos there's no excuse not to check them out in practice. Visit www.audioease.com and www.icedaudio.com and see how you get on.



that should help with various kinds of editing. Ultimately, DP6 looks 10 years younger than its predecessor, and should be a really nice environment to work in.

DP6 is due to be released in 'Q1, 2008', so as you read this it'll either have just come out or will be along shortly. MOTU are not bowing to the pressure of Logic pricing: DP6's list price is $795, which should end up being the same $500 or £350 that DP5 currently typically sells for. As I write this, upgrade prices have not been set. For all the latest info, check out www.motu.com. As soon as I get my hands on a copy, rest assured that we'll be covering all the new features in detail.

828 Goes To Mk3

For some reason, MOTU left it until after the NAMM show to announce this one, but the 828 Mk3 is on the way, at a list price of $795. It's a souped-up version of the existing 828 Mk2, with the key additions being 16-channel ADAT optical interfacing (both in and out), digitally controlled analogue gains with 73dB of gain available, hardware limiters on the front-panel mic/guitar inputs, and a couple of guitarist-friendly send/return loops.

There are less obvious enhancements too, that could really make a difference to some users. Two headphone outputs are fitted, and the main volume knob can be programmed to simultaneously control all the outputs involved in a 5.1 or 7.1 surround mix, as well as conventional stereo. This, of course, would negate the need for an expensive outboard surround volume control.

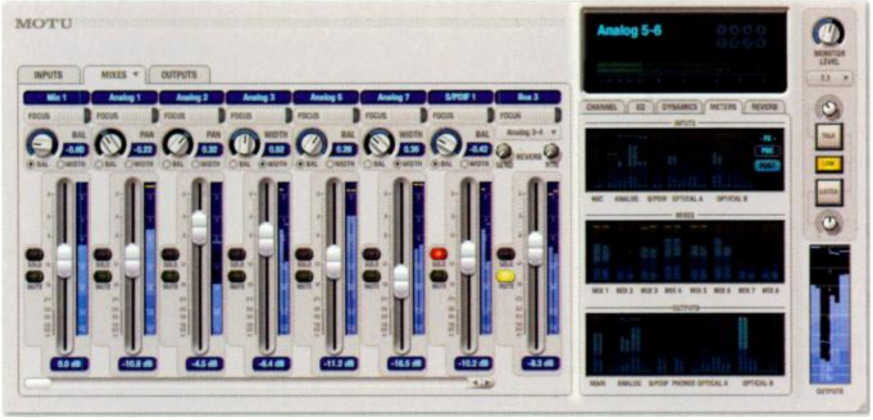
The biggest change of all — at least, compared to every previous interface MOTU have made — is under the hood, in the form of an improved CueMix near-zero-latency mixing architecture. The new system is called CueMix FX and — you've guessed it — includes DSP-driven hardware effects. While mixes can be set up and adjusted from the front-panel knobs, the Windows or Mac CueMix FX application makes working with them a much easier and more visually-oriented proposition. Three effects are on offer: Classic Reverb, a seven-band parametric EQ, and Vintage Compression. Apparently there's enough DSP 'oomph' to support at least one band of EQ and compression on every single input and output (58 channels) simultaneously, but the likelihood, of course, is that you'd apply DSP to far fewer channels at a time, and thereby be able to use more bands of EQ. Compared to all that EQ and compression there's just the one reverb, but it can be accessed with sends from every input, output or mix bus.

Since these effects are strictly bound to the interface itself — you can't run them as plug-ins inside DP, for example — how much use are they? Well, the reverb is seriously useful for providing musicians, especially singers, with reverb on their headphone monitor signal as they record.

The EQ and compression are much less useful for monitoring purposes, but since you can record with them, 'wet', into your DAW, they'll allow you to treat your 828 Mk3 much more like a dedicated outboard channel-strip. This could suit some people's way of working very well, allowing them to zero in on the sound they're after during the actual recording stage, rather than later, at the mixing stage.

The effects themselves look quite capable. The reverb has five different room types and plenty of other bells and whistles, while the EQ is, predictably, modelled after the 'British EQ' sound, and seems to have facilities extremely similar to DP's MasterWorks EQ. The dynamics section actually consists of a conventional single-band compressor switchable between Peak and RMS operation, plus a leveller/limiter based on the LA2A.

Owners of existing MOTU CueMix-enabled interfaces will inevitably be clamouring for the attractive CueMix FX console to support their audio hardware. The word from MOTU is that this will indeed eventually happen, but possibly some time after the 828 Mk3 is released. Also — and I'd be happy to be proved wrong on this — I imagine it's almost certain that existing interfaces will not be able to support DSP effects, and will instead just benefit from being driven by the improved CueMix FX environment, while also possibly acquiring a few new facilities such as bus master faders. We'll have to wait and see... SOS



The new CueMix FX application designed for the MOTU 828 Mk3 interface is significantly nicer-looking and more configurable than the current CueMix Console. Owners of older CueMix-equipped interfaces should eventually be able to use it with their hardware.

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DON WAS

Bob Dylan, The Rolling Stones, Willie Nelson, Bonnie Raitt

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by clicking Freeze in the RTAS plug-in and Process in the matching Audiosuite plug-in.

This plug-in looks like a real life-saver for capturing that 'once in a lifetime solo' that you didn't record on the run-through because it was only a run-through. You insert the plug-in on a channel you want to 'protect', and it records into the RAM buffer all the time, whether Pro Tools is playing, recording or stationary. Synaptricity also suggest using Flashback as a Master Fader insert so that it captures everything going through the mixer, and outline a number of possible scenarios for this. You can set the buffer size from one minute up to 15 minutes; if you're using a number of instances of this plug-in, you need to make sure you have enough memory, and the Flashback manual has some tables to help you work out how much you can use.

In HD systems, Flashback needs to be paired with a TDM plug-in to work properly.

Once installed, I found Flashback in the Delay group of plug-ins, which is a strange category to put it under. To start with I inserted it on a vocal channel to capture anything the singer might get up to. We did a run-through, and at the end I hit the Freeze button on the plug-in window, followed by the 'RTAS — Audiosuite' button. Then I opened the corresponding Audiosuite plug-in, placed the cursor on the track, hit the Process button and got a blank Region! It dawned on me, of course, that Flashback is an RTAS plug-in, and on HD systems RTAS plug-ins are put into bypass when you go into Record or Input Monitor, as I was. As the manual explains, the workaround is to insert a TDM plug-in before the Flashback plug-in.

So, with my TDM plug-in in place, we had another go, and this time it worked as expected. However, as Synaptricity make clear, there is no time-stamp on the RAM files, so there is no easy way to sync up the files created by Flashback: you will have to line them up by eye and ear afterwards.

I had high hopes for this plug-in when I first heard about it, but most of them have been dashed (although, to be fair to Synaptricity, they may have been a little unrealistic!). Not being able to sync up the Flashback files means a lot of trial and error to get them in the right place. In addition, covering a multitrack Session on a track-by-track basis would be very difficult, as you can't hit a master Freeze button for all active instances, and using it in the Master Fader limits you to the mix you have going through the Pro Tools mixer.

The audio that's captured by the Flashback RTAS plug-in can be inserted onto a track using the Flashback Audiosuite plug-in, but is not time-stamped, so you'll have to move it 'by ear'.

To cover that 'killer solo' moment on the first run-through, surely the easiest thing to do is to put Pro Tools into Record anyway, or, failing that, use Quick Punch (see the workshop on Quick Punch and Track Punch in the December 2006 issue, on-line at www.soundonsound.com/sos/dec06/articles/ptworkshop_1206.htm) to capture that apparently unrecorded solo. If I were trying to capture an improvisation over a loop, which is one of the scenarios outlined in the Flashback manual, I would use Loop Record and just keep going till I got 'the one', which would be time-stamped and ready to use in the Session.

The unique feature of Flashback is its ability to capture audio when the Pro Tools transport is not playing or recording, and I am sure that will save someone's life more than once, but as a safety tool during normal session work, I am not convinced of its usefulness.

£ **$199 from Synaptricity web site or £165.68 including VAT from Digistore.**

W **www.synaptricity.com/productsflashback.html**

Slacker

Steven Massey's Slacker is an experimental tool for performing automatic backups of your Session's audio files (if you want to know more about Steven, take a look at the Pro Tools workshop in the September 2006 issue: www.soundonsound.com/sos/sep06/articles/ptworkshop_0906.htm). Slacker comes as part of his free Massey Tools set of plug-ins, which also includes a high-resolution audio level meter; this, too, is in the early stages of development, so that to reconfigure the meter you have to edit the Preferences file. The third plug-in, Listen, is, as Steve describes it, "the opposite of a meter plug-in". I will let you try it out yourself, so you can benefit at first hand from the improvements to your mix that this plug-in will bring.

Slacker comes in the form of a mono RTAS plug-in that can be inserted anywhere in a Session. It does not affect audio on that

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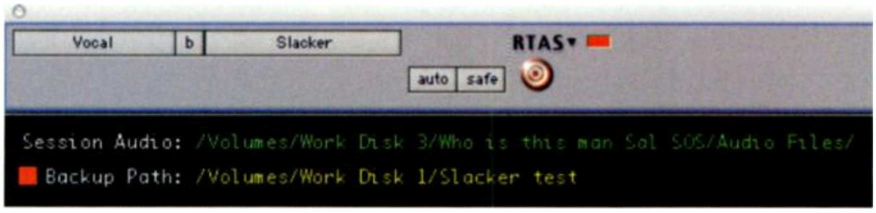

channel, acting like a permanently bypassed plug-in, and you only need one instance of it in a Session.

Whenever the Session transport stops, the plug-in will copy any newly created audio files in the Session audio folder to a specified backup destination. The square indicator next to 'Backup Path' in the UI will turn red to indicate disk activity. On an HD system, the plug-in clip indicator will also turn on during disk activity. If you're working with a very dense Session, Steve suggests that "you may wish to wait until the disk activity completes before restarting the transport".

Once the plug-in is inserted on a mono track somewhere on your Session, you need to click the area that says 'Click here to select backup destination folder' to configure a suitable backup location. Obviously, to make maximum use of this plug-in you should choose a folder on a different hard drive from your Session folder. Now, whenever you finish recording some audio, Slacker kicks in and copies any new audio files it finds in your Session's Audio Files folder to the backup location.

I tried it on a Pro Tools Session we were adding some vocal overdubs to, and sure enough, as soon I had stopped recording, the red lights showed in the plug-in window. I was surprised at how long they stayed on for, given that we had only added a one-minute mono file to the Session, so I took a look at the backup folder, to find that Slacker was busily backing up all the audio files in the Session, and not only the one we had just added! I also found the plug-in reacted so quickly that it didn't pick up that I had renamed a file immediately after recording. However, the next time I stopped recording, Slacker copied the renamed file across, leaving the original file intact in the backup location as well.

Overall, I found this little freebie, even in its v0.2.1 state, much more useful in protecting me whilst working than Synaptricity's Flashback. It would be great if Slacker would copy the Session file too — maybe Steve could get it to look at the Session Backup folder and back up the latest Session file every so often as well? In conjunction with the audio file backups, it would then be relatively easy to reconstruct a Session if the main drive failed.

I would recommend you try the Massey Tools bundle for yourself, and I know Steve would love to hear from you with your experiences of them. I certainly plan to run Slacker on Sessions where I am tracking. In the July 2006 Pro Tools workshop on backups and archiving I described my previous approach, which used Synchronise! X Plus to backup my Session folder; this had to be run manually during tea breaks and so on, which has meant that until now my backups could be up to an hour or so out of date. With Slacker, I know that within seconds of hitting the Stop button all the audio will be backed up. SOS

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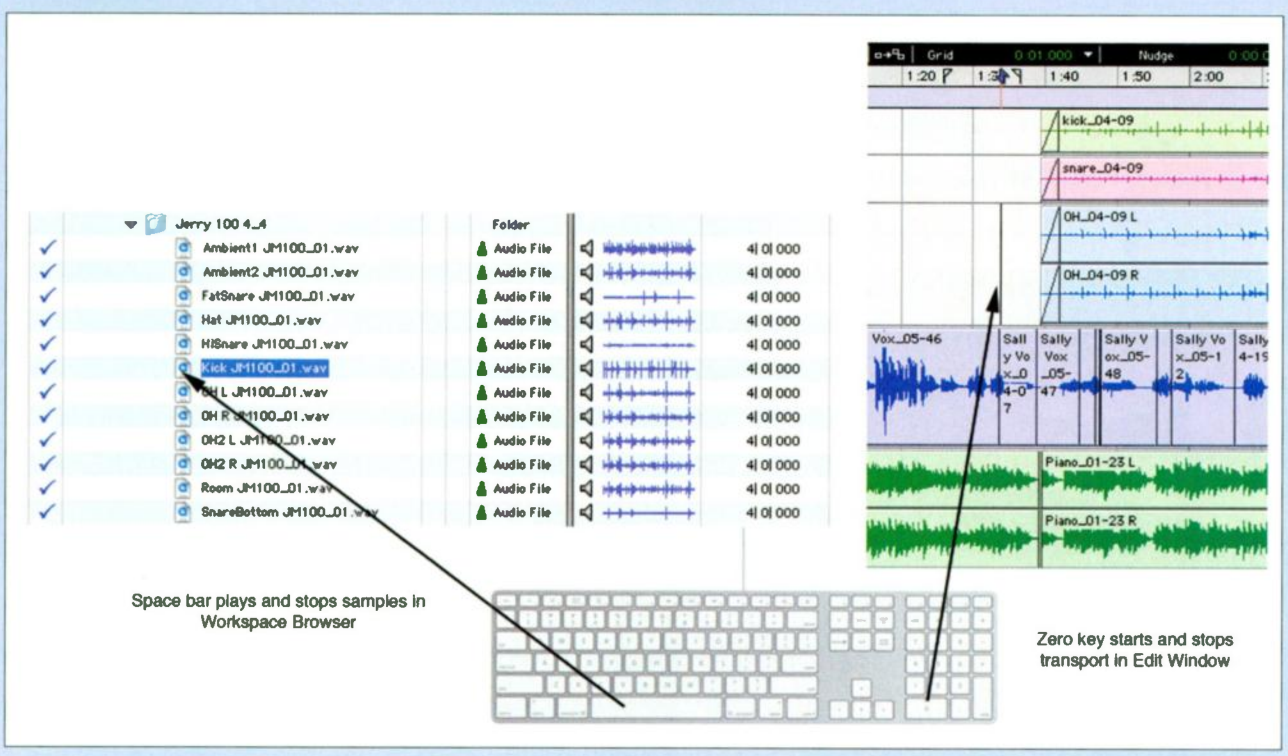
All Workspace And No Playback?

One of the new features in Pro Tools 7.4 is the option to start and stop playback of your Session while working in the Workspace Browser. There are two different keyboard shortcuts that you can use to start and stop Session playback. The first is the space bar and the other is the zero (0) key on the numeric keypad, providing you have Transport selected in the numeric keypad preferences on the Operations tab of the Pro Tools Preferences window.

When the Workspace Browser is the active window, these two shortcuts have different functions. The space bar starts and stops playback of selected samples in the Workspace Browser, while the zero key will start and stop playback of the main Session in the Edit Window. This arrangement makes it easy to preview loops and samples alongside your entire Session, without ever needing to leave the Workspace Browser.

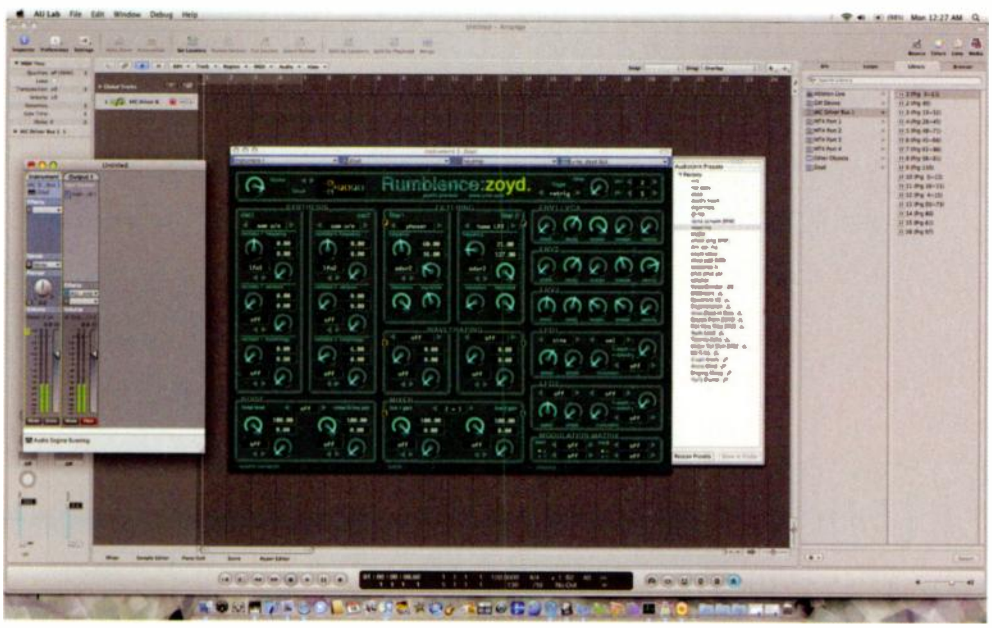


Have you ever wondered why Power PC plug-ins can't run in Intel applications, or why your 32-bit plug-ins won't work in 64-bit applications of the future? Apple Notes explains all.

Mark Wherry

While the Mac enjoys a well-deserved reputation for its ease of use, and a place in the hearts of a large number of musicians and audio engineers, Mac users have had to put up with a fair number of complicated architectural transitions over the past few years. It started with the introduction of Mac OS X in 2001, continued with the change to Intel processors in 2006, and will now happen again as we migrate to applications that can take advantage of 64-bit memory addressing.

Each of these architectural shifts has meant that new versions of the applications people rely on every day have to be created. And because so many musicians and audio engineers became more reliant on plug-in software effects and instruments in the '90s (was it really so long ago?), each shift has also meant an even bigger headache in getting all of your favourite plug-ins to work again.



Core MIDI can be used to send events from an Intel-based application (such as Logic Pro) to a Power PC application (here, AU Lab, running via Rosetta) over an IAC bus.

I don't know what the exact ratio of applications to plug-ins is, but it probably wouldn't be far-fetched to say that for every major application that needs to be ported there are probably several hundred plug-ins as well. And because there are so many, and so much diversity in the ecosystem that develops them — from students to seasoned pros — it's inevitable that some plug-ins never get ported to a new architecture, which is frustrating if you rely on such tools but you still need the latest Mac hardware or software.

Part Of The Process

In order to understand why it becomes impossible to run certain plug-ins with certain application software and hardware combinations, we need to take a brief look under the bonnet.

When you run an application like Logic Pro, Mac OS X creates what's known as a Process, which, simply put, is responsible for running the application in the OS. This is somewhat analogous to a recipe requiring a chef to cook it, where an application's code can be thought of as the recipe and the chef is the Process. If one chef gives his undivided attention to a single recipe, you'd need another chef to cook a second recipe, and similarly each application you run in OS X requires a new, independent Process. You can see how many Processes Mac OS X is running in the Activity Monitor, found under Utilities in the Applications folder.

However, when you use an Audio Unit in Logic Pro, instead of a new Process being created to run that plug-in, the Process that's running Logic Pro also becomes responsible for the plug-in, which is why you won't see plug-ins listed in the Activity Monitor. (Although my example refers to Logic Pro and Audio Units, the principle is the same for any host that runs a plug-in.)

Once you understand that an application and any plug-ins running within that application are part of the same Process, it's easier to understand issues like why you can't run Power PC plug-ins in an Intel application. Because a Process can only run the code for a single architecture, the Process has to be either running Power PC code (translated via the Rosetta translation technology on an Intel Mac) or Intel code. It also means that a Process has to run either 32-bit code or 64-bit code. So a single Process can't run Power PC and Intel code simultaneously, nor can it run 32- and 64-bit code simultaneously.

This ultimately means that because an application and its plug-ins run under the umbrella of a single Process, everything needs to support a single architecture, whether it's Power PC, Intel, 32-bit or 64-bit. And this is why Power PC plug-ins needed to be recompiled for the Intel (or Universal, supporting both Power PC and Intel) platform to be used in Intel-native hosts, and why 32-bit plug-ins will need to be recompiled as 64-bit plug-ins, to run in a 64-bit host application.

As an aside, the fact that an application and any plug-ins that are running in that application share a single Process also explains why 32-bit applications run into memory issues even if your Mac has a large amount of memory. Basically, a 32-bit Process in Mac OS X (running a 32-bit application) can only access 3GB of memory, and again, because the plug-ins running in an application share the same Process, they can only make use of that one 3GB pool of memory.

A 64-bit Process (created when a 64-bit application is run), on the other hand, can access more than 3GB of memory, which is why a 64-bit application running with 64-bit plug-ins will be able to make use of all the memory you have available, even though there will still only be a single Process running everything.

Inter-process Communication

Now we've discussed some of the technical details, you might be wondering why you couldn't just run two Processes at the same time, to eliminate compatibility issues. Why not have one Process running Intel code and another running Power PC code, and have the two communicate with each other? Well, not only is this possible, it's how two current

Ready For Review, It's Ten-Five-Two

It's not often that you can get excited about a minor 'point-one' upgrade to Mac OS X, but in the case of Mac OS 10.5.2 I'm prepared to make an exception. While such minor versions usually bring fixes, under-the-bonnet improvements, and the occasional incompatibility, 10.5.2 adds a couple of 'new' features to the user interface that address much of the criticism Leopard has received from existing Mac users.

When we discussed Leopard in January's Apple Notes (www.soundonsound.com/sos/jan08/articles/applenotes_0108.htm), two aspects of its user interface that didn't seem like a step forward were the translucent menu bar and the Stacks feature in the Dock. In the case of the menu bar, the new translucency meant you could see part of your desktop background underneath, enabling the menu bar to blend more seamlessly with the Desktop. In practice, though, this often made parts of the menu bar illegible, and looked ridiculous when running an application with a single window that takes up the whole screen, like Logic Pro or Pro Tools.

In 10.5.2 Apple have introduced a new option to let you disable the 'Translucent Menu Bar' in the Desktop part of the Desktop and Screen Saver System Preference, which is a big relief. And the menus themselves now also appear slightly less translucent than before, making them easier to read when there's lots of clutter on your screen.

Like many Mac users, I used to rely heavily on Tiger's ability to let you right-click on folders in the Dock to navigate the contents of those folders via pop-up menus, and, like many Mac users, was dismayed to find this behaviour replaced in Leopard with the useless Stacks feature. However, in 10.5.2 Apple have added a new List view to Stacks, which basically works just like the old Tiger behaviour, except that you now have to use a primary (left) click to open the pop-up menu, rather than a secondary (right) click, which I think I can live with!

Right-clicking a Stack, as in earlier versions of Leopard, lets you configure the Stack's behaviour, such as making it display as a List instead of a Fan or Grid, and 10.5.2 also lets you choose whether the Stack should show the icon for the folder placed in the Dock instead of the icon for one of the files within the folder. So finally, as with Tiger, my Applications folder now looks like my Applications folder again when placed in the Dock. Phew!

In other Leopard-related news, Digidesign have released a new 7.4.1 version of Pro Tools HD that's compatible with the latest Mac Pros featuring 5400-series ('Harpertown') Xeon processors running Mac OS 10.5.1; 10.5.2 is, alas, not supported at the time of writing. Still, this minor upgrade is important for Pro Tools users who purchased a new Mac Pro but couldn't officially run Pro Tools, since there was no way to downgrade the version of Mac OS X that came with the system to one that was approved by Digidesign.

Pro Tools HD 7.4.1 is not for users with earlier Mac Pros, however, so, like Pro Tools LE users, you'll need to wait for another phase in Digidesign's Leopard compatibility plan before you take the plunge with the latest version of Mac OS X.

products deal with the problem of running incompatible plug-ins on a modern system.

The first such solution is VST Bridge, supplied with the latest 4.1 versions of Cubase and Nuendo (and discussed in the Nuendo 4 review, *SOS* December 2007), which allows you to use Power PC plug-ins inside the Universal version of Nuendo running on Intel Macs. Presumably, it will also do the same for 32-bit plug-ins in the 64-bit versions of Cubase and Nuendo (as it already does on Windows), when they're available.

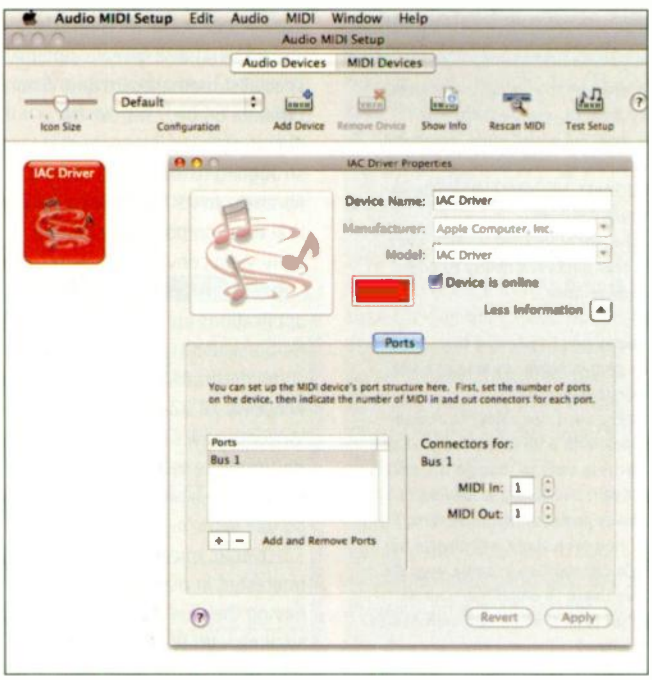
The second solution is the Intel Wrapper recently introduced by Spectrasonics to allow the Power PC-only Atmosphere and Trilogy instruments to run on Intel Macs. This isn't to be confused with the type of wrapper that would allow you to run VST plug-ins in an RTAS host (such as FXpansion's VST to RTAS Adapter), since these are typically plug-ins that host other plug-ins, meaning that the RTAS host, the wrapper plug-in and the VST plug-in still all run within a single Process. With Spectrasonic's Intel Wrapper, you run a special Intel-compatible plug-in in your host (under the Intel Process), and that plug-in communicates with a separate Power PC Process running the actual instance of Atmosphere or Trilogy.

If you don't use Cubase or Nuendo and have some old Power PC instrument plug-ins you'd like to still use, there is a workaround. If you have a Power PC host, you can still run this on your Intel Mac to access Power PC plug-ins, and if you don't have such a host you can instruct a Universal application to run as Power PC code under Rosetta, by enabling the 'Open using Rosetta' option in the Info window for that application, which you can access by selecting Get Info on the application's icon.

Because Rosetta applications also have access to Core MIDI and Core Audio, you can run the Power PC host alongside your main music or audio application and send MIDI to it via Core MIDI's built-in IAC (Inter Application Communication) Driver. This is disabled by default, but you can enable it by double-clicking the IAC Driver in the MIDI Devices page of Audio MIDI Setup and enabling the 'Device is Online' tick-box in the Properties window. Initially you will have one MIDI bus available, providing both an input and an output, but you can create as many buses as required by clicking the Add Port button.

In your sequencer you can select the IAC bus as the output to a MIDI track, and in the Power PC host you can select the same IAC bus as an input to the Audio Unit instrument plug-in you want to play. For audio, you could use one of the many plug-ins that send and receive audio over a network to communicate between the two applications on the same Mac, such as Apple's AUNetSend and AUNetReceive pair, but in practice I found the latency less than ideal, compared with routing the Power PC host directly to a Core Audio device.

You might notice that since Leopard now supports 64-bit applications, with user interfaces for applications that include both 32- and 64-bit code, there's a new option in the Info window for applications: Open in 32-bit Mode. This forces an application to run as a 32-bit Process and will allow similar compatibility workarounds to those we've described for running a Power PC host. However, unlike Power PC Processes, which take a performance hit because they have to run translated Power PC code via Rosetta, there's no such penalty for running 32-bit code in a 32-bit Process alongside a 64-bit Process. SOS



Using Audio MIDI Setup, you can enable the IAC Driver for sending MIDI data between two different Processes.

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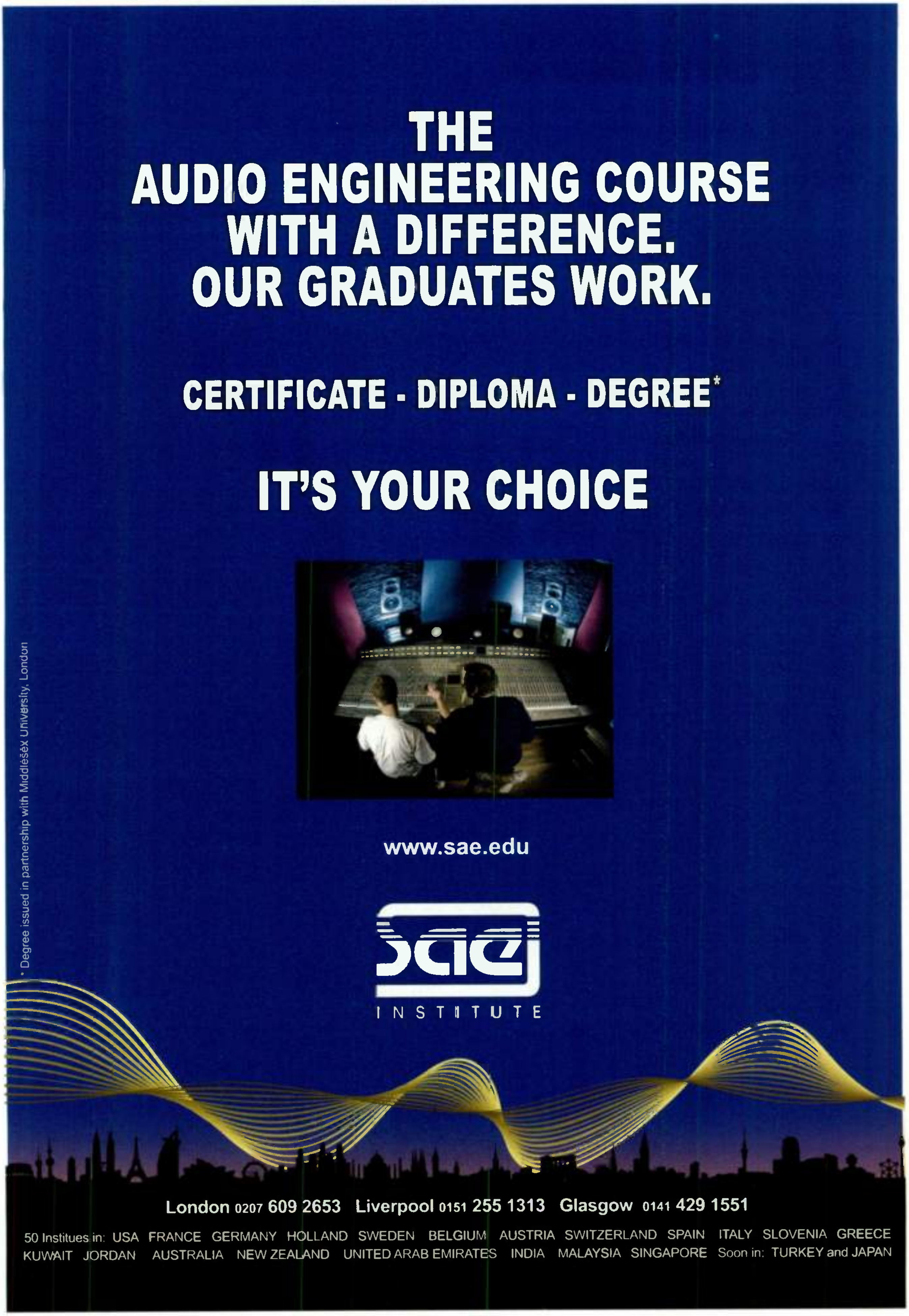
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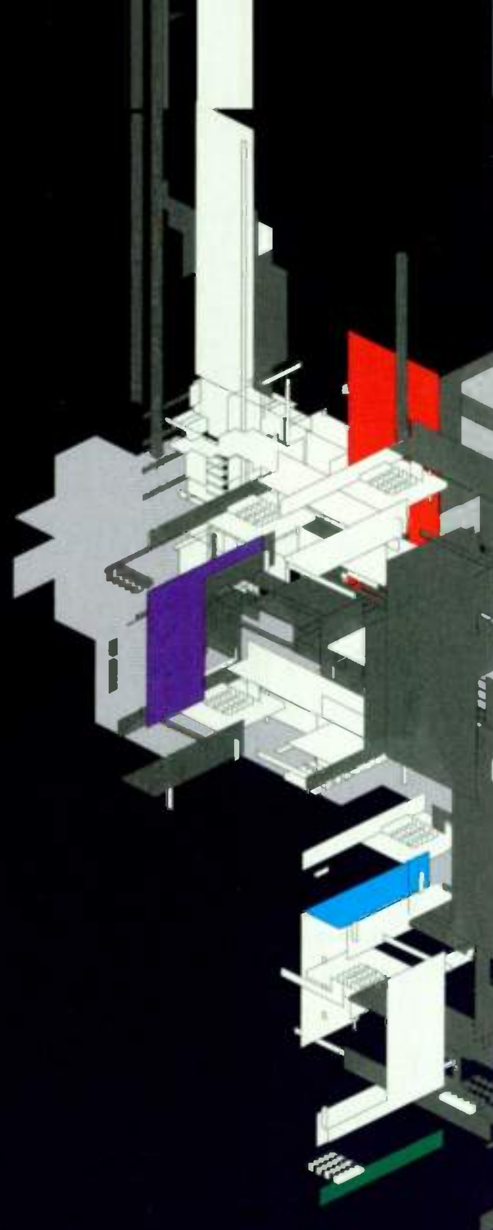

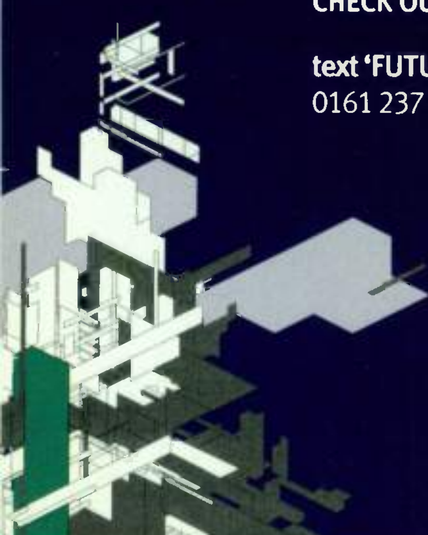

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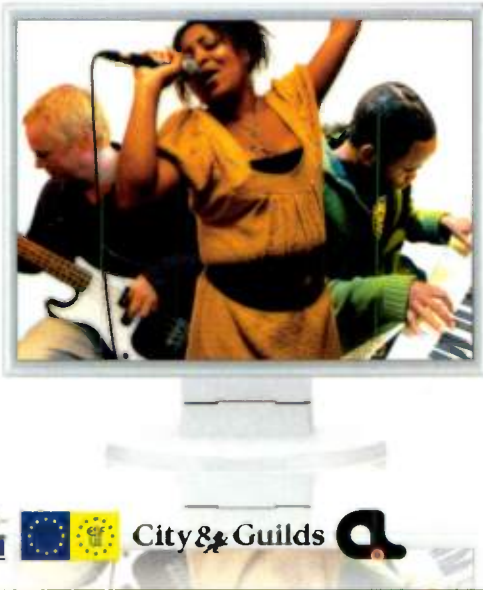

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ASAP Europe	020 7231 9661	129
Audient PLC	01256 381944	113
Audiofanzine.com	0033 143 223 926	47
Avid Technology Europe (M-Audio)	0871 717 7100	25, 177
ClearAngel	www.clearangle.net	65
Digital Village	01708 771900	31, 89
Directresolution Computers	01209 831499	61
Edirol Europe	020 8747 5949	59
Euphonix	001 818 766 1666	49
Focusrite Audio Engineering	01494 462246	15, 191
Freestyle Music Products	01924 455414	145
Golden Age Music	www.goldenagemusic.se	109
Guitar, Amp & Keyboard Centre	01273 671971	97
Hand In Hand Distribution	01579 326155	135
Inta Audio Computer Systems	0870 199 9397	77, 133
Joemeek	01803 215111	127
KMR Audio	0208 445 2446	165
Korg UK	01908 857100	OBC
Line 6 UK	01327 302700	101
M Corporation	0845 025 5555	55
Mackie UK	01268 571212	9
MB Media	01353 665577	187
Musicians Benevolent Fund	020 7636 4481	181
Musictrack	01767 313447	IFC
Musikhaus Thomann	0049 954 692 2355	50-51, 52-53, 178-179
Numark Alesis UK	01252 341400	35
Red Submarine	www.gear4music.com	153
Roland UK	01792 702701	41
SAE Institute	020 7609 2653	66-67
Sensoria	www.sensoria.org.uk	199
Scan Computers International	01204 474747	37
Sommer Cable	www.sommercable.com	33
Sonic Distribution	01582 470260	43, 71, 83, 195
Sonic8	08701 657456	159
Soniccouture	020 8948 6614	29
Sound Network	020 7665 6463	121, 197
Sound Recording Technology Show (SRT 08)	www.londoninternationalmusicshow.com	166-167
Sound Technology plc	01462 480000	IBC, 3, 105, 125
Source Distribution	020 8962 5080	44-45, 86-87, 138-139, 151, 169
Stirling SCV	020 8418 0778	75, 111
Studiospares	0845 644 1020	122-123
Synthax Audio	0049 813 391 810	72-73
Tascam UK	01923 819630	119
The Music Corporation	0845 025 5555	55
Time + Space Distribution	01837 55200	92-93, 131
Toft Audio Designs	01803 215111	161
TSC	020 8400 1234	1
Turnkey	020 7419 9999	Loose Insertion 18-19, 21, 23, 223
Unity Audio	01440 785843	63, 98-99, 155, 163
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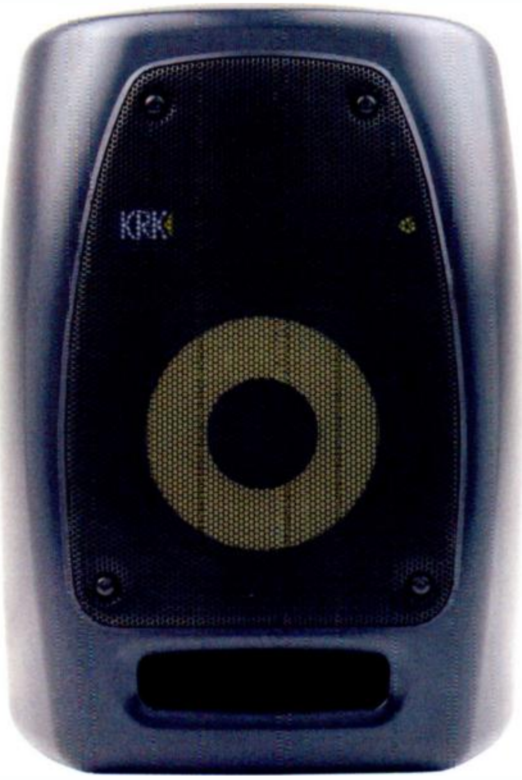

Over 20 years ago, frustrated by the fact that he couldn't find an accurate and clear studio monitor, a sound engineer called Keith Klawitter began designing and making his own speaker systems. Soon after, his peers started to take notice, and Keith rapidly turned his quest for clarity into a thriving business: KRK Systems.

Today, KRK monitors are still highly regarded and used by many top producers. Instantly recognisable by their yellow cones, they are found in the mix rooms, control rooms and dubbing suites of studios across the globe.

Just over a year ago, at the 2007 Frankfurt Musikmesse, KRK released their VXT range of active studio monitors. There are three models: the VXT4 (reviewed in *SOS* August 2007 and on-line at www.soundonsound.com/sos/aug07/articles/krk_vxt4.htm); the VXT6; and the largest in the range, the VXT8.

This month, thanks to KRK's UK distributors Focusrite, we've got two pairs of VXT6 monitors (which normally cost £678 per pair) to give away. The VXT6 is a ported, two-way active monitor with a six-inch woven Kevlar woofer and a one-inch silk-dome tweeter. At the heart of its bi-amped design are a 60-watt amplifier that handles the low frequencies and a 30-watt amplifier that serves the tweeter. A wide, front bass-reflex port allows even LF extension down to 49Hz, and a grille, which is a newly announced accessory, protects against cone damage. Back-panel features include a combi input that accepts XLR and jack connectors, and a tamper-proof potentiometer for adjusting the system level. There are also switches for engaging a peak limiter, a ground-lift circuit and an automatic muting function. The last cuts the power amps if no input signal is detected for 20 minutes. A further two switches allow the user to shape the frequency response of the monitors to best suit their environment.

If you would like to be in with a chance of winning a pair of these great monitors, simply fill out the entry form at the bottom of this page and post it to the address on the coupon, or enter electronically via the *SOS* web site. Please make sure you answer all the questions and complete the tie-breaker. We also require your full address, including postcode, and your daytime telephone number, so we can contact you if you win. The closing date for entries is **April 30th, 2008**.

Thanks to Focusrite for supplying these prizes.

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Sonnox Competition Winner

Mark West, from East Yorkshire, won a Sonnox Six Pack of plug-ins in our September 2007 competition. Mark comments: "The suite is helping to give better focus to both overall mixes and individual instruments. Sounds retain their individuality even with a large amount of processing." Thanks to Source Distribution for the prizes.

questions

What was the name of KRK Systems' founder?

- a. Kenny Ken ☐
- b. Keith Klawitter ☐
- c. Krusty the Klown ☐
- d. Kevin Keegan ☐

What material is the cone of the VXT6's woofer made from?

- a. Buckminsterfullerene ☐
- b. Aerogel ☐
- c. Woven Kevlar ☐
- d. Sulphur Hexafluoride ☐

After how long do the power amps in a VXT6 power down when no input signal is present?

- a. 12 seconds ☐
- b. 10 minutes ☐
- c. 20 minutes ☐
- d. Six hours ☐

KRK tie-breaker

One of KRK's ranges was called Rokit. If you were to name products after a self-propelling transportation device, what would it be and why? Answers in 30 words of fewer, please.

....

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Name ....

Address ....

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Daytime tel. no: ....

Email: ....

If you would like to receive more information about KRK Systems' products, please tick or cross this box. ☐

Post your completed entry to: KRK Competition April 2008, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge CB23 8SQ, England.

the small print

1. Only one entry per person is permitted. 2. Employees of SOS Publications Ltd, Focusrite, KRK, and their immediate families are ineligible for entry. 3. No cash alternative is available in lieu of the stated prize. 4. The competition organisers reserve the right to change the specification of the prize offered. 5. The judges' decision is final and legally binding, and no correspondence will be entered into. 6. No other correspondence is to be included with competition entries. 7. Please ensure that you give your DAYTIME telephone number on your entry form. 8. Prize winners must be prepared to make themselves available in the event that the competition organisers wish to make a personal presentation.

Is landing a studio job a numbers game?

Joe Deller

With literally hundreds of music technology courses on offer, from a wide range of establishments that are often equipped with contemporary and expensive equipment, there are evidently a great many people wanting to work in the music industry. However, take a look at any recent copy of *Sound On Sound* and compare the number of ads offering courses to the number of situations vacant. There is a serious mismatch. Current estimates vary between hundreds and literally thousands of applicants for every situation that is vacant.

Even my small project studio receives 10-20 emails or calls each week from people wanting either work experience or paid work. My web site states that I don't have positions, yet still the emails and calls come.

If I were to advertise a position, I would expect to get hundreds of applications, from a range of candidates, not just music tech graduates. If I advertised an experience-only position, there would still be plenty of applicants, and having a music tech qualification would be no guarantee you would be any nearer the top of my pile.

Historically, the recording industry has been one where you very much start at the bottom, as a tape-op or even a tea-maker, and put in a lot of hard work, often with poor or no pay. Things are getting harder, as technology is removing some of those traditional routes into the industry — no more tape-ops — yet many of the people contacting me think that a qualification is going to be a shortcut, and that they are going to land a producer-style role at the outset. They might just get lucky, but the probabilities are against them.

Many studios are changing the nature of the work they do, or are closing. The digital revolution means that many artists no longer need the facilities offered by a traditional studio. However, the property situation today means that rents are higher than ever, and studios often do not give the best 'yield' per square foot. A local, well-established studio recently closed, as the land it sat on was worth hundreds of thousands of pounds to the landlord compared to the yearly rental; more than ever, budget is everything, both for the clients and the studio.

For a company to employ you, you need roughly to be worth at least twice your salary. In the UK, a company has to cover National Insurance contributions, holiday pay and sick days, provide you with appropriate tools and space, and foot the bill for other costs. If you want a salary of £20K, your value needs to be double that. Can you bring £40K worth of client bookings to the business? Perhaps not, but if you can show that employing you will save thousands in services currently sourced externally, that will certainly help.

It's not just your technical know-how that will get you a job. When working with groups of musicians, you need a basic level of competence, but even if you are the world's finest Pro Tools expert, if you don't work well with people — especially tired and tense artists — you are likely to find getting work, and staying employed, quite a challenge. On the other hand, if you can make people feel comfortable around you and can tease out the final take that makes the track work, you're much more likely to be asked back. You have to know when to be the benevolent dictator and when to be the quiet, efficient engineer. Studio work often means long and unsociable hours, dealing with difficult situations with diplomacy, often for not much financial reward or recognition.

Despite the challenges, there are jobs to be found and, as ever, perseverance and research are the keys to landing one. You may have to do a fair amount of low-paid or unpaid work, and finding the balance between under- and over-valuing yourself is something that takes time. You might get a lucky break, but chances are that you're going to be in for a long slog before you get left alone to produce or mix. Flip through books such as Howard Massey's *Behind The Glass*, and you'll notice that many of the top producers and engineers worked their way up the ranks, and all share a passion for music made by others, not just their own creative efforts.

So study hard, but get out and see a range of gigs, especially gigs by bands you can't stand. Figure out why their audience is enthusiastic, and be enthusiastic about that enthusiasm. One day your foot in the door is going to be relying on it. SOS

About The Author

Joe Deller is a self-confessed '70s glam-rock veteran and software escapee who manages a small project studio in Oxford (www.rockrooms.co.uk).

If you would like to air your views in this column, please send your submissions to soundingoff@soundonsound.com or to the postal address listed in the front of the magazine.


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