Everything For The Recording Musician ■ Hardware ■ Software ■ Reviews ■ Techniques ■ Interviews ■ News

SOUND ON SOUND™

The UK's Biggest Selling Music Recording Magazine

August 2006 £4.99

win! Focusrite Gear worth £1465

Duende

32 channels of SSL mix power in your DAW!

Solid State Logic
SOUND || VISION

Emulator X2
Have Emu created the ultimate soft sampler?

Noise-free Guitars
Kill hum and buzz with our step-by-step guide

Icon D-Control & D-Command
In-depth review of Digidesign's revolutionary control surfaces

Creamware B4000
The Hammond sound without the hernia!

Donald Fagen
"You don't need any of that fancy stuff."

Dollars from downloads
How to make on-line distribution work for you

SOUND ON SOUND

www.soundonsound.com

08

9 770951 681092

Take your music to the ends of the earth

in just a few clicks...

- Instant access to 8GB of world instrument sounds, loops and phrases together in one window.
- Browse sounds by category or by region of the world.
- Create layered stacks in seconds.
- Tempo-sync features let you browse & play loops in tempo with your music.
- Sliced loops let you rearrange beats or play them via your MIDI controller.
- Drag phrases and loops directly into your host software, tempo-locked to your project time line.
- Convolution reverb provides authentic acoustic spaces, from natural settings like caves and forests to spacious concert halls and soaring cathedrals.
- Advanced programming for live performance, including key switching, velocity crossfades and zone splits.
- Play up to 64 parts simultaneously via 64 MIDI channels.
- Unlimited polyphony and ultra-low software latency.
- Submix to 17 stereo output pairs.
- Disk streaming conserves RAM resources and quickens load times.
- Supports all major Mac/PC plug-in formats. Stand-alone version included.

MOTU
motu.com

Ethno
INSTRUMENT

Africa: Balafon, Kora, Ngoni Donso, Sanza, Valiha... **Asia:** Koto, Liuquin, Shakuhachi, Shamisen... **Australia:** Aboriginal Percussion, Didgeridoo... **Celtic:** Bagpipes, Flutes, Guitars, Harp, Harmonica... **Eastern Europe:** Balalaika, Balkanish Instruments... **India:** Electric Sitar, Harmonium, Tambura, Sitar... **Middle East-Mediterranean:** Baglamas, Bouzouki... **Occidental:** Accordions, Dobro, Jumbo Bottleneck... **South America:** Cuatro, Latin Percussion... **Spanish-Gypsy:** Flamenco Percussion & Guitar... **West Indies:** Cuban Percussion, Steel Drums... **World Synths:** Ambient Chords, World Pads... **Xtra Percussion:** Agogo, Crecels, Daires, Darbuka...

MUSICTRACK LIMITED • 2 THE GRANARY BUILDINGS • MILLOW • NEAR DUNTON • BEDFORDSHIRE • SG18 8RH • TEL: 01767 313447 • FAX: 01767 313557
www.musictrack.co.uk

TSC

THE SOLUTIONS COMPANY ◈ London's Largest Apple Centre

Solution Expert

Home Page | Hardware | Software | Cameras | Solutions | Music | Accessories

Pro Speakers

GENELEC

8020A *Biamplified Monitoring System*
- Minimum Diffraction Enclosure™
- Directivity Control Waveguide™
- 4" Woofer, 20 Watt + 20 Watt — £339 (£398.33 inc VAT)

8030A *Biamplified Monitoring System*
- Minimum Diffraction Enclosure™
- Directivity Control Waveguide™
- 5" Woofer, 40Watt + 40Watt — £552 (£648.60 inc VAT)

8040A *Biamplified Monitoring System*
- Minimum Diffraction Enclosure™
- Directivity Control Waveguide™
- 6.5" Woofer, 90 Watt + 90 Watt — £850 (£998.75 inc VAT) EX VAT

KRK

V4 MkII *Two-Way Active Powered Monitor*
- 1" Soft Dome Tweeter, 4" Kevlar Woofer
- 90 watt Bi-amp
- 23.6 x 15.9 x 19.7 cm — £187 (£219.73 inc VAT)

V6 MkII *Biamplified Monitoring System*
- 1" Soft Dome Tweeter, 6" Kevlar Woofer
- 120 watt Bi-amp
- 34.6 x 24 x 25.4 cm — £249 (£292.58 inc VAT)

V8 MkII *Biamplified Monitoring System*
- 1" Soft Dome Tweeter, 8" Kevlar Woofer
- 180 watt Bi-amp
- 40.3 x 27.9 x 30.5 cm — £379 (£445.33 inc VAT) EX VAT

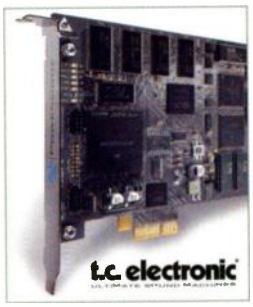
DSP Cards

TC ELECTRONIC

PowerCore Kompact
- FireWire connectors and the external power supply
- 12 powerful plug-ins right out of the box
- Portable solution — £349 (£410.07 inc VAT)

PowerCore Express — £765 (£899.00 inc VAT)

PowerCore Firewire — £680 (£799.00 inc VAT) EX VAT



FOCUSRITE

Liquid Mix Firewire
- Mac Only: Requires Mac OS X10.3.9 or higher, Intel supported
- 20 EQs and 40 Compressors out of the box, more online
- 32 Channels of Compression and 32 EQs
- Unique 7-band super EQs can be created — £424 (£498.20 inc VAT) EX VAT

Controller Keyboards

Remote SL25

novation — In music, anything is possible.

- 2-octave USB controller keyboard
- 8 musically-responsive trigger pads
- Fantastic new automap mode
- 2 giant lcd screens

Also Available:
Remote SL37 £314 (£368.95 inc Vat)
Remote SL61 £358 (£420.65 inc Vat)

£280 EX VAT (£329.00 INC VAT)

EDIROL

PCR-M50 *USB MIDI Keyboard Controller*
- 49 velocity-sensitive keys
- 27 assignable controls
- Switchable MIDI Interface
- 12 velocity curves — £140 (£164.50 inc VAT)

PCR-M80 *USB MIDI Keyboard Controller*
- 61 velocity-sensitive keys
- Sustain and Expression pedal ports
- Switchable MIDI Interface
- 27 assignable MIDI controllers — £186 (£218.55 inc VAT) EX VAT

M-AUDIO

Keystation 49E	£68	(£79.90 inc VAT)
Keystation 61	£Call	
Keystation 88LE	£Call	
Keystation Pro 88	£255	(£299.63 inc VAT)
Axiom 25	£92	(£108.10 inc VAT)
Axiom 49	£140	(£164.50 inc VAT)
Axiom 61	£160	(£188.00 inc VAT) EX VAT

Audio Interfaces

TC ELECTRONIC

Konnekt 24D
- Real-time DSP effects; Fabrik R & Fabrik C based on MINT™
- Analog volume control for perfect integration with powered speakers
- Network up to 4 units via the TC NEAR™
- Outstanding IMPACT™ mic preamps — £348 (£408.90 inc VAT)

Konnekt 8 — £220 (£258.50 inc VAT) EX VAT

FOCUSRITE

Saffire LE
- Ultra-low latency Firewire interface
- Unprecedented audio and conversion performance for the price
- 24-bit, 96kHz integrated A-D/D-A
- 6 inputs, 8 Outputs — £203 (£238.53 inc VAT)

Saffire — £254 (£298.45 inc VAT)

Saffire Pro 26 I/O — £424 (£498.20 inc VAT) EX VAT

MOTU

828 MkII (Firewire & USB)
- CueMix DSP™ near-zero monitoring system
- Front-panel control & Stand-alone operation
- Two mic pre-amps with switchable 48-volt phantom power
- 8 channels of 24-bit ADAT optical input/output
- 20 inputs / 22 outputs — £469 (£551.08 inc VAT)

Ultra Lite — £369 (£433.58 inc VAT)

Traveller — £649 (£762.58 inc VAT)

896HD — £649 (£762.58 inc VAT) EX VAT

DIGIDESIGN

M-Box 2
- Superior sound quality
- 4 simultaneous inputs and outputs with analog and digital I/O*
- 2 analog inputs and two analog outputs
- 2 channels of S/PDIF digital I/O (RCA)
- 48V phantom power for condenser microphones
- 24-bit/48 kHz audio, MIDI in/out
- Zero-latency monitoring, USB powered — £276 (£324.30 inc VAT)

002 Rack (Firewire & USB) — £680 (£799.00 inc VAT) EX VAT

APOGEE

Ensembler — £1195 (£1404.13 inc VAT) EX VAT

RME

FireFace 400 — £550 (£646.25 inc VAT)

FireFace 800 — £765 (£898.88 inc VAT) EX VAT

MACKIE

Onyx 1220 — £361 (£424.18 inc VAT)

Onyx 1620 — £553 (£649.78 inc VAT)

Onyx 1640 — £850 (£998.75 inc VAT)

Onyx Range Firewire Cards — from £288 (£338.40 inc VAT) EX VAT

Microphones

Samson C01U

USB Condenser Microphone
- Smooth, flat frequency response
- Cardioid pickup pattern
- Large 19mm diaphragm
- Heavy gauge mesh grill
- USB connection cable included
- High-pass filter and Phase Switch

£Call now — Or go online at gotsc.com

Complete Range of Rode, Audio Technica and Sennheiser Microphones available online at www.gotsc.com

Software

Logic Pro 7.2 Universal

The industry-leading application for music creation and audio - provides the most comprehensive collection of music creation tools

Plus SAVE £70!

£549 EX VAT (£645.08 INC VAT)

Save £70! Logic Pro 7.1
When you add Logic Pro to any new PowerBook, MacBook Pro or PowerMac G5*

*At time of purchase. Cannot be used in conjunction with any other offer. Incl VAT.

TSC: No. 1 Amalgamated Drive
West Cross Centre, Great West Road,
Brentford TW8 9EZ

TEL: 020 8400 1222
Call 9am - 6pm Mon - Sat

WEB: gotsc.com
Check Out our website for all the latest prices!

Money Isn't Everything

While we at *SOS* all like to try out and write about the latest and greatest gadgets, we're also conscious that some people have very tight budgets. One of the more common questions we get asked is about how to make good quality recordings without spending a fortune on gear. Fortunately, the vast majority of recording equipment available today is capable of producing extremely good results, and by paying less, you tend to lose out more on features than on quality. Clearly, high-end gear does sound better, but I'm always amazed by what can be achieved with the cheaper stuff. Many of my early four-track analogue recordings sounded fine, even though I was recording a whole band with a full drum kit — and back then there were no affordable capacitor mics, so we used the same dynamic mics we used for gigging.

Nowadays most budget multitrackers offer a minimum of eight tracks, and digital technology has effectively made tape hiss history. Whether you choose hardware or computer software is up to you and depends on your preferred way of working, but there are plenty of options and all can produce great-sounding results in the right hands.

In addition to your multitrack system, you need some sensibly accurate monitor speakers and one or more microphones. You can compare speaker price and performance by looking up reviews on our web site, but with only a few exceptions, most monitors costing over a couple of hundred pounds can be used to handle serious mixing providing you familiarise yourself with them by playing commercial CDs in your studio at every opportunity.

Pretty much the same is true of microphones. As with all gear, you can spend more money and get more quality, but there are very few studio mics out there now that can't produce good results. If you pay £100 upwards for a large-diaphragm capacitor mic, you can expect it to do a more than reasonable job of recording vocals when used with a pop shield and basic acoustic treatment.

The real secret to making good recording without overspending is to spend your financial budget wisely but not to be restrictive with your personal time budget, because you need to build up the experience necessary to use your equipment to its best advantage.

In addition to trying out all the recording techniques you read about, you also need to be aware of the acoustics of your monitoring environment and, where necessary, take steps to improve things in that area. Listen critically to commercial recordings to see how sounds have been chosen and how they fit together, then apply what you have learned to your own recordings. After all, it is no use being able to record perfectly if what you're recording isn't right in the first place. When listening to records, ask yourself what style of vocal reverb they used. Can you hear the effects of compression? Can you hear levels being adjusted to make way for vocals or solos? Start off simply by combining only four or five elements in a mix, and whenever you get the urge to add something new, make yourself justify its place in the piece of music rather than simply adding more because you can.

There's so much to learn that the only sane way to start is by keeping your projects as simple as possible and then adding complexity as and when your experience can handle it. As you've probably guessed all along, choosing the right gear is only part of the story — you will always be the most critical part of your recording system.

Paul White *Editor In Chief*

SOUND ON SOUND
Media House,
Trafalgar Way, Bar Hill,
Cambridge CB3 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 Mike Senior
Columns Editor Debbie Poyser
News Editor David Greeves
Assistant Editor David Glasper
Staff Writer Chris Mayes-Wright

Publisher Dave Lockwood

advertising
E adsales@soundonsound.com
Group Sales Manager Robert Cottee
Classified Sales Manager Patrick Shelley

production
E graphics@soundonsound.com
Production & Event Manager Andy Brookes
Designers Alan Edwards & Andy Baldwin
Digital Media Production Andy Baldwin
Classified Production Michael Groves
New Media Manager Paul Gilby

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
W www.soundonsound.com/subs
Circulation Administrator Luci Smith
Circulation Assistant Lisa Pope

UK £48
Europe (EU) £60
Europe (non-EU) £75
Rest Of World £75
eSub 12 issues £24
eSub 6 issues £15
eSub 3 issues £9
Payable in Pounds Sterling through a UK bank

Printing Warners Midlands plc
Colour Scanning Impress-Repro By Design Ltd
Newstrade Distribution Warners Group Distribution Ltd, The Maltings, Manor Lane, Bourne, Lincolnshire PE10 9PH, UK.

SOS SOS SOS SOS **ISSN 0951-6816**
A Member of the SOS Publications Group

Solid State Logic

SOUND || VISION

Duende. The SSL for your Audio Workstation.

The legendary sound of SSL for your digital audio workstation

Up to 32 channels of console-grade SSL EQ and Dynamics processing

Powerful DSP engine based on SSL's award-winning C-Series mixing consoles

Relieves the plug-in processing burden on your computer's CPU

Single FireWire connection for fast set-up

DUENDE STEREO BUS COMPRESSOR

DUENDE EQ & DYNAMICS CHANNEL

You already know the sound of an SSL console – you've heard it on countless hit records from the last four decades. Now those superior mix characteristics are available within your digital audio workstation, seamlessly integrated as powerful plug-ins served up on a custom DSP platform.

Based on the digital technology behind SSL's C-Series consoles, Duende delivers the complete SSL mix experience – powerful channel and dynamics processing and the legendary Stereo Bus Compressor. The easiest way to get professional mixes that sound like a record.

www.solid-state-logic.com

DUENDE

Solid State Logic

SOUND || VISION

Distributed in the UK by Sound Technology plc. Tel: 01462 480000 Web: www.soundtech.co.uk

in this issue

August 2006
issue 10
volume 21

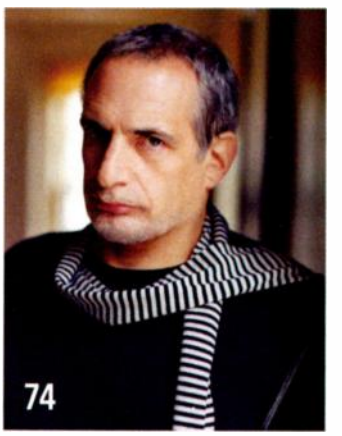
www.soundonsound.com

SSL Duende 112

techniques

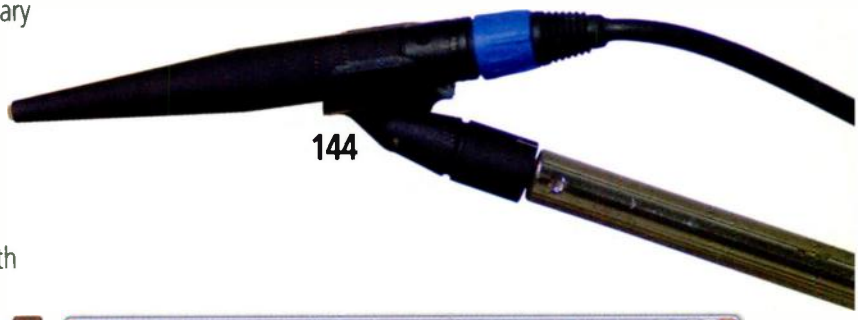
features

regulars



product tests


118


144

62

166

112

Live 5 gets hit for 6

Following Ableton's successes with *Live* v5, the backbone of many live performances around the world, they're upgrading it already! So let's cut to the chase — here are some of the new features that *Live 6* will offer.

Deep Freeze is a new function that allows frozen tracks to be edited while keeping processor load down. Clips can be cut, copied, pasted and consolidated, and you can draw and edit mix automation without unfreezing. A new 'flatten' mode commits all effects and automation to the track, rendering a new audio file. Movie Import lets you drag and drop movie clips into a session, create Warp markers and process the movie's audio like any other audio in *Live*. With a new instruments and effects racks feature, several instruments and effects can be grouped together and controlled in unison using eight macro controls. This allows simple editing of multiple parameters within the rack, using a single knob. An Essential Instrument Collection ships with *Live 6*, giving the user an array of useful sounds (like Steinway Grand, Fender Precision Bass and numerous string patches), as well as providing pre-programmed macro controls. Ableton will also supply maps for controller keyboards from Akai, Alesis, Behringer, Korg, M-Audio and Novation with the new version.

Other features of the update include improvements to some of *Live*'s current functions. The *EQ Four* parametric EQ module has been replaced by *EQ Eight*, giving the user twice as many filter bands as well as control of left, right, Middle and Sides channels. There is a new *Dynamic Tube* simulator, which emulates the sound of vacuum tubes, while *Live*'s existing *Saturator* module has been updated to include a user-definable waveshaper. *Live 6* also supports multi-core and multi-processor Macs and PCs, greatly improving its performance on those machines.

Alongside the launch of *Live 6*, Ableton are releasing the aptly named *Sampler*, which is an add-on to *Live*, similar in concept to *Operator*, *Live*'s first add-on module. *Sampler* has all the multisampling features you'd expect of a serious sampler and will import most sample formats, including *EXS*, Akai, REX, *Kontakt* and Soundfont. *Live 6* will be shipping in mid-September and although UK prices were yet to be confirmed at the time of going to press, the Euro prices of 549 Euros for *Live 6* (roughly equivalent to £380) and less than 200 Euros (£140) for *Sampler* should give you a rough idea.

M-Audio UK +44 (0)1923 204010 www.maudio.co.uk www.ableton.com

Happy Birthday NI!

Ten years at the top

This summer, software designers Native Instruments are celebrating 10 years of trading in the music technology industry. To mark the event, they have announced special offers on a variety of their products alongside a lottery with over £10,000 worth of prizes. The whole event, which has been dubbed NI-Versary, will culminate with a party in Berlin on 23rd September, to which NI lottery winners will be invited.

The following deals will be offered until September 10th: *Traktor DJ Studio* will cost £80 instead of £189.99; *Absynth 3* will be half price, at just £99.99; *FM7* is only £79.99, down from £219.99; and *Guitar Combos* is £50, instead of £129. An NI-Versary bundle of *Kontakt 2* (£229), which includes *Kontakt Experience* and a tutorial DVD, plus either East West *Adrenaline*, *Percussive Adventures 2 LE* or *Stormbreaks*, is also available as part of the promotion. Further offers include crossgrades for current Native Instruments users and discounted prices when upgrading to *Komplete 3* from *Kontakt*, *Guitar Rig* or *Reaktor*. The NI lottery will take place on the Native Instruments web site and winners will receive prizes from Gibson, Allen & Heath, JBL, Doepfer and Fujitsu-Siemens. More information can be found at www.ni-versary.com.

Arbiter Music Technology +44 (0)20 8207 7880.

www.arbitermt.co.uk www.native-instruments.com

Neve 8800 range

Build your own console

Neve's new 8800 range of analogue processing equipment offers a 'build your own' concept to potential Neve buyers. The new products are designed to integrate with the 8816 summing mixer and 8804 fader pack (reviewed this month on page 80) and are compatible with Neve's *Recall* Mac and PC software, which lets users save settings between sessions. Apart from the 8816 and 8804, the 8800 range (which is not to be confused with the Neve 8800 A-D converter) consists of three other products — the 8801 channel strip, the 8802 dual dynamics processor and the 8803 dual EQ.

The 8801 is a 1U mono channel strip based on Neve's highly acclaimed 88R console. It features mic, line and instrument inputs with switchable phantom power, phase inversion and

-20dB pad. There's also a four-band EQ section along with high-pass and low-pass filters. The dynamics section has a gate and a compressor with separate metering for gain reduction and signal level, and there's a flexible side-chain arrangement, with the option to add the low- and high-pass filters and EQ outputs to the side-chain. What's more, the EQ section can be switched to pre-dynamics, while the side-chain inserts can be switched to pre-EQ, providing a versatile

range of routing options. The 8801 is available now, costing £1650.

The 8802 dual dynamics processor is a two-channel limiting compressor and gate in a 1U box. Its features are similar to those found in the dynamics section of the 8801, with the addition of a Gate Attack control and the Key switch, which engages the side-chain. It has a link mode that, when active, connects the side-chains from the two channels, changing the unit's operation from dual-mono to stereo. The Neve 8802 costs £1250.

Finally, the 8803 (£1250) is a 1U dual-channel EQ featuring high-pass and low-pass filters, low- and high-frequency shelving bands and two fully parametric mid-range bands. Both channels have input gain controls, as well as 'EQ In' switches for quick comparison of

signal with and without EQ. For more on Neve's whole range, check out their web site.

AMS Neve +44 (0)1282 417281.

www.ams-neve.com

Back to the drawing board with CME

Have you ever wanted to design your own keyboard? Well, here's your chance. Chinese keyboard manufacturers CME have announced a graphic design contest for their UF range of master keyboards. To enter, all you need to do is download CME's design template, create custom graphics for the keyboard and post your design back on the site. The winner of the competition will receive the new VX7 controller keyboard, which features nine motorised faders, 12 velocity-sensitive trigger pads, eight knobs and nine rotary encoders. The first runner-up will get a feature-packed CME Bit Stream 3X controller. Further smaller prizes will be given to a second runner-up and three other qualifying entrants. For more information on the prizes and how to enter, check out CME's web site. We first announced CME's new VX-series controller keyboards in the news section of *SOS* April 2006, and we can now tell you how much they will cost: the VX5 will be £449, the VX6 £549, the VX7 £649 and the VX8 £749. The whole range will ship at the beginning of August.

Arbiter +44 (0)20 8207 7880.

www.arbitermt.co.uk www.cme-pro.com

Clarion Events to host Sounds Expo

Annual music technology and sound recording exposition Sounds Expo has been acquired by Clarion Events, who already host Music Live at the NEC in Birmingham. Kieron Seth, from previous organisers VCM Events, who launched Sounds Expo five years ago, remarked: "We would like to thank all the exhibitors and partners that have supported Sounds Expo and wish them well for next year's show." Sounds Expo 2006, which set up camp at London's Olympia back in March, featured seminars from *Sound On Sound* amongst exhibits from over 70 manufacturers, distributors and educational institutes. The next Sounds Expo, with Clarion at the helm, will return to Olympia in March 2007. See you there!

www.clarionevents.com

AAS release *Lounge Lizard Session*

Based on *Lounge Lizard EP3*, AAS's physically modelled electric piano soft synth, *Lounge Lizard Session* features four presets from *EP3*'s line-up of piano models — three Rhodes emulations and a Wurlitzer — plus a range of effects that includes distortion, tremolo and chorus. It can be used in stand-alone mode or as a plug-in in VST, Direct X (PC only), Audio Units or RTAS (both Mac OS only) formats, and offers a lower-cost way of accessing the *Lounge Lizard* sound. *Lounge Lizard Session* is suitable for PCs running Windows 98 or higher and Macs running OS 10.2 or higher, with Intel Mac support expected soon via a free upgrade. *Lounge Lizard Session* is only available as a download from AAS's web site, costing $99 (around £55). The full

Lounge Lizard EP3 plug-in is distributed in the UK by SCV.

SCV London +44 (0)20 8418 0778 **www.scvlondon.co.uk**

www.applied-acoustics.com

Make friends and music with Digital Musician collaboration community

Berlin-based software designers Digital Musician have announced an update for their *Digital Musician Link* (*DML*) VST peer-to-peer music collaboration software, with which members can record from remote locations using a VST-compatible host like *Cubase* or *Nuendo*.

The software, which is the brainchild of some of Steinberg's founding members, features a talkback system with webcam support, latency compensation and, depending on which version you use and your connection speed, audio transfer rates of up to 256kbps. Certain users can also use a file-transfer service to send audio files to each other.

There are three different account types — Free, Pro and Studio. The Free account is exactly that: just download the software from the web site and away you go, although transfer rates are restricted to 128kbps. The Pro and Studio options cost 89 Euros and 288 Euros per year respectively, but offer 256kbps transfer rates, unlimited file transfers and extra on-line features. All three accounts include membership of an on-line community of musicians, producers and engineers, where users can scout for collaborators to join their project, specifying genre and talent in their search, before auditioning and recording players over the internet using the *DML* VST plug-in. Already, Digital Musician has backing from EMI, and there are rumours that major labels may begin using private *DML* projects to run sessions from multiple studios.

Digital Musician also offer monthly and quarterly membership fees for their Pro and Studio accounts. One month as a Pro user will cost just 10 Euros, with a Studio account costing 35 Euros. A three-month membership will cost 25 Euros and 99 Euros for Pro and Studio members respectively. All users will need a fast Internet connection with at least 128kbps upload and 768kbps download speeds and a Mac running Mac OS 10.3 or above, or a PC with Windows XP SP2, as well as a VST-compatible host such as *Cubase*, *Live* or *Nuendo*. The talkback feature requires a basic microphone and, if you want to see your collaborator, a webcam. Check out the Digital Musician web site for more information.

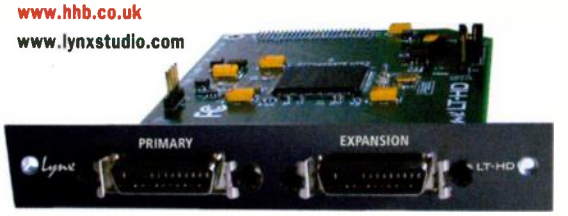
www.digitalmusician.net

Lynx announce Pro Tools option for Aurora range

High-end hardware manufacturers Lynx have released a new Pro Tools interface card for their Aurora 8 and Aurora 16 A-D/D-A converters. The LT-HD card uses the Aurora's LSlot expansion bay, allowing LT-HD-equipped units to interface directly with Pro Tools HD systems that use Digidesign's HD or Accel Core cards.

The LT-HD features one primary port and one expansion port, allowing a total of 32 tracks of A-D/D-A conversion per Core card when using a pair of Aurora 16s. Full track counts are supported at sample rates up to 192kHz, and Lynx's Synchrolock low-jitter word clock can be used as a master sync source when connecting to other digital equipment. For more information on the Aurora 16, check out Martin Walker's review in *SOS* June 2006. The Lynx LT-HD card will be available this month, costing £352.

HHB Communications +44 (0)20 8962 5000.
www.hhb.co.uk
www.lynxstudio.com



Digidesign announce Intel Mac support for *Pro Tools LE* and *M-Powered*

Digidesign are among many manufacturers who have recently made their software compatible with Apple's new Intel-based Macs. *Pro Tools v7.1.1* enables Mac Book, Mac Book Pro, iMac and Mac Mini owners to use *LE* and *M-Powered* versions of the software with the relevant Digidesign or M-Audio hardware. Other Digidesign software and plug-ins, such as *Music Production Toolkit*, *Digi Translator* and *DV Toolkit*, have also been made compatible with Intel Macs. Updating to *Pro Tools LE* or *M-Powered* v7.1.1 is free for all users of version 7 or higher. For more information and to download updates, visit Digidesign's web site.

Digidesign UK +44 (0)1753 655999.
www.digidesign.com

New features for Papen's *Albino 3*

Software designer Rob Papen has announced a new version of his *Albino* soft synth. Features in *Albino 3* include two new filters (Scream and Comb), new effects, a compressor and improvements to the reverb algorithm. What's more, each preset now has four separate layers, allowing you to construct more complex sounds. *Albino* costs just £132.95 from Time + Space Distribution in the UK. An upgrade is available for existing *Albino 2* users, at a cost of 39 Euros (under £30), although it's free if the software was purchased after December 2005. Visit the Rob Papen web site for upgrades and more information.

Time + Space +44 (0)1837 55200.
www.timespace.com www.robpapen.com

The road's **piano**

light, sound, action

30 years ago, Yamaha launched the legendary CP70. It was the first stage piano that actually played and sounded like an acoustic grand. Times change but the need for a realistic, versatile, portable piano remains. Benefiting from Yamaha's unrivalled digital technology, as well as a century's experience in crafting some of the world's finest grand pianos, our new CP Series reaffirms Yamaha's position as leaders in the field. You need never compromise your live piano sound again.

CP33 • 88 note GH keyboard • 64 note polyphony • 14 voices • 18 kg

CP300 • 88 note GH keyboard • 128 note polyphony • 530 voices • built-in speakers • 33 kg

To find out more, visit www.yamahasynth.com

YAMAHA
MUSIC PRODUCTION

A lovely pair from Brauner

We first saw Brauner's new VMX microphones way back in March at Frankfurt's Musikmesse. Now Brauner have announced that both the VMX and VMX Lite are shipping. In keeping with the company's impeccable reputation for studio microphones, both have been made to very high standards, boasting robust shockmount designs, high-quality tube circuitry and external power supply and pattern-control units. The main difference between the two models lies in the polar pattern, although they also ship with different shockmounts. The standard VMX (pictured) features infinitely variable pickup responses — controlled from the power supply unit — and a standard shockmount similar to that of Brauner's VMA and VM1 models. The VMX Lite features only two polar patterns, cardioid and omnidirectional, and comes with an unusual-looking shockmount with a clip-on design, which is similar to that of the Brauner Valvet. Both microphones have a very low self-noise figure of just 8dB, a maximum SPL of 142dB and a specially designed mic-to-power supply cable made by Swiss manufacturers Vovox.

Unsurprisingly, the Brauner VMX series doesn't come cheap. Although UK prices were unavailable when we went to press, you should be able to get a good idea from the Euro prices — 4032 Euros for the standard version and around 3220 Euros for the VMX Lite. Distribution is through SEA Vertrieb in Germany, but many of the UK's pro audio dealers will be able to source Brauner microphones.

SEA Vertrieb +49 (0)59 03 93 88 28
www.sea-vertrieb.de
www.brauner-microphones.com

Back to school with Sound Technology

Sound Technology, who exclusively distributed *Logic* before its parent Emagic was acquired by Apple, are now an Apple Authorised Training Centre. Their focus will be on the *Logic* 101 course, which is ideal for new users and aims to introduce *Logic*'s main feature set and user interface to students. Both *Pro* and *Express* versions of *Logic* are covered, with course topics including software instruments, Apple Loops, MIDI, audio recording and effects.

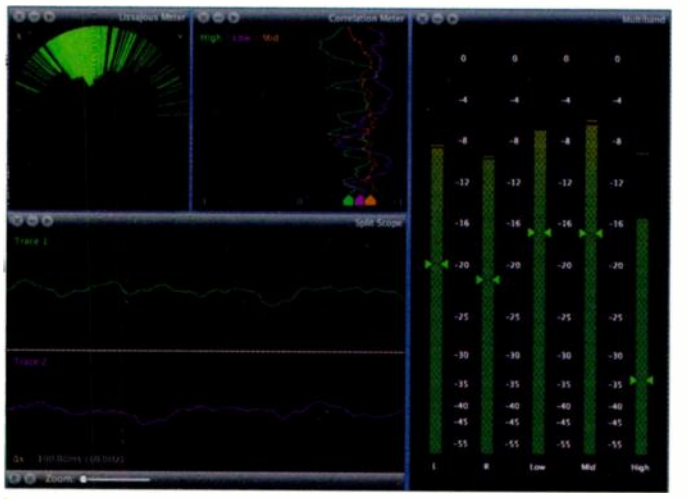
Sound Technology became sole UK distributor of *Logic* in 1992 and provided full customer support to users until August last year, when Apple began to offer software support under their Apple Care scheme. Staff from Sound Technology's *Logic* support team are being used to deliver the new course.

Included in the cost of *Logic* 101 (£470) is a multiple-choice exam. Upon successful completion of this, students will become an Apple Certified Pro. For those who don't want to take the exam, the course is available at a reduced price of £412. More information, including dates and contact details, can be found on Sound Technology's web site. Watch out for the forthcoming *Logic* 301 intermediate course.

Sound Technology +44 (0)1462 480000
www.soundtech.co.uk
www.apple.com/uk

Spectre Magic metering software for Mac

The latest application from US-based software programmers Audiofile Engineering is *Spectre*, an audio analysis tool for Macs running OS 10.4 (Tiger). It features a variety of tools, including spectrograms, spectrographs, Lissajous displays, VU and phase-correlation meters, alongside flexible multi-band displays, oscilloscopes and standard bargraph readouts. *Spectre* works as a stand-alone program and doesn't require integration with any host software. It was built using Cocoa, a Mac OS X application-programming interface that takes advantage of system-level processes like Quartz and Core Audio. This allows meter windows to be resized without interruption, and facilitates analysis of both local and networked audio devices. The latter feature means that a dedicated *Spectre* computer can be used as a multi-purpose metering tool on a networked system using Tiger's *AU Net Send* and *AU Net Receive* Audio Units. *Spectre* is available from Audiofile Engineering's web site and costs $150 (around £82).



www.audiofile-engineering.com

Analog Factory

One intuitive softsynth
2000 legendary sounds

- A huge palette of high quality synthesizer presets, for any musical style
- Smart Preset Manager : filter your searchs, and find your sound quickly
- Intuitive interface : edit each sound with the most essential parameters
- Integrate flawlessy with your midi controler keyboard in just a few clicks
- TAE® inside : advanced analog synthesis engine

www.arbiter.co.uk

YAMAHA
Distributed in the US by Yamaha Corporation of America

Arturia
www.arturia.com

Mastering for the McMasses

McDSP's *ML4000* mastering suite for *Pro Tools*

McDSP's latest release is *ML4000*, a two-piece plug-in set that includes the *ML1* single-band limiter and the *ML4* multi-band dynamics processor. *ML1* features what you'd expect from a single-band limiter, including variable threshold, knee, release and ceiling controls, plus a look-ahead mode and zero-latency operation. Its user interface is simple, too, offering input, output and gain-reduction meters. *ML4* uses the same algorithm, but adds four bands of multi-dynamics to the main limiter. Each band features a compressor, expander and gate, which can all be used simultaneously. Crossover frequencies are variable and can be controlled using the attractive graphical user interface. Both plug-ins can be used in mono or stereo modes, at sample rates up to 192kHz, depending on your hardware.

The *ML4000* package runs on Windows XP and Mac OS 10.4. It is available on its own — in TDM, RTAS and Audiosuite formats — costing £295, and as part of both *HD* and *Native* versions of the *Emerald Pack*, which cost £1615 and £795 respectively. Existing *Emerald Pack* customers can purchase *ML4000* for an nominal fee and there's an *LE* version included in McDSP's *Project Studio* bundle, which costs £351.

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

Universal Audio unveil Neve plug-ins and bundles

Universal Audio's UAD1 DSP-powered plug-in hosting systems already have a range of top-notch plug-ins, emulating past and present industry-standard equipment, but, until recently, none that bore the Neve name. So guess what: new for the UAD1 is the *1073 EQ* module, derived from the classic Neve console module of the same name. It features the same three-band EQ, high-pass filter, phase-inversion switch and input control as its namesake, and appears on screen in a similar form (although it displays horizontally instead of having the hardware's vertical design).

Included with *Neve 1073 EQ* is an *SE* version that uses less of the UAD1's DSP power, allowing higher instance counts. The plug-in is available in Audio Units, VST and RTAS formats (RTAS requires a VST to RTAS adaptor) from the Universal Audio web site, costing $249. There are also plans for further collaboration with Neve: software models of the 33609 buss compressor and 1081 EQ are on the cards (excuse the pun!).

Other news from Universal Audio is that two new packages have been announced for their recent UAD1e card for PCI Express. UAD1e Expert costs £820 and comes with $750 worth of plug-in vouchers for the my.uaudio.com shop, while UAD1e Express costs £390 and comes with $100 worth of vouchers. This allows the user to select their choice of plug-ins from the web site and build a custom setup. Check out UK distributor SCV's web site for details of upgrades and crossgrades for the UAD1, along with promotions for Universal Audio's entire hardware range.

SCV London +44 (0)20 8418 0778
www.scvlondon.co.uk
www.uaudio.com

DESKTOP THUNDER

MediaDesk.
Accurate, full-range monitoring on your desktop.

2+1

Making the subwoofer an integral part of the monitor system design enables the satellite speakers to be placed for the best imaging, and the sub to be positioned for the best bass response. Relieving the satellite speakers of the requirement to reproduce the low frequencies means that uncompromised, sealed-box designs can combine with sophisticated bass management to provide a seamless transition between sats and subs.
Hate that 'scooped out' sound? So do we.

MediaDesk. The first true 2.1 full-range professional monitoring system designed to produce a seamless spectral experience for computer audio recording and production, or listening to music without any compromises.

A powered system with a small footprint and a compact sub that fits under your workspace, MediaDesk uses 2-way satellite speakers with a dedicated 8" subwoofer. Blue Sky's Bass Management System puts the full, articulate bottom end exactly where it should be.

Easily expandable to a 5.1 Surround system, MediaDesk brings professional grade monitoring to a whole new world of applications in your home, studio, production suite, or broadcast studio.

5+1

With homes full of DVD players and HDTV a reality, surround sound isn't the future– it's here and now. That's why every 2.1 Blue Sky monitoring system is easily and quickly upgradable to 5.1. And such is the performance of Blue Sky surround sound monitors, many of our professional customers also choose to buy Blue Sky systems for their homes.
Welcome to Blue Sky's fresh new approach to accurate, full-range, stereo and surround sound monitoring.

"They sound remarkable. I even bought a second set of Sky One 5.1 monitors for my home for critical listening there."
Jerry Harrison– Keyboard/ Guitarist Talking heads

"Since I do so many different types of projects, from remixes to radio, I need consistent sound. That's where ProDesk monitors make all the difference."
Roger Sanchez– Grammy Award-Winning Engineer/ Remixer

blue sky

exclusive UK Distributor
beyerdynamic
www.beyerdynamic.co.uk
t: 01444 258258

Steinberg ship *Halion Symphonic Orchestra*

The latest incarnation of Steinberg's *Halion* software sampler features a massive 27GB sample library of orchestral sounds with an intuitive user interface. There are over 1200 brass, percussion, string and woodwind patches and combinations, each in 16- and 24-bit formats, which Steinberg claim can be used convincingly in everything from R&B to chamber music.

Halion Symphonic Orchestra (*HSO* for short) features Real Ambience — a function that adds recorded room ambience without the use of artificial reverb — along with the Crescendo Controller, which blends between sample layers in real time, allowing realistic control of the instrument's dynamics. There's also RAM Save, a freezing function that reduces strain on the computer's CPU.

HSO's user interface displays a strip of virtual knobs that control eight of the instrument's most useful parameters, below details of the patches in use, with options to change the MIDI channel, panning and output routing of each one. *HSO* is currently compatible with Windows XP on the PC and Mac OS 10.4 on Apple G5s, with a version for Intel Macs to follow shortly. It can either be used as a stand-alone software instrument, with Rewire support, or with a suitable host as a VST, Audio Units or Direct X plug-in.

The full package costs £399.99, or you can choose to crossgrade from any other (non-Steinberg) orchestral software instrument for £199.99. There are also upgrades available for *Halion String Edition* users — £149.99 for *Volume One* users and £69.99 if you have *Volume Two*. If you don't already own a Steinberg USB dongle, you'll need one for authorisation — they're obtainable from UK distributor Arbiter's web site for £19.99. Watch out for a review in *Sound On Sound* over the coming months.

Arbiter Music Technology +44 (0)20 8207 7880.
www.arbitermt.co.uk **www.steinberg.net**

Novation announce Xiosynth

Novation have announced the Xiosynth (£229), a compact synth with built-in audio interface. It features 200 patches, 60 of which were programmed by the likes of Ferry Corsten, Roots Manuva and Shimon, and has an on-board 32-step sequencer (called X-Gator), an arpeggiator and a range of effects, which can all sync to incoming MIDI clock. Real-time sound-manipulation fun is provided by a touchpad next to the 25 semi-weighted keys, and there's a joystick for control of pitch and modulation.

The Xiosynth's two-in, two-out USB audio interface has a mic preamp featuring phantom power, and instrument and line inputs. Any one of the inputs plus a mono synth sound can be transmitted simultaneously to the host computer via USB. The device is class-compliant with Windows XP and Mac OS X, meaning that no drivers are required, although some are included for greater stability and lower latency in certain circumstances. The synth can run on six AA batteries.

If you like the look of the Xiosynth but need a bigger keyboard, a 49-key version is planned for late Summer.

Focusrite +44 (0)1494 462246 **www.focusrite.com**
www.novationmusic.com

Kore deadline extension

In last month's issue, we featured Kore, Native Instruments' new hardware controller and software host for instruments and effects. At the time, NI were offering a bundle comprising Kore and *Komplete 3* for just £999, until 30th June. Due to the promotion's popularity, they've now extended that cutoff date to 31st July. The extension also applies to *Komplete 2* customers who, in return for buying Kore, get a free upgrade to *Komplete 3*. For more information, visit UK distributor Arbiter's web site.

Arbiter Music Technology
+44 (0)20 8207 7880.
www.arbitermt.co.uk
www.native-instruments.de

Pro Tools • Live • Logic • Cubase • Digital Performer • Sonar

One for all.

PROJECTMIX I/O

mix surface | 24/96 interface | 8 preamps | LCD

Today, more professional music is produced at home than ever before–and the new ProjectMix I/O delivers what you need to take your computer-based studio and productions to the next level. Seamless integration with all major DAW software. The ability to record directly into industry-standard Pro Tools sessions. Faders so you can feel the mix with your fingertips instead of dragging a mouse. On-board display of critical parameters for intuitive operation. Motorized control to craft more accurate mixes. And professional multi-channel I/O including mic/instrument preamps, Lightpipe and S/PDIF. ProjectMix I/O is the universal solution that combines the best of the hardware and software worlds for a new standard in streamlined production.

- **record audio directly into Pro Tools M-Powered** > industry-standard format
- **10-bit touch-sensitive motorized faders** > totally intuitive mixing and editing
- **built-in 18 x 14 FireWire audio interface** > no additional hardware required
- **8 phantom-powered mic/line preamps** > pro input right on board
- **assignable rotary encoders** > control mixes, effects and synths
- **LCD display** > full track & parameter readout
- **dedicated transport controls & jog/shuttle wheel** > seamless session flow
- **ADAT lightpipe, S/PDIF & word clock I/O** > total digital connectivity

See it now!
www.maudio.co.uk/projectmix

GET M-POWERED

M-AUDIO
www.maudio.co.uk

M-Audio, 6th Floor, Gresham House, Clarendon Road, Watford, Herts, WD17 1LA tel: 01923 204010 email: info@maudio.co.uk

Pro headware from AT

Audio-Technica launch new headphones

Audio-Technica's headphone range has been updated with a new model, the ATH Pro 700. They are designed primarily for extended listening by DJs and sound engineers alike, boasting giant 53mm drivers with neodymium magnets and a frequency response of 5Hz to 33kHz. Their closed-back design and high output level should allow accurate listening in loud environments, while the earpieces rotate for single-ear monitoring.

Included with the ATH Pro 700s are a carry pouch and the all-important screw-on 3.5mm-to-6.3mm stereo jack-plug adaptor. The 1.2m coiled cable exits the headphones from the left ear only, preventing tangles. The new 'phones are available now, in black or silver, at £119.95 including VAT.

Audio-Technica UK +44 (0)113 277 1441.
www.audio-technica.co.uk

M-Audio hit the airwaves with wireless MIDI range

M-Audio have announced a new line of wireless MIDI keyboards, adding to their current line of wired controllers. The Mid Air 25 (pictured) and Mid Air 37 will have the same feature set as a typical USB MIDI keyboard, with velocity-sensitive keys, modulation and pitch-bend wheels and pedal input, alongside eight assignable knobs and, on the Mid Air 37 only, nine programmable faders. Both products ship with a power adaptor, but take batteries for fully wireless operation.

Each Mid Air controller is equipped with a wireless MIDI interface made by the Frontier Design Group (check out our review of their Tranzport wireless controller in *SOS* July 2005, at www.soundonsound.com/sos/jul05/articles/frontier.htm) and includes a Mid Air Receiver (pictured above), which uses USB to connect to a Mac or PC. The receiver also features a standard USB-to-MIDI interface, allowing the simultaneous use of wired MIDI devices alongside wireless controllers.

M-Audio also plan to release a stand-alone wireless MIDI interface, allowing any MIDI-equipped device to be used up to 30 feet away from the receiver. We're not sure how much it will cost yet, but it will be available from August onwards. You'll also have to wait for pricing information about the Mid Air 37, as it is still to be confirmed. However, we can tell you that the Mid Air 25 controller keyboard will be shipping this month at a cost of £199. Check out M-Audio's web site for the latest news.

M-Audio +44 (0)1923 204010.
www.maudio.co.uk

Celemony announce *Melodyne* v3.1

Celemony's acclaimed pitch- and time-correction software, *Melodyne*, has had an update, with version 3.1 released for both *Melodyne Cre8* and *Melodyne Studio* packages. The update mainly addresses Intel-Mac compatibility and functionality improvements, as well as featuring an alternative to the standard challenge and response licence-authorisation procedure whereby users can choose to use an iLok USB dongle, making it easier to move between multiple computers.

Melodyne's Rewire stability has been improved, as has the polyphonic algorithm in *Melodyne Studio*. The new version now automatically reads the tempo information contained in audio files, including Apple Loops and *Acid* files, and exports tracks with the MIDI root key embedded.

Celemony have also announced the first sound libraries for *Melodyne* — *MSL Electric Bass*, *MSL Saxophone* and *MSL Guitar* — which integrate into *Melodyne*, allowing precise melody manipulation of a range of samples.

Version 3.1 is free to all registered *Melodyne Cre8* and *Studio* users running version 3 and the update can be downloaded from the Celemony web site. *Melodyne* can be used in stand-alone mode, connected with Rewire, or used as a VST, Audio Units or RTAS plug-in.

Arbiter Music Technology +44 (0)20 8207 7880.
www.arbitermt.co.uk www.celemony.com

AAS *Lounge Lizard* correction

In the Korg K61P item in the news section of *SOS* June 2006, we said that IK Multimedia are the makers of the *Lounge Lizard EP3* virtual instrument. This, of course, was a mistake — *Lounge Lizard EP3* is made by Applied Acoustic Systems (AAS). Apologies to all those concerned!

www.applied-acoustics.com

Get that authentic vinyl sound with *Sonitex STX1260*

Danish software developers Otium FX have released *Sonitex STX1260*, a VST plug-in for Windows that adds 'sonic texture' to samples and audio files. The plug-in is designed to emulate the sound of vinyl records from the mixing stage to pressing and playback, as well as the characteristics of hardware samplers. *Sonitex* has controls for the density, stereo spread and level of clicks and pops, and features pitch-instability presets, 20 different types of noise, a low-pass filter, bandwidth control and bit-depth reduction. It is only available in VST format for Windows and costs $49 (about £26) from Otium FX's web site. A free trial version will be available soon.

www.otiumfx.com

Submersible announce Drumcore 2

In *SOS* February 2005, we reviewed Submersible Music's *Drumcore* (www.soundonsound.com/sos/feb05/articles/drumcore.htm), concluding that, if you want drum tracks laid down by professional drummers with speedy auditioning, *Drumcore* is a good way to go. Just over a year later, it has been updated to include a wealth of useful extra features.

The new version includes more styles from world-famous drummers such as Alan White, who has provided material in odd meters, including 5/4, 7/8 and 9/8; Jon Bishop, who plays a selection of jazz brush material; and Terry Bozzio, drummer for Jeff Beck and Frank Zappa. *Drumcore* gives you both audio and MIDI versions of the grooves, which were recorded at 10bpm increments to capture the player's natural style, and users

can import and export their own REX 2 and *Acid* files, using *Drumcore* as a librarian.

Other new features include enhancements to the drum module, which now has a separate audio output with pitch and pan controls for each drum pad. These can be sent to up to 32 Rewire busses, enabling the user to process each drum separately using their Rewire host. There is also a new 'Gabrielizer' feature, which can analyse an audio track and intelligently apply variations to the drummer's audio and MIDI content, derived from the groove of the original.

If that's not enough, extra *Drummer Packs* are available from Submersible. There are four in total, costing £59 each (from Lonnie Wilson, Matt Sorum, Sly Dunbar and Zoro), but they can all be bought as a bundle for £149. *Drumcore* v2 costs £159, but you can buy *Drumcore Deluxe*, which includes all four *Drummer Packs*, for £319. Submersible Music's products are distributed by MI7 in Sweden; visit their web site for more details.

MI7 +46 4640 63 06 970 **www.mi7.com**
www.drumcore.com

This is London Calling

We like to keep our ears to the ground at *Sound On Sound*, so this month we went along to London Calling, one of the music industry's newest events. Earls Court was buzzing with activity from the off, with high-street retailers Sound Control at the door with a host of special offers. The usual throbbing of PA systems was courtesy of Void Acoustics — they provided the sound for both the Blue Hippo live stage and the Pioneer Pro DJ stage.

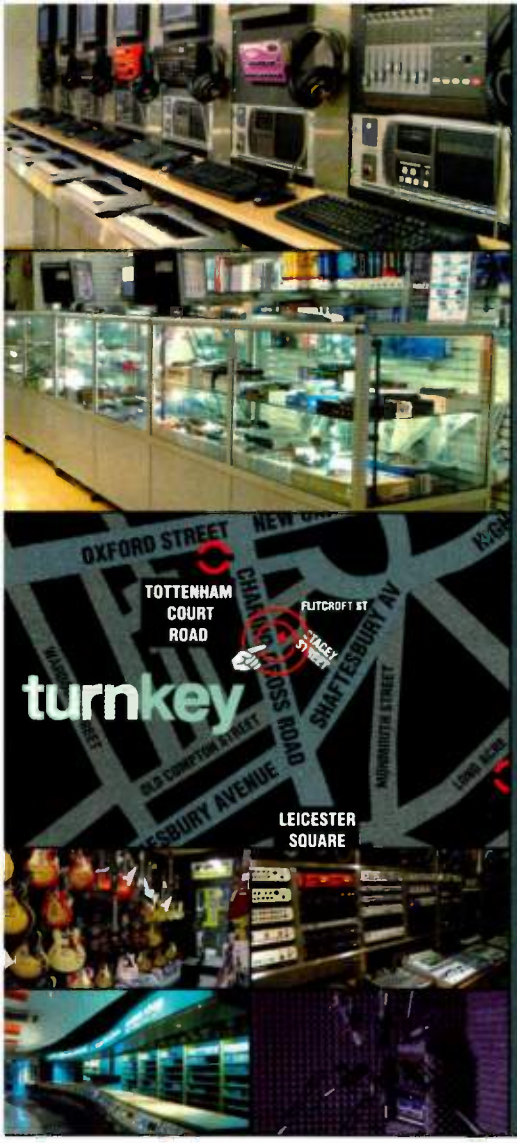
Among the exhibitors were M-Audio, who ran demos of Ableton *Live*, and Digidesign, who had their D-Command console on show. Record labels and on-line music companies also had a strong presence — independent breakbeat label and mail-order firm Streetwise Music had a stand, as did Ministry of Sound Digital and Positiva. Safesell and

emusu.com exhibited their DIY music-distribution services, and the MCPS-PRS alliance were there too, offering advice on licensing and royalty collection.

Educational establishments were well represented by the likes of Alchemea, SAE and School Of Sound. Most importantly, there was the VIP area and A&R pit, where record execs rubbed shoulders with musicians and producers, offering advice on what to do next.

London Calling will return to Earls Court at the end of June 2007.

www.londoncalling2006.com



turnkey

Now in business for almost 30 years, Turnkey is by far the largest single store operation in the UK, and as such we are able to invest unrivalled time, money and energy into making it a unique shopping experience.

SUBSCRIBE FOR FREE!

Get the Turnkey Buyer's Index and Soho Soundhouse Gear Guide delivered to your door FREE. Simply call us on 020 7419 9999 or go to turnkey.co.uk/catalogue

www.turnkey.co.uk

turnkey call 020 7419 9999 fax 020 7379 0093 www.turnkey.co.uk

EUROPE'S LOWEST PRICE GUARANTEE*

*Our guarantee is that we will beat any genuine quote on a like for like basis, on identical goods, from a UK or European dealer at the time of order. We must be able to substantiate the quote and the competitive dealer must have the goods in stock and available to sell.

THE No.1 in the UK

lowest price guarantee*

largest range of stock under one roof

biggest dealer for many suppliers

most comprehensive hi-tech demo facilities

most accessible in London and from all over the country

friendliest service in store and on the phone

only comprehensive catalogue

clearest website in the business

Store Opening times:

Mon - Wed: 10:00am - 6:00pm
Thursday: 10:00am - 7:00pm
Fri - Sat: 10:00am - 6:00pm
Sundays and public holidays: Closed unless otherwise advertised.

Phone lines open: 9am - 7pm Mon - Fri, 10am - 6pm Sat

HUGE SAVINGS!

Our buyers scour the planet daily to bring you the best deals anywhere - we've always got a wide range of superb value 'end of line' clearance deals available at a fraction of the original prices - just check out a few of the examples shown below!

£99 WORTH OF FREE STUFF!

While stocks last, we're giving away free VST plug-ins worth £99 with every telephone order for delivery over £200 – choose from Akai's superb DeccaBuddy intelligent harmonizer or the excellent QuadComp multi-band compressor.
(You must quote this offer at time of ordering to qualify.)

BUY NOW, Pay Nothing for a Year!

- No deposit! Nothing to pay for 12 months!
- Pay the balance in 12 months & pay no interest!
- Or pay over 36 months, 29.8% APR fixed for the term of the loan.

(Minimum spend £1,000 for 12 months, 6 months available for purchases over £500. See below for example)

REASON 3 & OZONIC
SOFTWARE & KEYBOARD BUNDLE

M-AUDIO

Price saving package deal on Propellerheads great Reason 3 software and the compact M-Audio Ozonic controller keyboard. Combines the leading all-in-one virtual studio software sequencer and mixing platform, with synthesis and sampling workstation and audio effects toolkit with flexible user configurations. Recycle looper, REX player, vocoder, step sequencer and '808' drumbox. New V3 M-Class Mastering Suite with Parametric EQ, Stereo Imager, Compressor, and lookahead Maximiser and Combinator tools, plus expanded sound libraries. Package includes the M-Audio Ozonic 37 note MIDI controller keyboard with Firewire 4 in 4 out audio and MIDI interfacing, self-powering facility and 40 user assignable controllers including 8 pots, 9 sliders and pitch / mod wheels. A powerful pairing!!!

RRP £629 **£329** MMAN-OZONICRE

1212M BUNDLE
SOUNDCARD & SOFT SAMPLER PACKAGE

E-MU

Exclusive cost saving E-mu bundle with 1212M external Firewire interface for ultra low latency MIDI and audio via Digidesign spec 24 bit 192kHz converters, 2 in 2 out balanced analogue, S/PDIF and ADAT optical. Latest version 1.5 software host Emulator X runs 24 bit 192kHz sampling with disk streaming, hardware accelerated effects, VSTi or standalone, E-mu morphing filters, graphic wave editing, etc. Finally, the Mo Phatt X sample library which includes 512 urban RnB, hip hop patches, SFX and groove patterns.

RRP £299 **£129** EMU-1212MBDN

CHECK turnkey.co.uk for the very latest deals!

We continually survey our competitors' prices to make sure you can't buy cheaper.*

NI KORE MUSIC PRODUCTION — RRP £370 **£349** NATI-KORE

ACCESS VIRUS TI POLAR SYNTH — FACTORY DIRECT **£1499** ACCE-TIPOLAR

DAVE SMITH MONO EVOLVER KEYBOARD SYNTHESIZER — FACTORY DIRECT **£849** DAVE-EVOLVEKB

EMU XBOARD 49 & MOPHATT X MIDI CONTROLLER & SOFT SYNTH BUNDLE — RRP £209 **£109** EMU-XBOARD49M

SENNHEISER FREEPORT VOCAL SET WIRELESS MIC SYSTEM — RRP £229 **£169** SENN-FPVOCAL

MACKIE CFX16 v2 16 CH ANALOGUE MIXER — RRP £550 **£449** MACK-CFX16V2

LITTLE PHATTY TRIBUTE EDITION by BOB MOOG ANALOGUE SYNTHESIZER — FACTORY DIRECT **£899** BIGB-PHATTYT

DSI EVOLVER RACK SOUND MODULE — FACTORY DIRECT **£999** DAVE-POLYEVOL

ETA PD8L POWER CONDITIONER — RRP £93 **£49** ETA-PD8LIEC

AKAI MPC2500 64 TRACK SEQUENCER / SAMPLER — ONLY **£999** AKAI-MPC2500

PRO TECH POPPER STOPPER WINDSHIELD — ONLY **£14** MICA-PSTOP

MACKIE HR824 ACTIVE NEARFIELD REFERENCE MONITORS — RRP £1400 **£899** MACK-HR824

ACCESS VIRUS CLASSIC SYNTHESIZER — FACTORY DIRECT **£699** ACCE-VIRUSCLA

EV SX300 PASSIVE PA SPEAKERS — RRP £1545 **£748** EV-SX300

MILLENIUM MIC STAND — FACTORY DIRECT **£9** MILL-MIC

SAMPLELAB BROKEN BEATS SAMPLE LIBRARY — FACTORY DIRECT **£59** CD-BROKEN

TANNOY SYSTEM 600A ACTIVE STUDIO MONITORS — RRP £849 **£349** TANN-S600A

VOYAGER RACK V3 by BOB MOOG RACKMOUNT SYNTH — FACTORY DIRECT **£1399** BIGB-VOYA/RV3

MAXELL CD-R SPINDLE 50 PACK BARGAIN MEDIA PACK — RRP £30 **£7** MED-MAXCD-50

AKAI MPD16 USB/MIDI DRUM PAD — RRP £180 **£79** AKAI-MPD16

Finance Example

Spend £1000. £0 deposit.

Either pay £1000 within 12 months of the date of purchase, total amount payable £1000, pay no interest.

Or 36 monthly payments of £40.49. Total amount payable £1,457.53. 29.8% APR typical. Interest calculated from date of agreement.

Payment by Direct Debit only. Credit subject to status. Applicants must be 18 or over.

020 7419 9999

email sales@turnkey.co.uk visit 114-116 Charing Cross Road London WC2H 0JR

WRH

The Carillon *ti* – a Breakthrough

ti total integration of *everything* you need to make music.

Carillon was founded to address the need of musicians to take the hassle out of computer music making – compatibility issues and the technical support merry-go-round were making the use of ground breaking technology a real grind.

Now having addressed that problem with the original AC-1 Audio Computer, we're going one step further with ***Total Integration***. We've used our reputation with suppliers to bring you the AC-1 ***ti*** – a world class audio PC with everything you need to make music from sequencing to scoring software, virtual instruments to controllers and top quality audio connections of all types totally integrated out of the box – just add a monitor. In addition to tweaking Windows to the max, we've optimised every PC component and application and ensured that everything works together completely smoothly and made it whisper quiet; and to get you up and running as quickly as possible, there's a comprehensive on-screen manual that covers the system as a whole including many easy to follow tutorials and troubleshooting guides. To top it all off, we've replaced the standard Windows desktop with our own super-clear interface that also keeps people from messing up your system.

Remote Support

A Carillon FIX modem included with every system allows our engineers to direct-dial into your machine and fix problems live!

Perhaps the most remarkable thing about the ***ti*** though is the price. With configurations starting from only £799 including music hardware and software that would cost well over £1,500 if bought separately, and free support, it's an absolute steal – you can't say fairer than that. Call or check the website for latest specifications.

Sonar Studio

A World Class Music Creation Environment

Always the best selling sequencer in the States, Sonar is now finally being recognised in the UK for the great application it is, with a top-notch feature list including up to 500kHz /32bit recording, track freeze, full delay compensation, MIDI FX, Rewire compatibility, loop construction tools, genuine Lexicon reverb plus a bunch of synths and FX. And let's be clear here – this is the FULL version of Sonar Studio 4, not a cut-down 'lite' version. Above all though, we chose Sonar for the ti for its killer combination of class leading functionality and a superb user interface – it really is a cinch to learn.

SOFTWARE

Loops

SampleLab Luscious Grooves – 5 star award from SOS with over 300Mb of a range of styles.

SampleLab Discography – another 5 star winner, full of house & breakbeat loops.

SampleLab Broken Beats – deep & dirty hip-hop, lo-fi & downbeat in full 24bit.

Sampling

Emu Emulator X – arguably the most powerful software sampler available. Includes a massive sound library.

Keyboard Instruments

Native Instruments B4 Express – Ultra-realistic Hammond sounds with software controllability.

Native Instruments FM7 Express – 64 great tweakable patches of classic FM.

Native Instruments Pro 53 Express – NI's first instrument - an utterly faithful reproduction of the classic Prophet 5.

Bread & Butter

Emu Proteus X – All 1024 presets from the legendary Proteus 2000.

Emu MoPhatt X – All 512 patches from the MoPhatt - an urban legend!

Drum Machine

Rayzoon Jamstix SE – a software drum machine that actually plays along with your tempo and dynamics!

Synths

SampleLab Analog Archive – the rarest collection of vintage synths ever sampled in awesome quality.

Emu Vintage X Pro Volume 1 – over 2Gb of vintage classics from the legendary Vintage Keys.

SFX

SampleLab Spatial Awareness – 9/10 from Computer Music - chill-out and ambient sounds abound!

Orchestral

Emu Virtuoso X – universally acclaimed as the best hardware orchestral module ever, all the patches are here.

Drum Hits

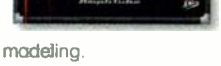
SampleLab Drum Fundamentals – over 2,250 drum sounds - natural and processed, acoustic and electronic

World

Emu Planet Earth X – a wealth of ethnic sounds from the masters in this area.

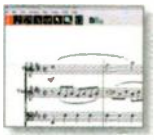
Amp Modeling

IK Amplitube LE – widely acknowledged as the best sounding amp modeling.



Scoring

Personal Composer 16 - quality score printing, transposition and part-extraction.



Ear Training

Music Goals - become the musician you know you can be!

DJ Mixing

Native Intruments Traktor DJ - full copy of V2 - acknowledged market leader.

NB Retail packaging is shown for illustration purposes only - all software and hardware is supplied pre-installed with on-screen manuals only

in Personal Audio Computing

from only **£799**

HARDWARE

MindPrint Front End

To enable the *ti* to interface to almost anything though, we've commissioned a completely new drive-bay unit, the HP-Pre. This features a superb mic pre-amp and instrument level input designed by channel strip gurus MindPrint, which also supplies 48v phantom power to allow you to record with any mic you like, straight out of the box. In addition, there is also a very clean high gain headphone amp built-in, crucially featuring an 'audio thru' facility which means you can still have your main soundcard outputs available for speakers, and can switch between keeping them on if the headphones are being used for foldback, or turning them off if the rest of your household need a break from your latest masterpiece!

MIDI Controllers

Eight front panel MIDI control knobs are provided and are handily pre-set and labelled for the most popular synth editing parameters and also amp modeling controls for the included AmpliTube LE. Of course you can easily re-route these to control pretty much anything you like - really brings music making to life!

Transport Control

Front panel transport controls not only ensure you always have these vital keys immediately to hand but also make the recording experience feel like you're using a dedicated piece of hardware.

Custom Keyboard

As well as the included optical mouse, there's also a QWERTY keyboard which we've custom made with all the Sonar shortcuts pre-printed in a clear colour coded system (Cubase and Pro Tools versions are optionally available) – great for getting to know your software quickly and easily.

Emu 0404

The No.1 Selling Professional Sound Card

When it comes to sound quality, Emu's PCI cards stand head and shoulders above the rest. 24bit/192kHz converters ensures absolutely pristine fidelity. I/O includes analogue, S/PDIF and MIDI, and if you need more, upgrading to another Emu card couldn't be simpler – just swap them out - no driver reconfiguration required.
Best of all though is the onboard DSP. In addition to allowing zero latency monitoring there's also a wide range of top quality VST FX included with zero hit on your CPU – like an extra GHz of processor speed.

Carillon Custom Shop

For more specific requirements, come to the Custom Shop - choose one of our Core systems and we'll build your dream!

AC-1D

Dual Core 1
- Intel Dual Core 3.2GHz
- 1GB DDR2 RAM
- 80GB+320GB SATA RAID
- DVDR/RW

only **£1269**

Dual Core 2
- Intel Dual Core 3.4GHz
- 2GB DDR2 RAM
- 80GB+640GB SATA RAID
- DVDR/RW

only **£1899**

AC-1X

Xeon
- Intel Xeon 3.4GHz x2
- 2GB DDR RAM
- 80GB+800GB SATA RAID
- DVDR/RW

only **£2499**

Laptop

Centrino
- Intel Core Duo T2400 1.8GHz
- 100GB 7200rpm HD
- 1GB DDR2 RAM
- DVDR/RW

only **£1399**

AD-1

24bit/96kHz recording, huge virtual instruments and massive track counts mean YOU NEED STORAGE!

Our AD-1 series of Firewire and USB2.0 hard drives are compatible with Macs and PCs, are optimised for use with audio recording software and are near silent in operation. Now available in sizes up to 400Gb!

- **AD-1 LE** Firewire 400
- **AD-1 Std** Firewire 400+USB2.0
- **AD-1 Plus** Firewire 400+800+USB2.0

From only **£99**

www.carillondirect.com
info@carillondirect.com

+44 (0)20 7692 7629

Also available from Turnkey & Soundcontrol.

Carillon

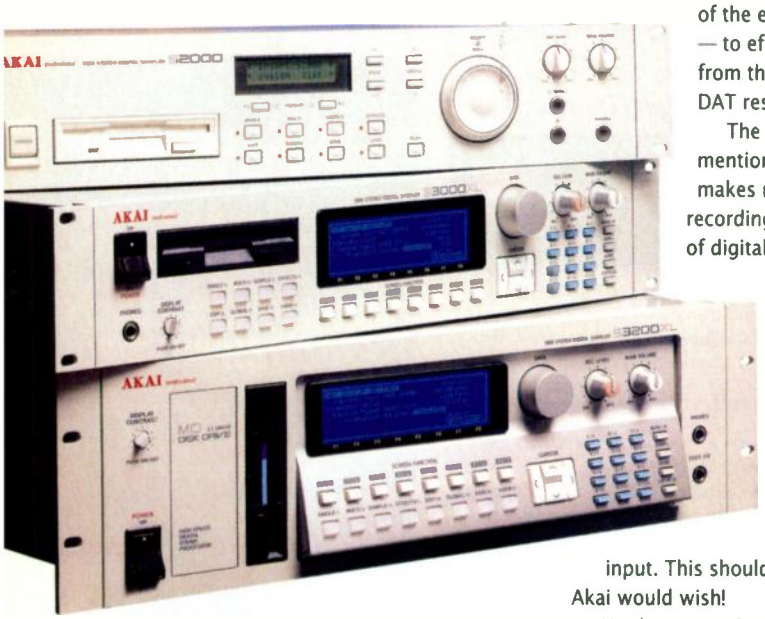
* Prices include VAT unless otherwise stated. All prices and specs correct at time of going to press but we reserve the right to change these without notice. Trademarks are the property of their various owners.

Q&A

Your technical questions and queries answered

How do I back up my Akai sampler's library?

I've been thinking about using the DAT backup/restore function to back up my Akai 3200XL sampler library. I've tried using my Sony CD-writer instead of a DAT machine and, although this records the data, it does not activate the restore procedure in the sampler when I play it back in. Could it be that the converters inside the CD-writer are 24-bit? If so, are there any 16-bit machines out there that you could recommend, as DAT machines seem to be unavailable?
Ian Tanner



Akai's digital output-equipped sampler range featured a DAT backup and restore function.

***SOS* contributor Steve Howell replies:** The only backup option available at the time of the S3200XL (and other Akai samplers of the era) was DAT, which was a more popular format then than it is now, of course. Via the sampler's digital out, the DAT backup function would send digital data — the programs and sample headers — and then the raw sample data (audio), so if you were to listen to the backup of, say, a choir sound, it would typically go 'blip blip blip thhhhrrrrrp blip blip thhhrrrrp aaaaaah aaaaaah aaaaaah aaaaaah aaaaaah' and so on, like a digital version of early cassette backups. If you wanted to put your samples back on the sampler's disk, the DAT restore function would feed the data back into the machine in the same format as it came out. By today's standards it was clunky and primitive, but it usually worked well.

Most significantly for this case, the sampler had no knowledge of the recording medium. At the time it was usually DAT because that was the only affordable digital recordable media available, but if the digital output of the 3200XL was fed to the digital input of a CD recorder, there is no reason why data wouldn't be backed up in the same way. And if the DAT restore function is invoked, it will wait for data — if that happens to come from a CD, the sampler won't know and couldn't care less; all it's concerned with is receiving a stream of digital data, which the CD will duly provide. After all, many Akai sound-library CD-ROMs of the era offered this facility — to effectively 'load' sounds from the CD-ROM using the DAT restore function.

The fact that you've mentioned D-A converters makes me think that you're recording an analogue version of digital backup, using the CD recorder's analogue inputs. You may as well do that to a cassette recorder — I'm afraid it simply won't work! To solve your problem, plug your sampler's digital output into your CD recorder's digital input. This should record the data just as Akai would wish!

Not long ago, *Sound On Sound* featured a series on sampling, called 'Forgotten Science — The Lost Art Of Sampling', which ran from August 2005 until February 2006. This may be of interest to you.

Do I need to build a room within a room?

I'm planning to build a shed studio with concrete walls, no windows and a drop-down ceiling, and I'm concerned that I won't have enough sound reduction from the outside. Will it be necessary to build a 'room within a room' to achieve decent soundproofing?
Joe Weiderhold

Editor In Chief Paul White replies: You won't need a room within a room, but it's a good idea to plan for a heavy ceiling with at least two layers of plasterboard and heavy rockwool insulation above. Suspended office ceilings don't do much for soundproofing, as they are fairly thin and have nothing above them. Also, the door would be better doubled up — one door opening outwards and the other inwards — separated by the thickness of the wall. Fit seals to the doors on all four sides, to ensure an airtight fit. The concrete shell can be lined with double plasterboard layers on battens with rockwool insulation. This will provide very adequate isolation. If you want to record drum kits, consider making a floating plinth of chipboard resting on 30mm-thick, high-density rockwool insulation slabs. You'll need acoustic treatment to stop the room sounding too live, but a few panels of 30mm or thicker rockwool slabs, covered in thin fabric and hung from the walls, ideally with an inch or two of air gap behind, should do the trick.

To maintain a healthy supply of oxygen, you'll need some ventilation, and probably air conditioning too, as it will get very hot during summer. It's also a good idea to install a dehumidifier, as this will keep your equipment condensation-free in the winter.

As you will know, we run monthly *Studio SOS* features where the *SOS* team visit readers' home studios and tackle some of their niggling issues. More often than not, these features offer advice on acoustic treatment. I'd also specifically recommend that you take a look at 'Making Rooms', from *SOS* July 2000 (www.soundonsound.com/sos/jul00/articles/faqacoustic.htm), and our five-part 'Practical Acoustics' series that ran in *SOS* from July to November of 1998.

What's the difference between these soundcards?

I need a new soundcard, as I'm upgrading to a 64-bit system, and I've been looking at M-Audio's Delta 1010 and Delta 1010LT. What's the difference between them and will they work with my new machine?
***SOS* Forum Post**

PC Music Specialist Martin Walker replies: Well, both are good products, but ▶

The Virus TI

Hardware Synth. Software Plug-In. Audio & MIDI Interface. Control Surface. And more!

The incredible new VIRUS Total Integration (TI) Synthesizer Line from Access is all of these and more! TI vastly expands the synth engine of the VIRUS, then extends it with VIRUSControl™, allowing it to appear as a soft-synth, perform as a USB Audio/MIDI interface, and perform as a universal midi control surface – truly a breakthrough product!! TI takes the legendary Virus sound to extremes, with more of what you loved, plus exciting innovations Dual-DSP, 80 note polyphony*, Wavetable and 9-layer HyperSaw OSC models, 16 delays, 16 reverbs, programmable arp (per patch!), tap tempo, and more. Memory is massive: 2048 ROM, 512 RAM, plus new self-contained multis, with 16 singles you can tweak and transform without affecting other banks. Besides a great synth, TI is a professional audio/MIDI control centre. VIRUSControl™ lets it appear as a softsynth (with sample accurate timing!) Add plugins to the sound, or use TI to process tracks with no CPU overhead. New +4dB balanced analog and SPDIF, backed by 24bit-192kHz converters, make it the perfect main I/C for your rig. TI even doubles as a 32-scene MIDI programmable control surface! The new VIRUS TI is a revolution, and with evolution ensured by free upgrades, this is just the beginning. Order today, and discover the one synth that ***can finally be all things to all musicians!***

VIRUS|TI PØLAR

VIRUS|TI

The Specifications:

- Dual DSP for 80 note polyphony*
- Wavetable, HyperSaw™ and original VIRUS oscillator models. 3 main oscillators p. voice (+sub)
- Parallel Reverb, Delay, Chorus, Phaser, Ring Mod, Distortion and Analog Boost FX per part
- 2 multi-mode filters: 4 original VIRUS, plus MiniMoog™-type self-oscillating models
- VIRUSControl™ VST/AU Plug-in links your synth to a Mac or PC to enable it to appear as a dual-stereo mix sample accurate soft synth with additional editor/librarian functionality
- Universal Controller mode with synth templates and 32 user memories
- 6 balanced studio-grade +4dB outputs with 192khz D/A, switchable soft clipping, and surround-sound capabilities. S/PDIF in/out. 2 24-bit inputs
- Tap tempo based on SyncXtreme™
- Freely programmable arpeggiator in every patch
- Knob Quantize syncs any changes to clock interval for step-sequence sounds
- 3 LFOs with 68 waveforms in loop or envelope modes and 2 ultra-fast envelopes
- Expandable Single Patch Memory: 2048 ROM (rewriteable), 512 RAM
- Two different 16-Part multitimbral modes: "Sequencer" mode (16 simultaneous loaded entries from single banks) plus 16 new multi-programs (with 16 embedded singles each). Both modes allow all effects on every part

* 80 note polyphony based on average patch load. Oscillator/filter model selection can result in slightly less or even more.

Available in the UK from

turnkey

020 7419 9999 www.turnkey.co.uk sales@turnkey.co.uk
114-116 Charing Cross Road London WC2H 0JR

SOUNDCONTROL

0870 067 1234 www.soundcontrol.co.uk info@soundcontrol.co.uk
Twenty Five Stores Nationwide

VIRUSControl™ VST/AU Plug-in

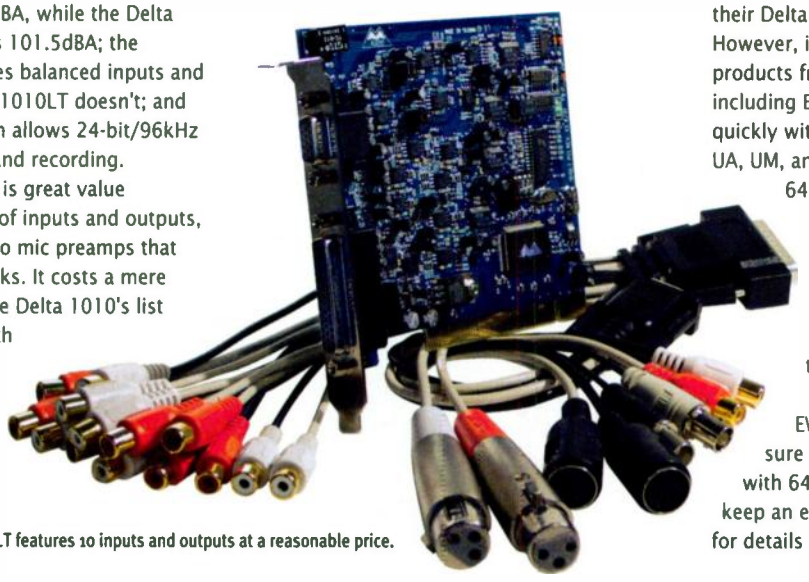
links your synth to a Mac or PC to enable it to appear as a dual-stereo mix sample accurate soft synth with additional editor/librarian functionality.

access

www.access-music.de

Q&A

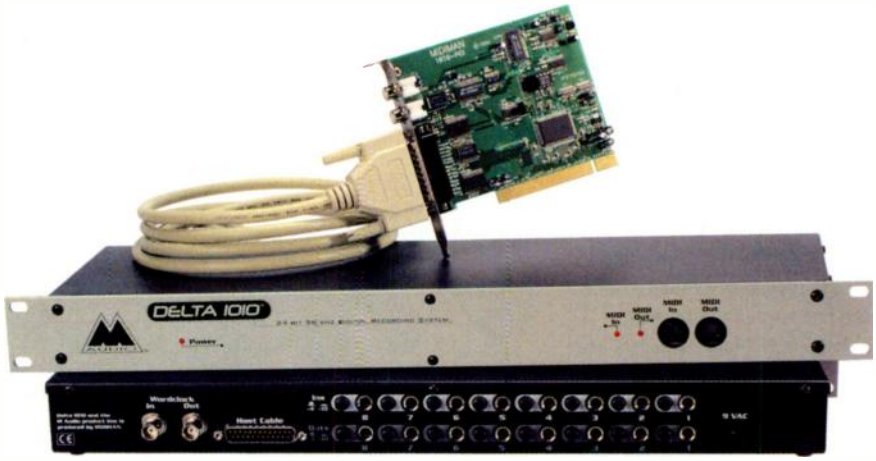
▶ although they provide a similar number of inputs and outputs, they are very different beasts in reality, and you can't expect quite the same audio quality from the budget LT version as from the original rackmount model. This is evident when comparing M-Audio's published specifications. The dynamic range of the standard Delta 1010's D-A converter is 117dBA, while the Delta 1010LT only manages 101.5dBA; the standard 1010 features balanced inputs and outputs, whereas the 1010LT doesn't; and the rackmount version allows 24-bit/96kHz full-duplex playback and recording. However, the 1010LT is great value considering its range of inputs and outputs, especially as it has two mic preamps that the standard 1010 lacks. It costs a mere £199, compared to the Delta 1010's list price of £399, but both products offer fantastic value for money. Remember that if you go for the LT you have to contend with the many cables that ooze from the back of the PCI card. Unless you have a patchbay, or are going to leave all connections in situ, re-plugging your soundcard will involve fumbling around behind your PC's tower. The standard 1010 has a rather more convenient breakout box.



The Delta 1010LT features 10 inputs and outputs at a reasonable price.

For more information on the Delta 1010, check out my review in *SOS* January 2000 (www.soundonsound.com/sos/jan00/articles/midiman1010.htm).

As for the 64-bit compatibility issues, you're in luck: M-Audio were the first manufacturer to announce 64-bit drivers for their Delta and Firewire interface series. However, it might be worth considering products from other manufacturers, including Edirol, who also stepped in fairly quickly with a raft of drivers for lots of their UA, UM, and PCR products. Others offering 64-bit support include Emu, for their DAS (Digital Audio System) interfaces and their XBoard USB/MIDI controller keyboard; Lynx, for their Lynx Two/L22/AES16 range; RME (only their Fireface 800 at present); and Terratec, for most of their EWS, EWX, DMX and Phase products. I'm sure there are others on the market with 64-bit compatibility, but you can keep an eye on manufacturers' web sites for details of new updates.



The Delta 1010 remains M-Audio's flagship PCI soundcard after six years of production.

Q Can guitars really kill people?

Years ago I touched a guitar that was plugged into an amp against a radiator, and it blew the amp and melted the guitar string. Afterwards someone told me that this was caused by a ground loop, but I've never actually understood what that means.
***SOS* Forum Post**

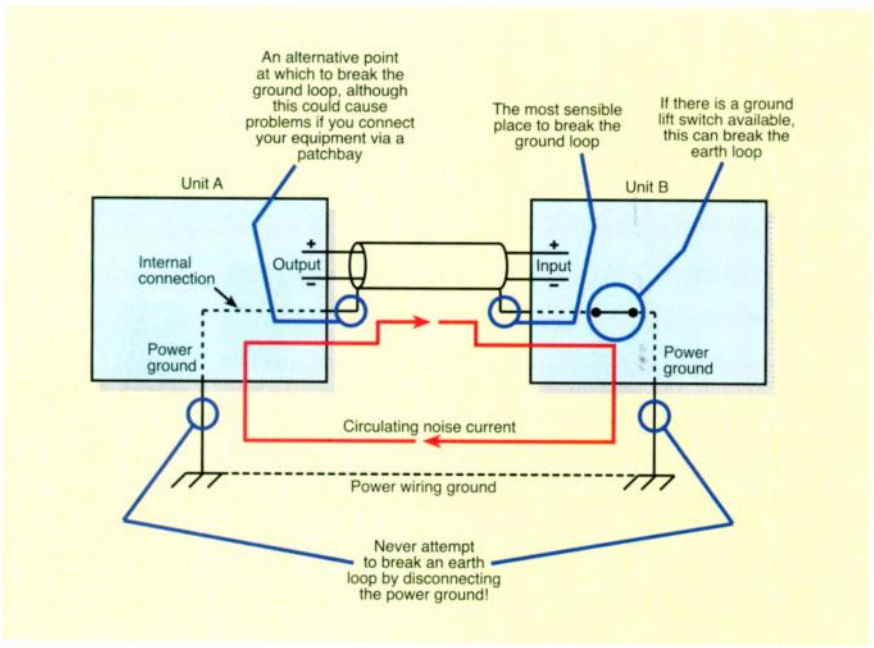
Technical Editor Hugh Robjohns replies: That wasn't a ground loop — that was a faulty amp with a missing safety earth. There was no loop, because there was no earth at all until touching the radiator provided the missing link. Had you sat on the radiator and then picked up the guitar, you might well have been playing Emaj7 on a harp on a fluffy white cloud by now!
This is a distressingly common and life-threatening situation often caused by guitarists (or their so called 'technicians') in a futile effort to stop audible hums.

A ground loop is different — it occurs when there is more than one ground path between two items of equipment. Usually, one path is the screen of an audio cable connecting the two pieces of equipment and the other path is via their chassis safety earths in the mains plugs. Inside the equipment, the audio screen earth is often linked directly to the chassis earth, hence the possibility of a loop. If the two bits of equipment are plugged into the same mains socket, their chassis safety earths are effectively tied together at the same potential, and so there is unlikely to be any circulating ground current, despite the apparent ground loop. However, if one item is plugged into a different mains socket, its chassis safety earth might be grounded some way away from the other equipment's earth, and there can be a small difference in potential voltage between them. Silly as it sounds, earth is not the same everywhere. The potential voltage difference between their two chassis earths can cause a small current to flow, and since the earth provides ▶

a reference for the audio electronics, that flowing current causes the earth reference voltage point to vary slightly. This can be heard, usually as a low-level hum or buzz. Ideally, the solution is to make sure that everything is earthed at one central point, so that everything shares the same common earth reference point. The easiest way to do that is to plug everything into a star arrangement of plug-boards fed from a single socket (assuming suitable power capacity). If that can't be done, the safest solution is to break the loop by isolating the audio cable screens at one end. The cable is still screened, but there is no longer any possibility of a loop, so the hum currents can't flow around it. Inserting transformers in the signal path can also break the loop, and this solution is common in outside broadcast and live sound rigs. DI boxes feature transformers for this purpose too.

Problems arise when uninformed people decide to break the loop by removing the safety earth in the mains plug instead. This does break the loop, obviously, so any related hum will disappear. But it also means that the equipment is no longer earthed, and thus any fault that occurs in the equipment is now life-threatening! Sometimes you don't even need a fault to cause dangerous problems, though. Most equipment has filtering on the mains input to stop mains-borne noise getting in (or out). If you disconnect the mains earth in the plug, the nature of that filtering is such that the (previously earthed) chassis — and everything connected to it — 'floats' up to half mains voltage (making it about 115V in the UK). This means everything that should be safely earthed — all the exposed metalwork, including guitar strings — now carries a life-threatening voltage. Going back to your guitar incident, the strings on the guitar are supposed to be earthed through the guitar lead to the socket on the amp. That, in turn, is usually connected to the amp's chassis earth, and thus through the mains plug to the mains safety earth. Metal radiators are connected to the mains safety earth point too, as is all house plumbing.

So if you have a guitar amp with the safety earth disconnected in the plug, the chassis is likely to rise to 115V, and everything that should be at 0V (earth) is now cooking with gas! Rest the guitar strings on anything that really is earthed (the radiator in your example, but a mic stand holding a mic that is earthed through its cable to a mixer is another very common alternative), and you have mains power now flowing directly through the equipment to find a real ground. This is almost certainly going to severely damage or destroy the amp, the guitar and — if you get yourself between the two — you as well! The classic way for budding pop stars to die is because of an amp with the mains safety earth disconnected. They rest one hand on their guitar strings and reach up with the other to hold a properly earthed mic. That leaves them with 115V effectively straight across their heart. Sweaty hands make very good conductors and it only takes a few milliamps of current flow to stop the heart. Musicians are electrocuted in this way every year, and while I'm all in favour of Darwinian evolution, sometimes the good guys get caught out too.



Lifting the mains earth like this could lead to serious injury.

The lesson is that you should never remove the earth from a mains plug. If you have hum problems, break the screens on the audio cables or use isolating transformers in DI boxes.

Q How do I arrange my monitors for surround sound?

Is there an easy way to measure speaker angles and distances for surround sound?
***SOS* Forum Post**

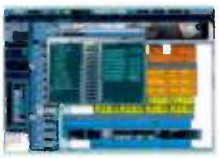
Technical Editor Hugh Robjohns replies: You will need some masking tape, a piece of string, a protractor and a helper (and maybe some Hobnobs). The first step, which I imagine you have already taken, is to decide on the listening position. Mark it with a cross on the floor using the tape, then decide on the centre axis and how far away the centre speaker should be. Mark their positions on the floor with tape as well. Next, stretch the string between the listening position and the centre speaker location and mark the string with a tape 'flag' so that the distance from the listening position to the centre speaker is known. Next, using the protractor, find the appropriate angles for the other speaker positions — typically 30 and 110 degrees around from the centre line in each direction, although some people working with surround sound prefer a more square arrangement, using angles of 45 and 135 degrees. With one end anchored at the listening position, rotate the string and, at the appropriate angles from the centre speaker axis, mark the floor at the distance from the listening position marked on the string. That ensures that all the speakers lie on the circumference of a circle, with the listening position at its centre. Finally, place the speakers on the marks, aim them directly at the listening position, connect everything up, and align the system for levels. Good luck! SOS

Why Carillon Recommends E-MU

Sound Quality
Outstanding audio

E-mu's 'm' series cards feature the very same 24bit/192kHz convertors as Digidesign's multi-thousand pound Pro Tools HD systems, providing pristine audio & an incredible 120dB dynamic range.

E-MU FX
Zero CPU VST FX



Powerful on-board DSP on all cards in combination with E-MU Power FX software gives you 16 simultaneous effects right inside your sequencer with absolutely zero hit on your CPU's performance.

0404 PCI card audio interface with 2 in 2 out analogue jacks offering 24bit/96kHz AD/DA conversion and AES/EBU compatible S/PDIF coaxial or optical I/O with 96kHz support.

1212m Built around the very same 24bit/192kHz convertors as Digidesign's multi-thousand pound ProTools HD system, the 1212m has balanced stereo analogue I/O on 1/4" jacks, MIDI I/O, S/PDIF I/O and ADAT I/O.

EmulatorX2
Legendary sampling

Emulator X2 software brings E-mu's years of sampling experience to the desktop in what is quite possibly the most powerful software sampler around, but with an interface that's a delight to use.

1616 PCMCIA CardBus with MicroDock breakout box, providing 16 I/O with 2 XLR 48v mic pres, 4 in 6 out analogue - S/PDIF & ADAT optical I/O. Comes bundled with a range of E-MU and popular sequencer / audio software.

Price
Nothing comes close

from only **£!!!!!**

Even at regular prices, this range of soundcards represents superb value for money, but Carillon's exclusive package deals with full boxed versions of Steinberg's Cubase make it absolutely unbeatable.

1616m All the features and specifications of the 1616, except this Mastering version uses Digidesign spec 24 bit 192kHz low jitter A/D and D/A converters.

Exclusive E-MU package deals - *only* from Carillon.

AC-1D

Dual Core 1
- Intel Dual Core 3.2GHz
- 1GB DDR2 RAM
- 80GB+320GB SATA RAID
- DVDR/RW

only **£1269**

Dual Core 2
- Intel Dual Core 3.4GHz
- 2GB DDR2 RAM
- 80GB+600GB SATA RAID
- DVDR/RW

only **£1899**

AC-1X

Xeon
- Intel Xeon 3.4GHz x2
- 2GB DDR RAM
- 80GB+800GB SATA RAID
- DVDR/RW

only **£2499**

Laptop
Steinberg Cubase SL & E-MU 1616 card

Centrino
- Intel Core Duo T2400 1.8GHz
- 100GB 7200rpm HD
- 1GB DDR2 RAM
- DVDR/RW

only **£1799**

BUY ANY CARILLON DESKTOP AUDIO PC...

Steinberg Cubase SE & E-MU 0404 card

only **£129!**

Steinberg Cubase SL & E-MU 1212m card, Emulator X software & E-MU Mophatt X

only **£299!**

Steinberg Cubase SL & E-MU 1616 card

only **£359!**

Steinberg Cubase SL & E-MU 1616m card & Emulator X2 Studio software

only **£539!**

Steinberg Cubase SX & E-MU 1616m card & Emulator X2 Studio software

only **£799!**

...AND GET ONE OF THESE E-MU/CUBASE PACKAGES AT AN INCREDIBLE LOW PRICE!

Carillon **020 7692 7629**

www.carillondirect.com
sales@carillondirect.com

Email your queries, comments and tips to: sos.feedback@soundonsound.com
Or post to: Crosstalk, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge, CB3 8SQ, UK.
Visit the *SOS* Forums via www.soundonsound.com

Big Rig

At the end of Dave Stewart's East West/Quantum Leap *Symphonic Orchestra* article in your June 2006 issue (www.soundonsound.com/sos/jun06/articles/ewsymphonicupdate.htm), there's information on David Newman's setup. I'd like to know how you install and network a number of PCs with the various libraries on them and connect to a Mac running *Logic*. I'm planning on upgrading and would like to get this information before deciding what to do.
Matt Huber

Composer David Newman uses a network of six PCs to produce his demos.

***SOS* contributor Dave Stewart replies:**
To ease the CPU, RAM and logistical strain caused by modern large-scale orchestral sample libraries, many composers (generally the well-paid kind) run their libraries on multiple computers. PCs have traditionally been preferred to Macs for the job because Tascam's popular *Gigastudio* sampler runs only on PC. Also, PCs are cheaper! Connecting a Mac running *Logic* to a number of PCs is no different, in principle, to sequencing multiple hardware samplers: the sequencer (*Logic*) outputs MIDI data, and a hardware MIDI interface connected to the Mac distributes the data to the PCs, each of which requires its own MIDI interface to receive it. The PCs obviously need soundcards too — you can kill two birds with one stone by using a soundcard with a built-in MIDI interface. In such a system, the PCs operate independently and don't require networking.

It's worth mentioning that MIDI data can now be transmitted and received by multiple computers over a wireless LAN network, which could, in theory, save the expense of buying hardware MIDI interfaces and cables. Although this kind of networking may eventually become the norm, I don't know of any composers who do it at the moment!

If you're planning a multi-computer setup, it's best to start simply and add machines when required. This allows the system to grow organically at a sensible pace determined by your musical needs. For a simple orchestral arrangement of strings, brass and percussion, using one or two playing styles per section or instrument (say, sustains or staccatos for the strings and brass and straight hits or rolls for the percussion), you could probably install and run all the samples on one computer. It would have to have enough RAM (you need a minimum of 2GB for this kind of work) and your software sampler would need to support at least 32 MIDI channels. Most sound libraries run on both Mac and Windows, so anyone new to the field could install the samples on a second drive on the Mac running *Logic* and see how it copes with the workload before splashing out on a second machine.

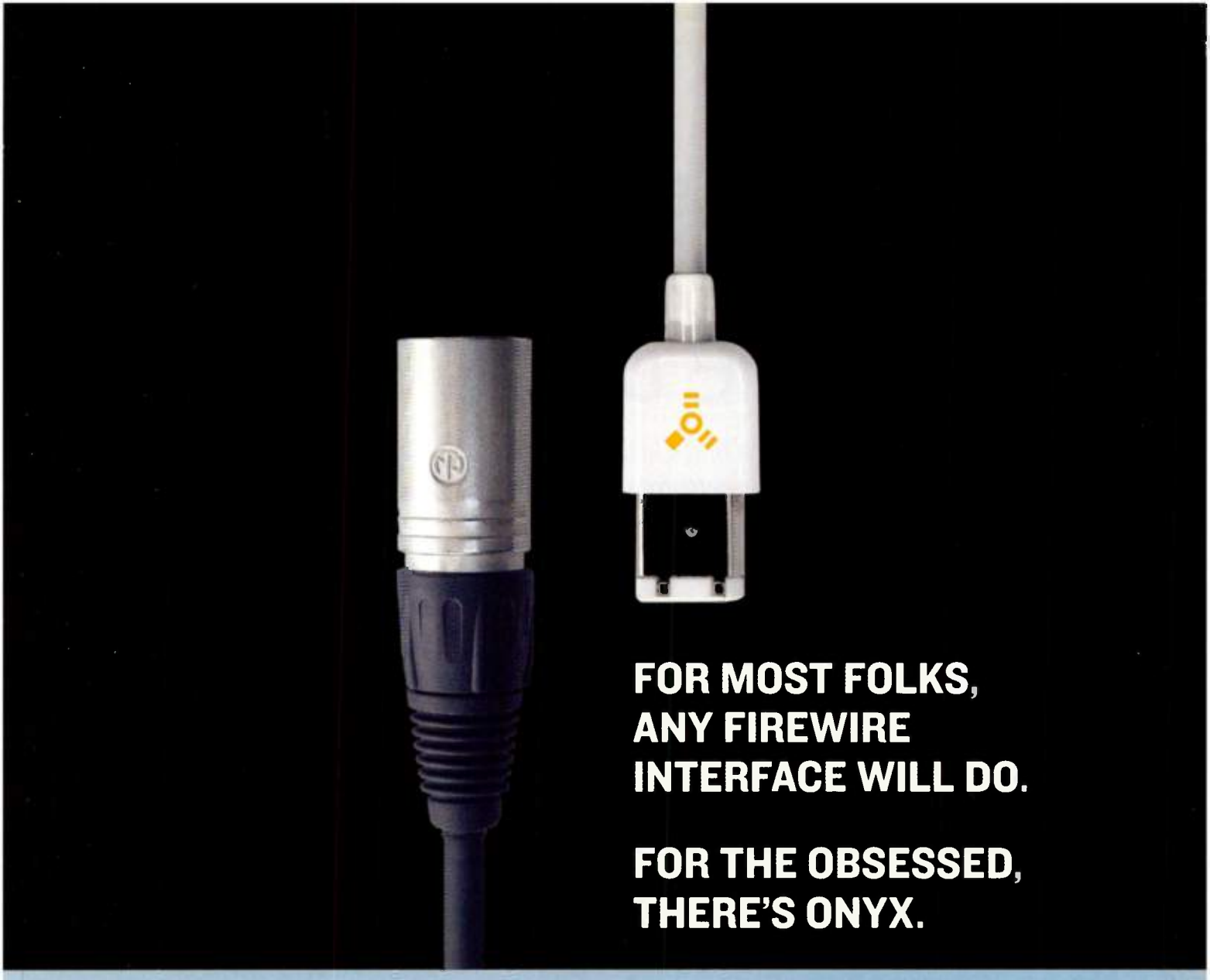
As the instrumentation expands and musical detail increases, you may reach a point where buying a second computer is inevitable. Strings use a lot of different articulations that tend to gobble up RAM, so you could install all your string samples on one PC and group brass, woodwind and percussion together on a second. However, if your writing for brass also begins to demand more instruments, sections and fancy, RAM-hungry playing styles, you could segregate your brass on the second PC and buy a third for your woodwind instruments, percussion, harp, piano, harpsichord, organ, choir — and so on.

The traditional way of connecting the computers is to use MIDI cables. One MIDI cable supports 16 MIDI channels, so if you were to connect an eight-way MIDI interface (such as Emagic's Unitor 8) to the Mac, you'd have 128 MIDI channels available for your arrangement. You could run MIDI for four PCs from one Unitor 8, assigning two of its MIDI outputs to each, to provide 32 independent MIDI tracks on each PC. In this setup, each PC would require a simple two-way MIDI interface, but to allow for expansion you might want to consider buying four-way or eight-way interfaces for one or more of the PCs. If you need more than 128 MIDI tracks, you can also connect multiple MIDI interfaces to the Mac. It's possible to keep the MIDI track count down by using keyswitch programs (which are now provided in most orchestral libraries). These enable multiple articulations to be accessed on the same MIDI channel.

The machine room at film composer Hans Zimmer's studio has 10 PCs running Tascam's *Gigastudio*.

To refer back to your question, composer David Newman uses a total of six PCs — two for strings, one for brass, one for woodwinds, one for piano and percussion and one for miscellaneous sounds — in a wired MIDI network, using four Unitor 8s on the Mac and an RME HDSP 9652 Hammerfall PCI soundcard (which has two MIDI Ins and Outs) on each PC. UK composer Andrew Blaney spreads his Vienna Symphonic libraries across three Macs and two PCs, using an eight-way MOTU MIDI Timepiece AV and three five-way MOTU Micro Lite MIDI interfaces on his main Mac (which runs MOTU's *Digital Performer* sequencer), and a combination of Micro Lites, RME HDSP 9652 soundcards and a two-way MOTU Fast Lane on his slave machines.

These are fairly large systems, but don't despair — as suggested above, you can run a scaled-down sampled orchestra on one machine and use it as a template for your arrangements. Concentrate on the musical ideas and instrumentation and don't worry too much about elaborate playing styles such as trills, rolls, grace notes and runs. You can fake these using the basic samples, before substituting samples of the real thing later, on an instrument-by-instrument basis, recording each style as audio as you go. It takes longer and requires more planning, but the end result should be none the worse for that. SOS



FOR MOST FOLKS, ANY FIREWIRE INTERFACE WILL DO.

FOR THE OBSESSED, THERE'S ONYX.

Ordinary FireWire audio interfaces are fine for capturing your musical ideas on the go. But if you're the type of musician or engineer who won't compromise quality, then you need to audition the Onyx 400F Studio Recording Preamp with 192kHz FireWire I/O.

This professional 10-channel premium mic preamp and audio interface features four boutique-quality Onyx mic preamps, with superior headroom, sonic detail and clarity vs. the competition (123dB dynamic range and .0007% THD, measured in the real world). The Onyx 400F also offers TRS inserts for plugging your favorite outboard gear into your signal path before sending it to your Mac or PC. And an internal 10 x 10 DSP Matrix Mixer with 64-bit floating point processing and full recall—a feature not found on any other FireWire interface, at any price.

With mastering-grade 24-bit/192kHz AKM® audio converters, true 192kHz operation at full channel count, a powerful standalone operation mode, and robust aluminum-and-steel construction, the Onyx 400F boasts fanatical attention to every last detail. Not to mention exceptionally open, natural and revealing sound worthy of your finest projects. Visit www.mackie.com/onyx400f to feed your obsession.

MACKIE ONYX 400F: FOUR BOUTIQUE MIC PREAMPS W/ 192KHZ FIREWIRE I/O. PLUG IN TO ONYX.

www.mackie.com ☎ +44 (0) 1268 571212 ☎ 425 487 4333 (Outside U.S.)

MACKIE.

WRH

Roger Nichols: Across The Board

Mixing music might seem like a black art, but here's a simple approach to help the newcomer get their tracks together.

Roger Nichols

This month's column leans toward the new 'audio recording enthusiast' who is trying to figure out how it all works. I have been there, although it was back when stereo was a new gimmick.

After you unpack the boxes, connect all the gear, look at the covers of the manuals (nobody has ever actually read a manual, right?) and try to start recording and mixing, how do you figure it all out? Like learning to play a guitar, learning to paint, or tuning the fuel injection on a Ferrari, first you need to be pointed in the right direction, and then you need experience.

There are hundreds of books to point you in the right direction. There at least 10 books covering each major DAW, including Pro Tools, *Sonar*, *Cubase/Nuendo*, *Digital Performer*, Adobe *Audition* and *Logic*. There are at least that many writings about recording and mixing in general, and a few books try to teach mastering. For some people, however, reading the books doesn't help much more than sleeping with the book under your pillow. Along with reading, you need to have some practical experience.

As far as equipment goes, you can learn about recording and mixing with what you already have. It doesn't take much of a computer these days to get the ball rolling. If you can't afford the software, there are dozens of free multitrack recording programs available, as well as demo versions of most of the high-end programs. You can actually start mixing before you start recording. There are multitrack versions of songs from Nine Inch Nails and other groups on-line for download. There are some books that include multitrack material to mix from Waves, and Charles Dye's *Mix It Like A Record* DVD.

It Goes To 11

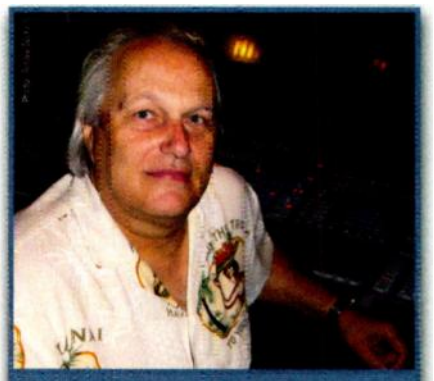
The biggest hump to overcome when mixing is that everything sounds better when it is louder. Everything! You can have the crappiest mix on earth, but if you play it loud enough, it will sound great. The worse the mix, the louder you have to play it.

You see this all the time in the studio when the mix is almost done and the band shows up for playback. The guitar player comes over and wants the guitar up just a little. Then the bass player sneaks over and turns up the bass. The drummer is not far behind, complaining about the lack of punch in the drums, and up crawls the drum group. After everyone finishes, the mix is right back where it started, but louder. The band is happy, approves the mix and leaves. The engineer turns down the master fader to get the overall level back to normal, and prints the mix.

If you can change the level of an instrument in the mix by a tenth or two-tenths of a dB and you can hear the change that you made, the mix is getting pretty good. If you can change some other instrument by that amount and not really hear any difference, then chances are that the instrument you are playing with is not where it should be. The problem may be down to level, or EQ, or compression, or reverb, but it is not quite ready for prime time. I will discuss all four and point out things to watch for.

EQ Balance

Think of any instrument as two separate components: the low end of the instrument and the high end of the instrument. As an example, let's take an acoustic guitar. The highs are important because they give the attack of the note or the raking of the strings that extra presence to cut through the track. The low end creates the body of the sound, the warmth of the instrument in the track. If the balance between these two halves is not right, then you will never get it ▶



Roger Nichols has been professionally involved in the music business since 1968, working as a staff recording/mixing engineer at ABC Records and Warner Bros before becoming an independent engineer/producer in 1978. His work with Steely Dan in particular has led to a string of Grammy Awards and nominations, including a Best Engineered Album award for *Two Against Nature*. An advocate of digital recording since 1977, Roger designed and built the first digital audio percussion replacement device and has lectured on digital audio around the world.

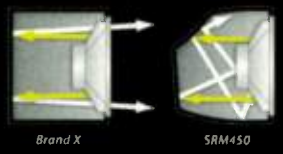
Multitrack iTunes?

Major record labels are gearing up for download of your favorite multitrack material. The record labels are actually releasing these multitracks to copyright the content. A copyright is not valid in the US unless the material is distributed publicly. The multitracks in the vaults are, therefore, not copyright material. The melody and the lyrics carry a copyright, but that is for the writers and publishers. The record company has a copyright (P) on the physical CD, cassette, or vinyl album that is shipped, but that only covers the final two-track mixes as they occur on the finished product. No protection is offered toward the multitrack recordings. Interesting. So, if they offer the multitracks for download, and leave off the lead vocal or melody line, they don't have to pay anyone anything. Artist gets nothing, producer gets nothing, songwriter gets nothing... Hmmm. Anyway, there is going to be lots of material to test your mixing skills, for a few dollars more.

UK laws are slightly different, though I doubt that record companies ever send copies of the multitracks for registration with UKCS. UK law is better than the US in that all you need to do is produce the work for it to be copyright; the root of the US record company stance is that you cannot copyright chord progressions or tempo, so without the melody and lyrics, there is nothing unique about the work. An example would be Tina Turner's 'Private Dancer', written by Mark Knopfler. When you heard it, you thought it was Steely Dan's 'FM' track. The bass line, the rhythm, the tracks seem to be uncannily similar to 'FM', but the melody and words are different, so no case.

SOUND BETTER. PLAY LOUDER.

THE SRM450: LEGENDARY SOUND AND QUALITY IN A COMPACT PACKAGE.



The SRM450's rugged molded cabinet is asymmetrical to trap internal resonances.

The SRM450's titanium high-frequency compression driver is loaded onto an exponential horn with multi-cell aperture throat. This ensures that everyone in the audience hears the same thing, even if they're standing on the sides or in the back.

Inside: Dual FR Series high-current power amplifiers deliver 300 watts LF and 100 watts HF actual RMS power to the transducers.

There's a built-in metal handle on top. Side handles are rubber/plastic, and are placed at the SRM450's center of gravity for balance when carrying.

Inside: Sophisticated parametric EQ, phase alignment and time correction circuitry give the SRM450 the most accurate sound of any molded active loudspeaker in its class.

The SRM450's custom high-output low-frequency transducer is servo-controlled for tight, accurate bass down to 55Hz.

The SRM450's superb sound quality and precise pattern control make it an ideal stage monitor.

There's no substitute for hours and hours of practice, but the SRM450 2-Way Active Loudspeakers can sure improve your sound once you finally hit the stage.

Tighter bass, cleaner vocals, a better defined mix. And none of that screechy "ouch factor" that haunts so many other active and passive loudspeakers on the market.

Need some convincing? Just visit your authorized Mackie dealer and do a side-by-side listening test with Brands X, Y and J. Then visit the store's Service Department and see how many SRM450s are *not* in for repair. That should say it all.

Want to sound better and play louder? Plug into a Mackie SRM450 and see exactly what our legendary active loudspeakers can do for your music.

MACKIE SRM450: SUPERB SOUND, HIGH OUTPUT, AND BUILT TO TRAVEL. PLUG IN TO ACTIVE SOUND.

www.mackie.com ☎ +44 (0) 1268 571212

MACKIE.

to sit in the track correctly. There will always be places where it seems too loud or too soft.

Start with the attack of the notes, or the raking of the strings in the case of a rhythm acoustic part. Change the level of the acoustic guitar until the high end sounds right in the existing mix. Get the level to where you can hear a level change of two-tenths of a dB. Now stop. Start with a parametric EQ set to a low Q of around 2, giving a very wide, smooth curve, and the frequency set for around 250Hz. Move the gain of the EQ up or down to get the low part of the guitar into the right relationship in the mix. Try to think of the low end of the guitar as a separate instrument. Get it to sound good in the mix. When you get the low end of the guitar in a place where you can hear a two-tenths of a dB change in the level, you are doing great.

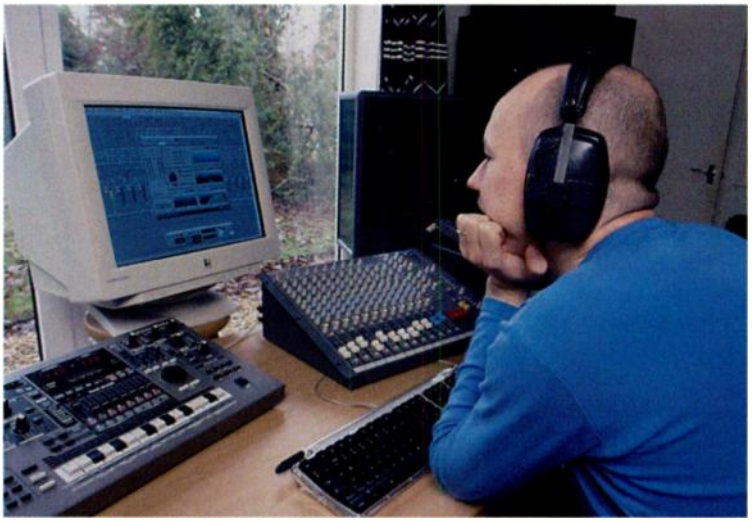
Now you are going to do one more thing. It is usually the low end of all the instruments adding up that makes the mix sound muddy. My rule is, if you can't hear it, you don't need it. Turn on the lowest band of the EQ and make it a high-pass filter, or use the high-pass filter if it operates independently. Start with the frequency all the way down. Leave all of the tracks playing, with the acoustic guitar in the mix. Start raising the frequency of the high-pass filter slowly until you can hear a change in the sound of the low end of the acoustic guitar. Look at the frequency on the high-pass filter. Now reduce the frequency by about 15 percent. That means if the frequency reads 200Hz, then move it down 30Hz to about 170Hz. If you do this to each of the mid-range instruments, it will clean up the muddiness problem in your mix.

Compression Doesn't Always Suck

If some notes just stick out or get lost and EQ by itself doesn't help, then maybe you should use a little compression. Notice the key word: a *little* compression. Of course the best way to set a compressor is to listen to the results, make an intelligent judgment, and twist all the knobs until it sounds right. Or, you can start with an easy rule of thumb that can get you most of the way there.

Set the ratio to 2:1. Depending on the compressor, use either the input control or threshold control while the music is playing, and lower the threshold (or raise the input gain) until you see the gain-reduction meter jump into action. Keep adjusting until the gain reduction peaks at -3dB to -4dB. For now we will just leave the attack and release values at whatever the defaults were when you loaded the compressor.

This is now a pretty good starting point for compression. It is not even going to really sound compressed. You now have a little more control over the dynamic range of the instrument without hurting the sound of the instrument. Now readjust the level of the instrument to get it settled in the track. When you can start hearing small changes in level, you are in the right place.



If you're new to it, mixing a multitrack project can seem overwhelming.

Sometimes EQ, sometimes compression, and sometimes a combination of the two may be necessary to get the desired results. Try the EQ first, as this will maintain the greatest amount of dynamic range in the instrument. If you can't quite get it, try a little compression.

On The Level

We already have a pretty good mix going, but how do you know if it is the best possible mix? Think of it this way. It is always easier to tell when something is wrong than when something is right. This is true of instrument levels in a mix. While listening to the mix, one at a time change the levels of each instrument in varying combinations. Turn the bass up 1dB. Does the mix sound worse? If the answer is yes, then turn it back down. Turn the bass down 1dB. Does this sound worse? If the answer is yes, then the original level was right for the bass. Now try one of the guitars. Then the piano. If an instrument sounded better at a new level then leave it there and start back through the cycle.

This can be a time-consuming task, and you should take lots of notes so you can get back to previous settings, but by the time you get through the process and have eliminated everything that sounds worse, then you have the final mix. Cool, huh?

Hello... Hello... Hello...

Reverb is the last thing to add, because it depends on all of the other levels being close to the final ones. Normally, when you add reverb, your initial impression is that it sounds more three-dimensional and lush. This is true in a way, but it also starts sounding like the whole band is playing with you in the shower. I don't mean playing with you, I mean playing along with you on their instruments. I mean... Never mind.

Solo each instrument that has reverb added, along with the reverb. Turn the reverb down until it no longer sounds like a big reverb, but more like real-world ambience. 'Yup, sounds like the acoustic guitar was recorded in a nice-sounding room,' should be the goal. There is a place for special effects if you want, but make sure there is room in the track by backing off on the other reverbs.

One more reverb trick I always use, both with reverb plug-ins and external reverb hardware, is to insert a one-band EQ after the reverb return. Set the EQ to high-pass filter and set the frequency to 250Hz. Reverbs usually generate some of their own low-frequency artifacts that are not part of the music. Also, low-end reverb just makes the mix rumble. Limiting this unnecessary sound makes the reverb accentuate the music in the middle of the spectrum, and the nice splash on the snare and the vocal, without adding mud.

Your Work Is Never Done

This is by no means a complete course on mixing, or the definitive answer to all of your troublesome tracks. This is meant to help someone new head in the right direction. Use these tips and expand on them. See what works for you.

If you do things differently and make great-sounding records, then perfect. I want to know how you do it. Share your methods with others who are trying to figure it out. I like buying CDs and listening to new bands. What I do not like is bad-sounding CDs. Put together someone who knows how to make great-sounding records with a band who know how to write and play great songs and you've got something! SOS

Discover Dreamtek and buy online by visiting www.dreamtek.tv

DREAMTEK

020 8533 2222

What makes Dreamtek different...

- Apple Solution Expert for Audio and Video
- Specialising in system configurations
- Expert trainers on hand for all your training needs
- Business leasing and educational pricing available
- Superb service and advice for all your audio requirements

"If you want to talk to the real "experts", in our industry... start with Dreamtek"

Johnny Burke - Iron Maiden

STOP DREAMING, GET TRAINING...

Do you want the kind of "in between the line" advice on tools and techniques to get the best from your equipment?

At Dreamtek we have a dedicated team of experts on hand for your every training need from Logic Pro / Pro Tools/ Digital Performer / Recording and Mixing techniques and many more.

Contact us about training at our East London training facility on 020 8533 2222 or email training@dreamtek.tv

Please call Dreamtek for pricing on PowerMac, Powerbook & iMac Audio Systems

Apple Powermac G5

PowerMac G5 Dual 2Ghz from £1287 inc VAT

PowerMac G5 Dual 2.3 from £1608 inc VAT

PowerMac G5 Quad 2.5 from £2115 inc VAT

The NEW Apple Powerbook with Dual-Core Intel Engine

MacBook Pro 17" Intel Dual-Core 2.16Ghz from £1747 inc VAT

MacBook 15.4" Intel-Core 2.16Ghz from £Call

MacBook 15.4" Intel-Core 2Ghz from £Call

Logic Pro Workstation Bundle

PowerMac G5 2.5GHz Quad/250GB/SD
+ 2GB Ram Total
+ 250GB Audio Drive
+ Logic Pro 7
+ RME Fireface 800 Audio & Midi Interface

From £3899 inc VAT

NEW...

MOTU Microlite £495 inc VAT

Focusrite Liquid Mix £Call

Apogee Ensemble £1404 inc vat

What's recommended...

MOTU Traveler

£Call

Focusrite Liquid Channel

£2149 inc VAT

Digital Performer £399 inc VAT

Spectraonics Sylus RMX £168 inc VAT

Logic Pro from £428 inc VAT

RME Fireface 800 £899 inc VAT

Focusrite Saffire SL NEW £Call

Digidesign M-Box 2 from £319 inc VAT

Genelec 8040a £509 inc VAT

Universal Audio UAD1 £898 inc VAT

Ableton Live 5 from £179 inc VAT

Discover Dreamtek and buy online by visiting www.dreamtek.tv

Emu Emulator X2

Software Sampler For Windows

Not content with adding a huge raft of improvements to their already comprehensive *Emulator X* soft sampler, Emu have also removed the need to use it with their own soundcards, so now everyone can get in on the fun.

Martin Walker

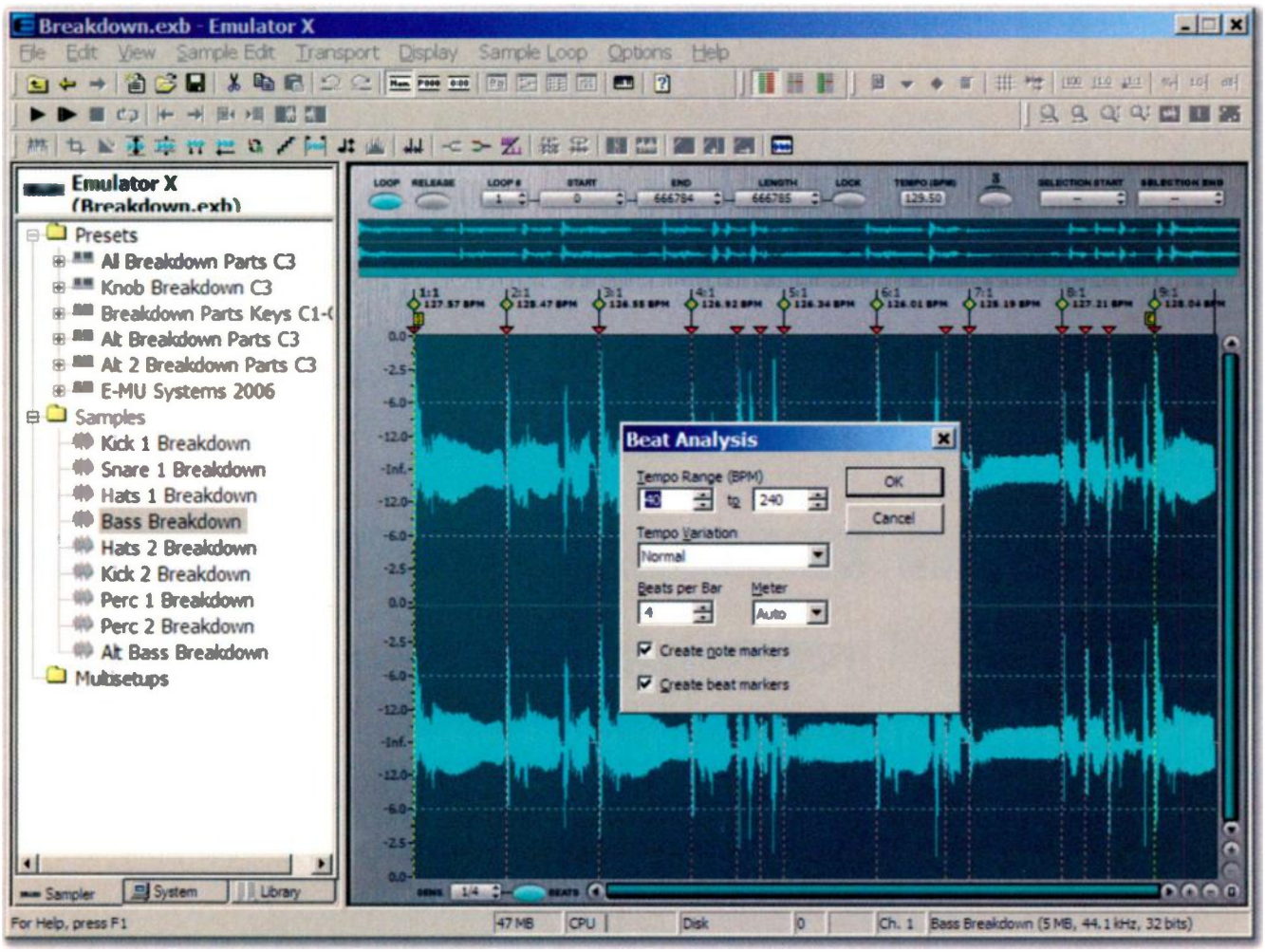
Emu's *Emulator X* software sampler caused a huge stir when it was originally released in early 2004. Running either as a stand-alone program or as a VST Instrument, it shipped with the Emu's 1820M audio interface, acclaimed for its breakthrough audio quality, and featured an incredibly comprehensive synthesis engine including more than 50 Z-Plane filters previously only found in Emu's hardware samplers, extensive 'patch cord' routing options, multi-wave LFOs, an integrated waveform editor and a range of DSP tools for sample manipulation. Later that year saw the release of the cheaper *Proteus X*, which omitted the user-sampling functions but was otherwise very similar, at an even lower price. Now, *Emulator X2* and *Proteus X2* take things to a new level of sophistication.

Emulator X2 can be bought in three versions: as an upgrade bundled with a single new *Xperience* library CD-ROM for existing *Emulator X* or *Proteus X* owners; as an add-on with the complete 3GB four-CD sample library to any of Emu's current audio interfaces or controller keyboards; or as a full retail product with the 3GB library and the Xmidi 2X2 USB MIDI interface that effectively acts as a dongle. You can read more about the library CD-ROMs in the box later in this article.

The add-on and retail versions are a big departure for Emu, since previously you could only use the *Emulator X* software if you had one of Emu's PCI soundcards; *X2* will also be of interest to those who want a capable soft sampler but who already have a perfectly good audio interface.

Installation

Out of interest I left my existing *Emulator X* installed to see what happened, and was very



Chopping audio files into beats and bars takes only a few seconds with the easy-to-use Beat Analysis function.

pleased to see that the *Emulator X2* setup detected this and offered to uninstall it prior to replacing it with the newer version, so there were no conflicts. *X2* has a slightly more restrained appearance, with darker 'brushed aluminium' panels and fewer shiny metal highlights, and I felt this looked more sophisticated and was easier to read.

There are also some fundamental changes to the sampling engine, with the stand-alone *Emulator X2* now supporting 64 instead of the previous 32 MIDI channels; the VST Instrument version remains limited to 16 MIDI channels, although of course you can load multiple instances in your DAW. There's also a new External Edit option so you can work on samples in your favourite editor, while the Voices & Zones page has been heavily reorganised to accommodate controls for the new features.

Chopping & Changing

For many musicians, the highlight of *X2*'s new features will be Twistaloop. This can chop audio grooves into their constituent beats, create new loops and regions from them, play them back at any tempo without changing pitch (including locking to external song tempos) using audio time compression/expansion, and even jump between multiple loops in real time using MIDI controllers. At the heart of Twistaloop is a new beat-detection engine. You just choose the sample you want to work on, click on the Beat Analysis function of the Sample Edit window, and a small dialogue appears offering various tempo and timing choices. The default settings will be fine with most material, but it does help to restrict the engine to the approximate tempo range of your loop, and to indicate how many beats there are in each bar. The further options can help if your loop has sudden changes of tempo rather than gently wandering ones, or is a short sample with a rigid tempo.

Once analysed, your loop appears in the Sample window with its bars and beats highlighted by markers, and adjusting the Sensitivity setting lets you determine how many 'notes' there will be — a low sensitivity will mark the loop as several longer sections, for instance, while higher settings let you chop the loop as far as individual hi-hat hits of a complex drum groove. Sensitivity can either be set as a percentage below peak level, or switched to Beat intervals ranging from eight bars to 16th notes. Triplets and duplets are also catered for, and if necessary you can fine-tune both beat and note markers by clicking and dragging.

Now comes the fun bit: with Snap To Marker enabled, you can quickly and easily highlight any section of your audio defined by the note markers and declare it a loop by using the Make Loop From Selection command. You can do this as many times as you like with different audio sections from this one sample, and when you've finished, move to the preset's Voice Processing page and access these multiple loop selections for playback.

Using the new 'Loop (n)' rotary control you can choose which of your selections to play, and depending on the setting of the associated Start At Loop button, you can either play all of the sample until your looped section begins and then starts to loop, or just play that loop section. You can also route MIDI velocity, aftertouch, or other external MIDI controllers to the Loop Select function, so you can change it in real time during a performance or automate it from your sequencer.

Once your audio has been chopped into beats and you've selected multiple loops from them you can play different ones back in real time at different tempos using the Twistaloop and Loop (n) controls.

You can even use the Loop functions with multisampled voices, where the various samples used may contain different numbers of loops. In this scenario the control changes to 'Loop (%)' and, for example, a sample with two loops would change over at the 50 percent setting while another with three would switch at 33 percent and 66 percent settings. Another new feature lets you take the various looped sections of a single sample and map them to different keys in the Voices and Zones window — this would, for instance, let you create a multisampled instrument from a single long recording containing all its separate notes, or play back different sections of a drum loop together using different keys.

With the rotary Twistaloop control of the Voice Processing page set to 'Off' your loop will play back at its original tempo, but once on, you can set it anywhere between half and double speed, while the pitch stays constant. There are some artifacts at slow speeds, but they are not worse than with many other such functions, and I doubt that groove-manglers will mind them at all anyway.

Each preset can have its Twistaloop loops locked to the global master tempo, while the Twistaloop Override settings for the entire preset bank let you globally turn this on or off or abide by the individual preset settings. When Off the various drum loops in a particular preset may all originate at different tempos, but once analysed by the beat-detection engine the same selection of loops will now sync perfectly together, and you can change the tempo in real time to follow your song.

The well-written PDF manual provides copious examples for all the new features to explain all the subtleties, but I found it a doddle to import samples, chop them up into sections, and then assemble them into performance presets. The *Xperience* CD-ROM also contains plenty of well-designed examples to show off these new features, including one-note jam sessions where you can pick and choose which combinations of instruments play using the mod wheel: improvising jazz drummers, velocity-switched wah-wah guitar riffs, presets that have different loop lengths in each stereo channel for ever-evolving riffs, and monster presets containing dozens of synchronised riffs that you can drop in and out at will.

Multi-function Generator

The upgraded *X2* engine provides three new Multi-function Generators for each voice, and I suspect most musicians will drool over their capabilities. Each provides between one and 64 steps, and you get at them on the new FN Gen tabs on the Voice processing page. Operation is almost self-explanatory — there's a graphic window showing your programmed ▶

Sound On Sound

Emu *Emulator X2*

pros

- Anyone into loops will love Twistaloop's ability to chop multiple audio files into segments and sync them all together in real time with varying tempo.
- Sound designers will drool over the new Multi-function Generators, built-in convolution, and Morph Filter Designer.
- *Emulator X2* no longer requires an Emu PCI soundcard to function.
- Synth Swipe is a great help to those moving from hardware synths to software sampling.

cons

- Such a comprehensive sampling engine can easily consume lots of CPU unless you're careful.
- The feature set may seem overwhelming to new users.

summary

Emu's *Emulator X* was already one of the most musical soft samplers available for the PC musician, but the *X2* version takes it to new heights of creativity!

There are three new Multi-function Generators so you can perfect 64-step sequences, arpeggiators and complex envelopes, while the Morph Designer Filter lets you create your own filter responses with up to six stages.

steps, and you can zoom into shorter sequences. The easiest way to enter new sequences is to draw them in with the mouse, but you can switch in a handy selection of vertical 'grids' to restrict values to semitones, octaves, major and minor triads or scales over a ±32 semitone range. For finer control you can instead select individual steps and input the value using inc/dec buttons or text entry, and there's also a 'Smooth' function that glides between each step.

The step rate can be specified over a very wide frequency range from one cycle every 12.5 seconds to 18 per second, or switched to BPM, where the options change to a huge selection of tempo rates, from eight whole notes to 32nd notes (including triplet settings), derived from the master tempo setting. You can sync the sequence so that each new key-press starts a new one at the beginning, so each new note starts at a random step in the sequence, or let the first note played start the sequence and then all subsequent notes follow the same pattern in exact channel sync. Apart from the usual choices of forward, reverse, forward/back and one-shot sequence directions, there are also random and Brownian (random but always one step from the previous position) options.

There's also an optional Gate signal for each step so you can trigger other events at any point during the sequence, and to further spice up the proceedings, various Multi-function Generator parameters appear as modulation destinations, enabling you to change their rate, direction and length, or even re-trigger them from the beginning, via other parameters or external MIDI controllers, all in real time. Luxury indeed!

The most obvious application for these Multi-function Generators is to design 64-step sequencers or arpeggiators by patching them to note pitch, but as we've come to expect from Emu, there's a vast array of other options, such as creating complex multi-step envelopes for the filter or amplifier sections, letting the timing of one Multi-function Generator be controlled from another for multi-speed sequences, and so on.

Mighty Morphing Filter Designers

Talking of filters, Emu have already generously provided more filter responses (55) than the average user can shake a stick at, but for the true explorers of synthesis the new Morph Filter Designer lets you at the bits that have previously been denied to all but Emu sound designers. Accessed via the normal filter menu, the Morph Designer lets you combine anywhere between one and six filter sections in series, each with a choice of low-pass, high-pass, or EQ (peak or dip) responses.

For each stage you enable (and watch out for your CPU overhead, because of course they all require more processing) you get Frequency and Q controls for both the lowest and highest setting adjusted by the Morph wheel, so you can move smoothly between these extreme settings and hear the effect of the filter sections in combination. Given that the Morph value is also a modulation destination and can therefore be controlled by MIDI velocity, envelope level, LFO, or of course the new Multi-function Generators, this provides you with huge creative potential that's streets ahead of most other sampling products, whether software or hardware.

Transform Multiply

Transform Multiply is a new tool added to the already comprehensive roster of Sample Edit Tools (apart from the usual trim, fade, normalise, change gain, reverse and channel swap options, this already includes Bit Reduction, Time Compression, Pitch-shift and

Emulator/Proteus X Libraries

Emulator X2 is bundled with a four-CD-ROM sample library almost identical to that shipped with the original *Emulator X*, containing the excellent 1000 X-Producer presets from Emu's Proteus 2000 hardware synth module, the Hip-hop Producer pack, Saint Thomas strings, a 20MB GM soundset, the Studio Grand piano, and the *Beat Shop 1* set of kits, loops and grooves, plus a new CD-ROM labelled *Twistaloop Xperience*, which showcases all the new *X2* features.

Within certain limits you can load or import lots of sound formats into both versions of the *Emulator X* sampler; supported formats include EOS, EIII, *Gigastudio*, Akai, *Halion*, *EXS24*, Soundfont 2.1 and WAV, but anyone considering buying a software sampler will also be interested in what other libraries are available for it. Emu have now released quite a few sample libraries specifically for *Emulator* and *Proteus X* owners — the first six volumes all contain sounds based on Emu's well-respected hardware synth module range, but more recent ones provide a huge range of new sounds covering a wide range of genres, so here's a brief rundown on what's currently available:

- *Vintage X Pro Synthesizers*: three volumes totalling 11 CDs, containing 8GB of classic analogue and digital synths and keyboards.
- *Mo' Phatt X, Planet Earth X, Virtuoso X*: complete sound sets from the original hardware modules.
- *Beat Garden X*: cutting-edge techno and dance instruments plus acoustic instruments and percussion.
- *Protean Drums X*: drums and percussion from acoustic kits to R&B, rock, jazz and more.
- *Techno Synth Construction Yard X*: dance, ambient and techno sounds from Rob Papen.
- *Street Kits*: urban dance drum sound library from Eddie Bazil of Samplecraze.
- *Beat Shop 2*: samples and grooves.
- *Modern Symphonic Orchestra*: five-DVD set containing 10GB of string, brass, woodwind and percussion.
- *Old World Instruments*: strings, winds and percussion from around the world.

BLACK SAFFIRE

SOUND IS PRECIOUS

Is it still there?

That mellow low end tremor of a thumbed bass string. The piercing tone squeezed out of the top fret. The velvet texture of a rich deep vocal. That endless resonating decay, as you strike each and every key.

Every subtle nuance coming together to make your sound.

And if you're not careful, it will all be lost.

Think about this when you consider your next interface.

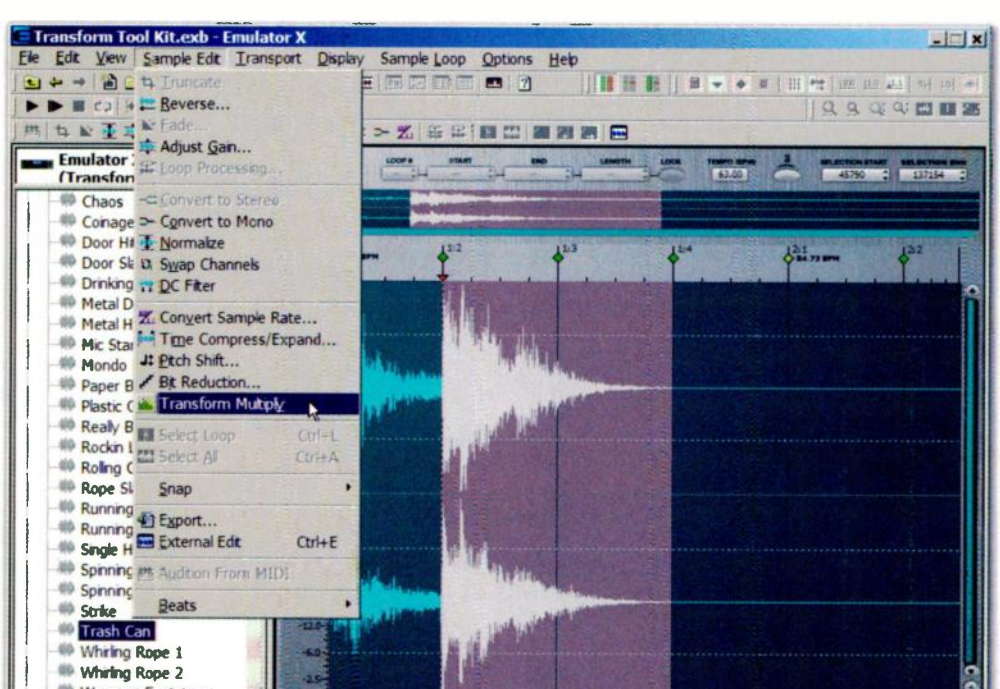
Focusrite Saffire LE - because your sound is precious to us

If your sound is precious to you, go to www.focusrite.com/saffireLE

audio heritage | sonic integrity | professional solutions

t: +44(0)1494 462246 e:sales@focusrite.com

Focusrite
www.focusrite.com



Convolution made easy — you can apply any section of one audio file to any section of another, enabling complex new sounds to be built up in stages.

▶ Sample Rate Convert). The name may be unfamiliar, but musicians who have used convolution reverb plug-ins will recognise the results straight away — Transform Multiply performs exactly the same tricks, but this time as an off-line rather than real-time process, which means the results don't require any further CPU when played back.

To use Transform Multiply, you select the desired sample in the sample bank, highlight any portion of it to act as the 'treatment', and then use the Copy function (or Ctrl-C keyboard shortcut) to place it on the Windows clipboard. Then you select the sample that you want to treat, select any or all of it for treatment, and finally choose the Transform Multiply tool from the Sample Edit menu and choose an intensity from the pop-up dialogue window — 0 percent is totally dry, intermediate values are more suitable for reverb-style treatments, while 100 percent provides the most extreme totally 'wet' effect.

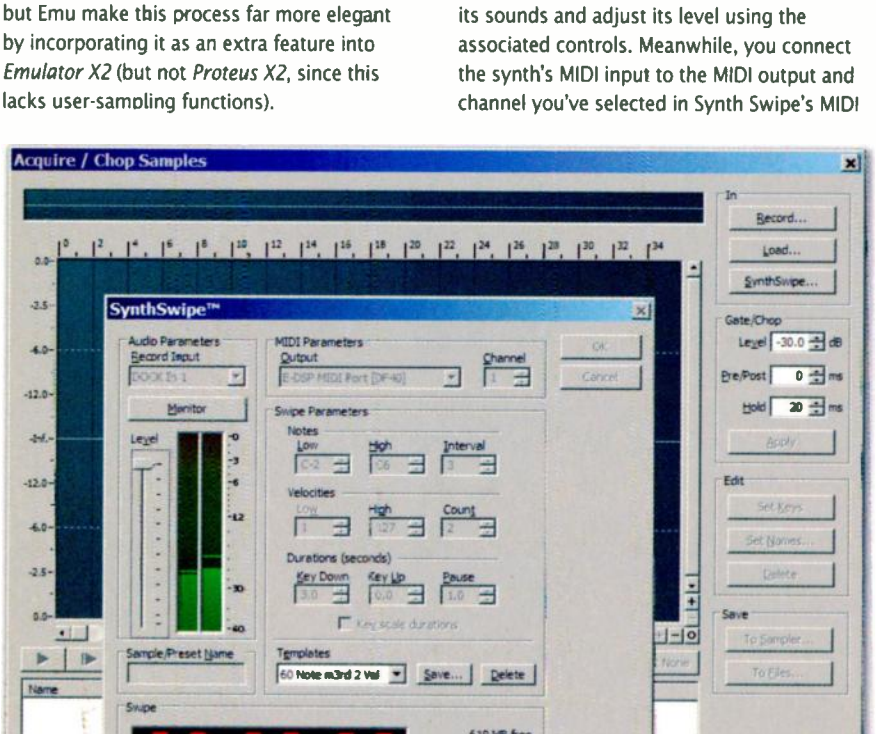
As always with convolution plug-ins, you don't have to restrict yourself to reverb treatments: applying shorter impulses derived from microphones or acoustic instruments can add their 'flavour' to existing sounds, while using speech will result in vocoded-like results, and repeated sounds will give exotic echo units.

You might think Transform Multiply is a poor relation to the real-time effects of a convolution reverb, but you'd be wrong. In conjunction with the new Twistaloop beat-detection engine you can quickly extract rhythmic sections from the source sample and convolve these with any section of the destination sample — within a few seconds I was applying different convolution treatments to individual beats in a loop, resulting in totally new sounds. This is yet another tool that yields some fascinating and totally unique results if you put in some effort, and once again goes way beyond what's on offer in other samplers.

Synth Swipe

And so to Synth Swipe. As more and more of us move from hardware to software synths, but don't want to leave our old sounds behind, it becomes increasingly important to have a way of capturing them in an automated manner. John Walden recently reviewed the stand-alone Skylife *Sample Robot* utility in *SOS* March 2006, but at £169, this is probably too expensive for most people wanting to transfer the contents of a few synths to their hard drives. A much cheaper Essentials version has since been released, but Emu make this process far more elegant by incorporating it as an extra feature into *Emulator X2* (but not *Proteus X2*, since this lacks user-sampling functions).

Synth Swipe is accessed from *X2*'s Acquire/Chop Samples page, and when you click on its Synth Swipe button a new dialogue window is launched. The Audio Parameters section lets you choose which input your synth is connected to, while you can monitor its sounds and adjust its level using the associated controls. Meanwhile, you connect the synth's MIDI input to the MIDI output and channel you've selected in Synth Swipe's MIDI



Synth Swipe helps you transfer the sounds of existing hardware synths across to *Emulator X* automatically.

Test Spec

- *Emulator X2* build 2.0.0.0490.
- PC with Intel Pentium 4C 2.8GHz processor with Hyperthreading, Asus P4P800 Deluxe motherboard with Intel 865PE chip set running 800MHz front side buss and 1GB DDR400 RAM, running Windows XP with Service Pack 2.
- Tested with Steinberg *Cubase SX* v3.1, Cakewalk *Sonar* v5.2.

parameters section.

Next comes the tedious bit — deciding on the lowest/highest notes to command your synth to play, how many notes in between, how many velocity values to be sent for each of these notes, and the note on and off durations. Now you just click on the Record button, and the audio output for each chosen note in turn will be captured at however many velocity settings you've chosen.

Once your note samples have been recorded, you return to the Acquire/Chop Samples page where they are automatically chopped into note lengths, and can also be auto-normalised and auto-looped *en masse* using a selection of algorithms with crossfade options, and finally turned into an *Emulator X2* preset. Then you can apply any of the normal engine treatments such as further filtering, LFOs and so on. Conversely, you could also capture the output of *Emulator X* itself (or another soft synth), in order to render complex CPU-intensive presets into versions that are less stressful to your processor. It took me a few minutes to tweak the recording levels to avoid clipping and explore the various options before my first Synth Swipe, but having done one I found it incredibly easy — this is an extremely powerful addition to the *Emulator X* feature set!

The X Factor

Some of you may have noticed my enthusiasm for the *Emulator X2* seeping through this review. I'd seen demos of the new features at trade shows and been suitably impressed, but it was still far more fun using them all than I'd expected, and occasionally jaw-dropping in the creative department. There are now quite a few soft samplers, but *Emulator X* has always had a good reputation for its audio quality, and the *X2* version stands out for its incredible versatility, although CPU overhead can rocket if you avail yourselves of too many of the features in a single preset.

Overall, *Emulator X2* is a major upgrade to an already impressive soft sampler, and as a £60 upgrade for existing *Emulator X* users should be snapped up immediately, as it offers a stunning range of new features. If you've already got another item of Emu hardware mentioned in the list of the beginning of this review, £190 seems a bargain for such a comprehensive soft sampler and 3GB bundled library. I also suspect that severing the requirement for an Emu soundcard, as the full retail version does, will now tempt others who already own a perfectly good audio interface but are looking for a soft sampler. SOS

information

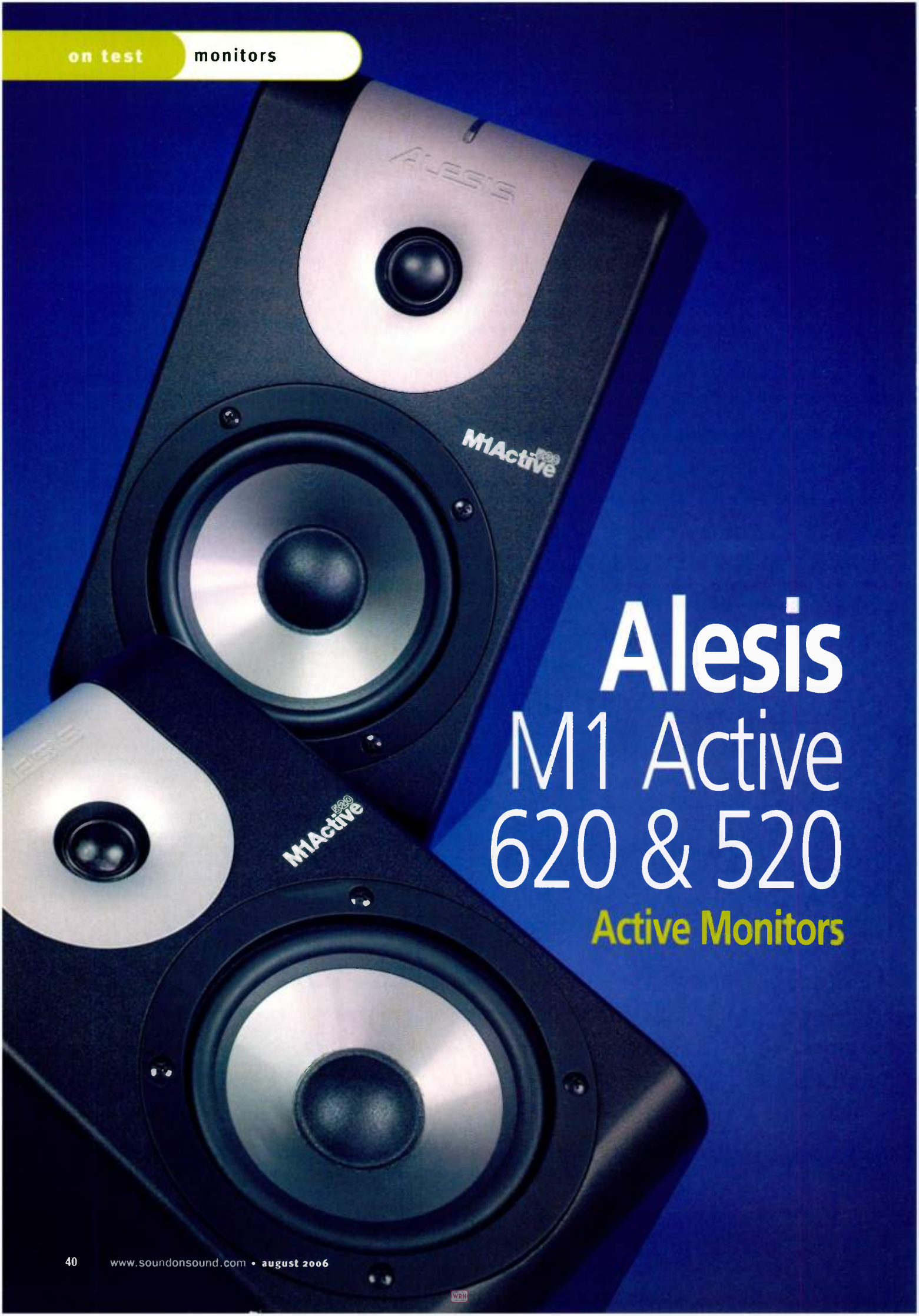
£ ***Emulator X2*** **retail version including Xmidi 2X2 MIDI interface £219.99;** ***X2*** **upgrade for existing** ***Emulator/Proteus X*** **owners £60;** ***X2*** **add-on for owners of other Emu products £189;** ***Proteus X2*** **add-on for owners of other Emu products £120. Prices include VAT.**

T **Emu +353 1 433 3201.**

F **+353 1 806 6788.**

T **Emu Europe +353 1 433 3205.**

W **www.emu.com**



Alesis M1 Active 620 & 520

Active Monitors

Stylish new powered monitors that won't break the bank.

Paul White

In previous issues, I've often mentioned the Alesis M1 Active MkII monitor as a good, budget-conscious solution for home studio monitoring. Now there are two new sets in the range, the M1 Active 620 and M1 Active 520, which are meant to sit alongside the M1 MkII, rather than replacing it. The new models certainly look very different to the M1 Active MkII, with their computer-style power switches on top of the cabinet and their blue illuminated power indicators (pictured overleaf). At heart, however, they are based on the proven engineering principles of a ported two-way active system, in this case with the port at the rear of the cabinet

Features

The cabinets are solidly built, with one-inch-thick front baffles. An attractive black textured coating is laminated onto the surface of the box and the cabinet's front edges are radiused to reduce diffraction and to improve their appearance. Both the woofer and the tweeter waveguides are recessed flush with the front baffle and fixed with cross-head screws. The smaller M1 Active 520 utilises a 75W amplifier, while the 620 has a 100W amplifier, and both models are magnetically shielded for use near CRT video monitors.

Measuring 165 x 267 x 197 mm (WxHxD), the compact M1 Active 520 weighs 6kg. Alesis give its frequency range as 56Hz to 20kHz, but as no reference level is given for these measurements (-3dB is the usual reference), this isn't hugely informative, and neither is a maximum SPL figure supplied. The one-inch, silk-dome tweeter receives 25W of power, while the five-inch woofer (a light but stiff polypropylene cone set into a rubber surround) receives 50W. The crossover frequency is set at 2.8kHz.

Weighing 9.6kg, the M1 Active 620 is somewhat larger than the 520, at 216 x 380 x 260 mm (WxHxD) and has an extended low-frequency response, reaching from 20kHz down to 49Hz. It uses the same driver construction at its smaller sibling, but with a 6.5-inch woofer accompanying the one-inch tweeter.

SOUND ON SOUND

Alesis M1 Active 620 & 520

pros

- Sensibly priced.
- Great styling and solid construction.
- Sound detailed without being fatiguing.

cons

- There's plenty of low end, even from the little 520, but the bass sounds a little larger than life, especially on the 620.

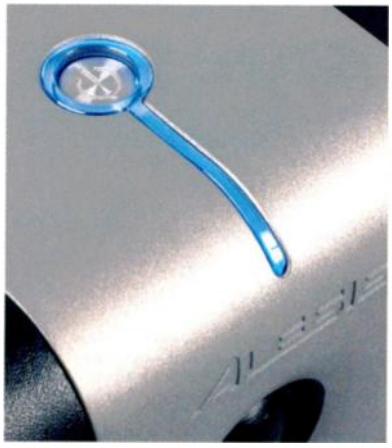
summary

While it is impossible to make a technically perfect monitor at any price, let alone at this end of the market, Alesis have again managed to find the best compromise, to deliver affordable monitors capable of great mix results.

Both models feature balanced XLR and TRS jack inputs and have an input-gain control and four slide switches, labelled Acoustic Space, Hi Boost, Mid Boost and Lo. Hi Boost operates a shelving filter with a turnover frequency of 3kHz and offers 2dB of cut or boost, as well as having a flat 'off' position. Mid Boost switches a 2.5dB wide-band boost at

ALESIS M1 ACTIVE 620 & 520

▶ 1.8kHz on or off. The Lo switch sets the built-in high-pass filter's roll-off frequency at either 56Hz, 80Hz or 100Hz. The Acoustic Space switch compensates for the build-up of bass frequencies that occurs when the speakers are used close to walls or to corners: there are free-space, half-space and quarter-space settings, which introduce an appropriate amount of bass cut. One final feature worth mentioning is that the monitors' blue power lights flash red to warn of clipping.



When the input is overloaded, the speakers' power lights flash red.

On Test

In my studio, the larger 620 monitors came across as detailed and smooth, with exceptionally good stereo imaging and plenty of low end. However, I felt the low end had a slightly plummy coloration that tended to make the bass sound just a bit more warm and round than normal. This characteristic is noticeable in many small to medium-sized studio monitors and is probably down to the port tuning artificially bolstering up low bass notes and kick drums before the response falls away at the lower cutoff point. It's not a problem, providing you get used to the speakers in situ, using good reference recordings that you know well. However, my own preference is for more sonic honesty in this area.

I preferred the 620s with the Hi and Lo Boost switched off, the high-pass filter at its lowest setting and the Acoustic Space switch set to its centre position. I had the speakers around 35cm away from the wall and well away from corners for these tests, and overall I found the sound revealing, with a satisfying low end. There's also a decent sense of front-to-back perspective, although not as much as you'd get from a really high-end monitor.

Switching to the smaller 520s revealed a broadly similar tonal character, but with a little less bass extension and seemingly a little less available level. Even so, the bass end is very credible and would perfectly suit many smaller home-studio spaces. If you crank up the level until the LED strips start flashing red on peaks, the subjective level is somewhat higher than I'd want to monitor at for any length of time. If you back off the level a touch, to stop the peak lights flashing, the sound is still clean and the bass is still under control.

Impressions

I've always thought Alesis monitors offered a good compromise between quality and affordability and these new models seem to follow in that tradition. Although purists could no doubt pick holes in some aspects of their performance, they are very easy to mix on and are revealing enough to highlight any problems in your sound sources. Both models have more bass extension than you might expect, and while I suspect that this isn't clinically accurate, it works well enough in most small to medium-sized rooms, providing you 'learn' the speakers using known reference material. Within their price range, the M1 Active 620s are hard to beat, although they have one or two worthy opponents, not least Alesis' own M1 Active MkIIs. The smaller 520s are a great choice for studios set up in smaller spaces or where desktop monitors are required, and even though they look slightly on the diminutive side, their performance is anything but. To sum up, both models are well worth investigating if you're looking for an effective active monitor at the lower end of the price range. SOS

Alternatives

There are lots of powered monitors to choose from at a similar price to the 520 and 620. Alesis' own M1 Active MkII is a possible alternative to the 620, and other choices include the Fostex PM1, the KRK Rokit range and the Tannoy Reveal 5A and 6D. Other alternatives include the Tapco S5 and S8 and the Event TR6 and TR8.

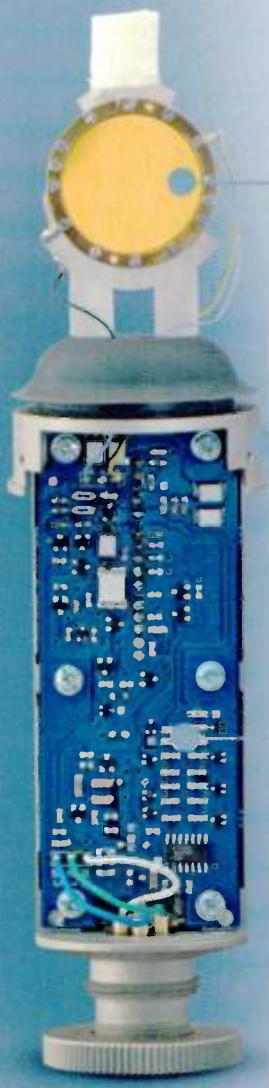
Information

£ M1 Active 620 £349.99; M1 Active 520 £269.99. Prices include VAT.
T Numark +44 (0)1252 341400.
F +44 (0)1252 353810.
E uksales@alesis.com
W www.alesis.com

BEFORE YOU BUY A MICROPHONE
GET THE INSIDE STORY

There are plenty of cheap mics on the market. But only Røde combines genuine affordability with advanced technology and the superior build quality necessary to make true high-end recordings.

So before you buy a microphone, be sure to ask about ...



... the capsule

Gold-sputtered 1" dual diaphragm, edge-terminated for improved low end response. Manufactured to the most exacting tolerances, and hand assembled in Røde's own laboratory grade clean rooms.

... the body

Precision machined body using high-quality materials and the latest automated systems to ensure negligible resonance and hence more accurate recording.

... the electronics

Surface mount electronics for pure sound and ultra-low noise.

10 YEAR WARRANTY

On all Røde microphones

RØDE NT1A

Great sound and ultra-low 5.5dB self-noise, all for £150

At this price, many people choose the NT1A as their first 'serious' studio condenser microphone. But with self-noise of just 5.5dB and a clear, accurate sound, the NT1A always remains a favourite. Typical comments from NT1A users include: "One of the quietest mics I've ever tracked with". "Fantastic for the digital medium – you won't believe the clarity and quiet". "One of the first mics I now grab for any source".

RØDE NT2A

You'll hear detail you didn't know was there

Combining characteristic Røde performance with switchable polar pick up patterns, high pass filter and pad, the NT2A sets the standards for both sound quality and versatility. Noise is virtually non-existent, while the sound is clean and accurate with a silky smooth high end and solid bottom. Equally impressive is the mid-range, which can sound 'scooped out' on so-called affordable microphones. Listen to an NT2A and you'll hear detail you didn't know was there.

RØDE K2

A true, high-end valve mic without the premium price

Rich valve sound, incredibly low noise and distortion, wide dynamic range, high SPL handling and the versatility of continuously variable polar pattern control from the dedicated power supply unit – all at an impressively affordable price. K2 is the microphone they said could not be built. Check it out at a Røde dealer – you're in for two great surprises.

RØDE® MICROPHONES

m.i.p.a.

HHB FIRST WE LISTEN

EXCLUSIVELY DISTRIBUTED IN THE UK BY HHB • T: 020 8962 5000 • W: www.hhb.co.uk

Señor Coconut

Uwe Schmidt: Recording *Yellow Fever!*

Yellow Magic Orchestra pioneered the use of electronic instruments and sampling. Now Uwe Schmidt, aka Señor Coconut, has used the same techniques to render their greatest hits as Latin dances, with contributions from all three original YMO members.

Tom Flint

The idea of reworking classic synthesizer tracks in the style of a Latin dance band may be inherently amusing, but *Yellow Fever!*, comprising 10 covers of Yellow Magic Orchestra songs, is anything but a joke. Far from being mere novelties, the songs are extremely well arranged and rhythmically complex, which is particularly impressive as they were constructed using little more than a laptop computer by just one man — Uwe Schmidt, creator of the Señor Coconut persona.

The Coconut Enigma

Over the past 20 years, Schmidt has made a habit of releasing off-the-wall records, mostly under pseudonyms such as Lassigue Bendthaus and Atom Heart. As far back as 1986 he helped set up the NG Medien label to manufacture and distribute 'audio tapes of diverse electronic music projects', later establishing his own aptly titled Rather Interesting label in an effort to promote new forms of electronic music. Despite maintaining a level of anonymity by the use of his numerous monikers, Uwe is a highly regarded producer and remixer, a reputation partially earned from his collaborations with high-profile international musicians such as the Yellow Magic Orchestra's own Haruomi Hosono, and experimental musician/producer Bill Laswell.

In recent years Schmidt has relocated to Chile, chiefly to experience a completely non-European musical culture and to nurture his interest in Latin rhythms, a process which eventually led to the creation of his *alter ego* Señor Coconut — an enigmatic South American bandleader specialising in Latin dance covers of Western electronic music!

Uwe explains how the idea evolved. "I started getting interested in covers about 10 years ago, having previously produced a lot of electronic music that you would categorise as techno. I was becoming fed up with the development techno had taken, and felt that it had stagnated a little, so I wanted to expand my musical language and have something different going. To do this I took a historical approach to my music listening and didn't buy any new records for years. I just bought old stuff and soon realised that there was a history of cover versions, particularly in Latin music.

"The first Señor coconut album, *El Gran Baile*, was more about combining audio tracks and cutting and pasting samples together to create a Latin atmosphere — it wasn't about songs or melodies. In those days I used an Akai S3200 sampler, Akai MPC3000 sequencer and an Akai eight-track digital recorder for overdubbing. It was a very minimal setup. There was a time before that when I'd started accumulating a lot of analogue equipment but I wanted to reduce the setup, so I replaced it all with the sampler and basically cut my entire record collection into pieces, categorising everything into long loops, short loops and

hits, and putting it all into folders. So for saxophone, for example, there were hits, lines, breaks, fills and categories like 'soft' and 'baritone'. I gathered everything I could without having a specific purpose for a sample. I was creating a database of sounds."

Krafty Work

The Señor Coconut record which really captured the imagination of the press and public was *El Baile Aleman*, an album consisting entirely of compositions written by the German electronic pioneers Kraftwerk, recorded as salsas, rumbas, merengues, cumbias, cha-cha-chas and baklans using samples of marimbas, trumpets, maracas and other typically Latin instruments.

"I started each song by spending a couple of days analysing the whole track — the core structure, melodies, stuff like that," remembers Uwe. "I'd spend so much time thinking about it that I'd have it recorded in my head and could imagine the modifications I wanted to make. At that stage the songs seem to automatically connect with sample grooves I have memorised from my database, so when I try out the line or loop I have in mind I usually find that I only have to modify it a little bit. Then it was a matter of constructing the songs by cutting and pasting samples taken from records rather than commercial sample libraries. For percussion, I certainly prefer the texture of records.

"I began building the rhythmical section before deciding which instrument should play what part during each song segment, always bearing in mind that there was going to be a coherent combo playing. I tried to imagine how a traditional arranger would handle the available resources and how they'd spread them throughout the song. Since I'm not a trained arranger this took me quite a while. The marimba rolls, for example, I programmed in quite a microscopic way, which took a lot of patience.

"I didn't have a total recall system back then, so I had to program the entire structure from top to bottom and had to mix the song down in one pass. I did use a Yamaha 02R digital mixer, but that had very limited recall, so it was very difficult and I needed a lot of concentration and focus. It usually took me a month to do a song."

Fever Symptoms

By the time work started on *Yellow Fever!* the Akai hardware had been replaced by a 1.33GHz Apple 12-inch G4 Powerbook and a Digidesign Digi 002 Rack Firewire interface, running *Pro Tools LE*. The new setup, with its total recall and more sophisticated audio editing tools, made it possible for Uwe to entertain a more ambitious method of production. Having had a number of years to reflect on the relative successes and shortcomings of *El Baile Aleman*, he concluded that using just his library samples would not enable him to obtain the musical complexity and subtle textures he wanted.

"I was listening to big band records from the '50s and '60s, where the tightness of the playing and sound was amazing, and at an early stage in the project I knew I wanted that kind of texture and feeling. One option ▶

Uwe Schmidt's minimalist studio, based around an Apple Powerbook and a Microkorg keyboard. The mic is a Soundfield ST250.

would have been to write the scores and then rehearse the whole album with a full band, but I decided that, for someone like me who doesn't have a band that is playing every night, it would be impossible to get that sound in a traditional way. You have to remember that those people played together for a long, long time and were doing the same set over and over, so it was not a big deal for them to go into the studio and record together. We're living in a different age, and there wasn't the budget for that anyway."

Uwe solution was to hire a trained arranger who could score the music, and then find some musicians to record the various parts one at a time, so he could later manipulate them from within Pro Tools. "Once I'd established that it was impossible to record the songs in the traditional way, my plan was to chop up the recordings, combine the parts with samples and build it all back together, and that's what I did. I quite literally spent two months manually cutting every note on this album and rebuilding the tracks! So although it was recorded with real instruments and musicians, in the end that was just the raw material."

Before any recording could begin, however, Uwe had to decide what Latin rhythms he was going to use, and then create guide templates. "When I'd selected my songs I imagined which Latin rhythm to use for each one and at which speed that should happen, bearing in mind that certain rhythms are connected to certain bpms. When that was figured out I generated some very basic guide rhythms which I either took from a sample library or programmed myself. I then synchronised the original Yellow Magic Orchestra song to the timeline by cutting and stretching it to fit.

"I sent the results to my friend Norbert Kraemer, who did the arrangements and produced the scores for the musicians following our discussions about which type of arrangement we wanted for each song, which instrument to use for each part or line, and so on."

German Latin

The instruments, including marimba, vibraphone, acoustic bass, saxophone, trombone, trumpet and vocals, were recorded over a 10-day period in a studio in Wellnoise Booth Studio, Cologne. All the mics were plugged directly into Uwe's Digi 002 interface and recorded straight to *Pro Tools* running on the 12-inch laptop. The musicians were each recorded one at a time, using a variety of microphones. A single AGK112 was selected for acoustic bass, while a pairing of Neumann KM84s was favoured for mallet, percussion, marimba and vibes. "For marimba and vibes I used the modern classical orchestra method of recording; placing the KM84s approximately 40 to 50 cm above the plates so that they captured the entire range of the instruments," recalls Uwe. "For some takes we also positioned a pair of the Studio Projects C3s under the marimba for a less percussive signal. We ended up with eight marimba signals which I mixed down to mono."

Everything is monitored through Uwe's Meyer Sound nearfields.

Although some of the trumpets and saxophone parts were also captured by the Studio Projects C3 microphones, a Sennheiser MD441 dynamic microphone proved ideal for the rest of the brass including, trumpets, saxes, trombone and bass trombone. All the 'horn section' instruments were close-miked to obtain a dry sound that would leave Uwe with effects options during the mix. "For some conga takes we also decided to record the reverb of a neighbouring room using one of the Studio Projects mics," adds Uwe, "but no other 'real' room reverb recording was used."

Bringing It All Back Home

Back at his 'Mira, Musica!' home studio in Santiago, Chile, Uwe set to work attending to the rhythms that would underpin each composition. Having begun his career as a drummer, before switching to programming, he was particularly concerned about getting this element exactly right. "My musical approach is always rhythmical, so I usually adapt everything to the rhythm. Since we recorded the parts one at a time there wasn't really a cohesive swing or groove to the rhythm section, so it was quite un-tight.

"I looked for eight or 16-bar guide grooves on Latin records and cut the sessions accordingly. It was a matter of looping the sample groove and then adjusting the audio by manually moving the notes so that everything was swinging in accordance with the loop. I also looked for bits and pieces in my library and combined them with the recording. I don't buy sample libraries, so they are all things I've collected over the last 15 years, some of which are fragments from the Akai S3000, although I occasionally use one of the standard *Battery* kits when I need a particular conga or shaker."

Not all the tracks easily lent themselves to interpretation in the Latin style. One of the most impressive songs on the album, 'Firecracker', was also one of the most problematic. "When the scores were written, I have to admit that I wasn't sure exactly what rhythm that should become," explains Uwe. "The melody didn't fit with any Latin style that I knew and there was a lack of time to think about it. When I listened back to the recording I figured it was too slow. The accent was a merengue but we'd recorded it at a cha-cha-cha speed. It didn't work at all and everything I tried sounded crappy, so I decided to make it faster, and that meant cutting it all, note by note. I never used time-stretch — every note was manually moved to its new position by dragging it along the timeline while the relevant section was looping. And I didn't

WIN an SH-201 Synth
www.roland.co.uk/xtour for details

Fantom X

and introducing the brand new SH-201 synth

Beat The Traffic > UK Production Tour

September Tour Dates

18th	Sound Control, Manchester City
19th	Sound Control, Leeds
21st	Sound Control, Oxford St, London
22nd	Dawsons, Reading
23rd	PMT, Birmingham
26th	Sound Control, Norwich
28th	PMT, Northampton
29th	Sound Control, Stoke
30th	Sound Control, Southampton

All start times are on www.roland.co.uk/xtour

October Tour Dates

2nd	Sound Control, Bristol
4th	Sound Control, Birmingham
5th	Sound Control, Manchester (Salford)
9th	Sound Control, Glasgow City
10th	Sound Control, Edinburgh
12th	Bonners, Eastbourne
14th	Absolute Music, Poole
19th	Andertons, Guildford
20th	Wembley Keyboard Centre, Wembley
24th	McCormacks, Glasgow
25th	Sounds Live, Newcastle
26th	Dawsons, Liverpool

Catch the Fantom-X UK Production Tour and discover how the Fantom-X series of workstations is changing the sound of music.

Awesome synth power, flexible sampling, fresh real-time control, supremely expandable and just as happy in the middle of your computer rig as it is on stage.

With artists all over the world switching to the Fantom-X sound, you'll be seeing them centre stage and hearing them on records all through the summer.

The brand new SH-201 synth also makes its UK debut on the tour, combining power and simplicity in one affordable synthesizer.

Beat the traffic during your journey home and call in: you could win one of the first SH-201's in the UK.

www.roland.co.uk/xtour

Roland

Because nothing matters more than music.

RECORDING *YELLOW FEVER!*

use shortcuts like snap to grid to help me place notes; I just kept listening to it until I got it right.

"Aligning a note on the beat doesn't always work, because different instruments have different attack times. For example, if a baritone saxophone player has to play a long, deep note, which is difficult, it takes him longer to gather breath and hit the note, so he's always off-grid. You can't snap it to grid and get it sounding good, so you have to analyse how the instrument is played. I like to zoom in and study the Session on screen visually. You can see that the saxophone player is always late, but it sounds good, so I take a mental snapshot of the timing and that helps me offset each instrument when I start adjusting notes.

"I also decided to have it speed up as the song goes along, so it actually goes from something like 106 to 130 bpm! I managed to do that part quite well so it's not too obvious, which was quite hard on a digital processor!

"I worked on blocks of 16 or 20 bars at a time, and each one I mixed separately using very complex automation to change scenes and levels. There's was no traditional mix approach of having channels where you set the EQs and effects and press Start. For example, I usually bounce the effects, so there's no fixed send for them. I was mixing blocks and then joining them together, basically."

Breaking The Illusion

Although some sections of *Yellow Fever!* sound almost like a traditional band playing together, other parts are interrupted by deliberate sample triggering and looping effects, and even the odd electronic synth squeal. Uwe explains his rationale for using such devices within the compositions. "Each segment or section of a song called for a certain complexity. Very often the YMO achieved different moments in the song by switching sounds on the synthesizer, but that wasn't appropriate in my arrangements, so I had to either cut out certain sections where it was getting repetitive, or do a lot of trickery like go from an acoustic part into an electronic part, thin something out, or play with effects. Whatever I did depended on how difficult or easy it was to keep one's attention throughout the song. If I was listening to a whole song and felt that there was something missing I'd consider doing

The Return Of The YMO

Not only is *Yellow Fever!* an album of Yellow Magic Orchestra songs, but it also features musical contributions from all three original members of the band itself. Having previously collaborated with YMO's Haruomi Hosono on two HAT albums, which were released on his own Rather Interesting label in Germany, and Hosono's Daisy World in Japan, Uwe decided to ask all three YMO founders if they would perform on *Yellow Fever!* "I've been in contact with Hosono for about 10 years, and with [*Ryiuchi*] Sakamoto for three, so the only person I didn't know was Takahashi," says Uwe. "Once you know certain people who are involved in the Tokyo music scene it's not hard to get in touch, but I tried approaching them officially through their management, which wasn't easy because they are very famous people in Japan. I couldn't get any feedback, but we'd heard from some people who were working with them that they were interested in participating. During this time the album was progressing so there was becoming less space for them to participate.

"I asked each of them to play on the songs they'd written and provided a few choices. Sakamoto decided to play Rhodes on his song 'Yellow Magic', and that worked pretty well. We sent him a reduced six-channel production so he could find space and the mood. I asked Hosono to sing on 'The Madmen', and Takahashi to sing backing vocals on 'Limbo'. I left the mixing of those songs until last in the hope they'd be able to participate, and fortunately they sent the material with about 10 days to go. There wasn't much to do to the vocals aside from selecting the best takes, but I had to do some chopping of Sakamoto's Rhodes to fit it into the spaces within the arrangement, which was already very full."

Between each of the 10 YMO tracks on *Yellow Fever!* sits a curious musical interlude, typically featuring grainy sampled loops and guest vocalist contributions; as Uwe explains, this is also a nod to the original recordings. "I'd decided on the track order before recording, so I had a clear idea of how I wanted the songs to join together. There are, I think, two YMO records with interludes, so I was simply quoting that idea."

The Yellow Magic Orchestra in full flight.

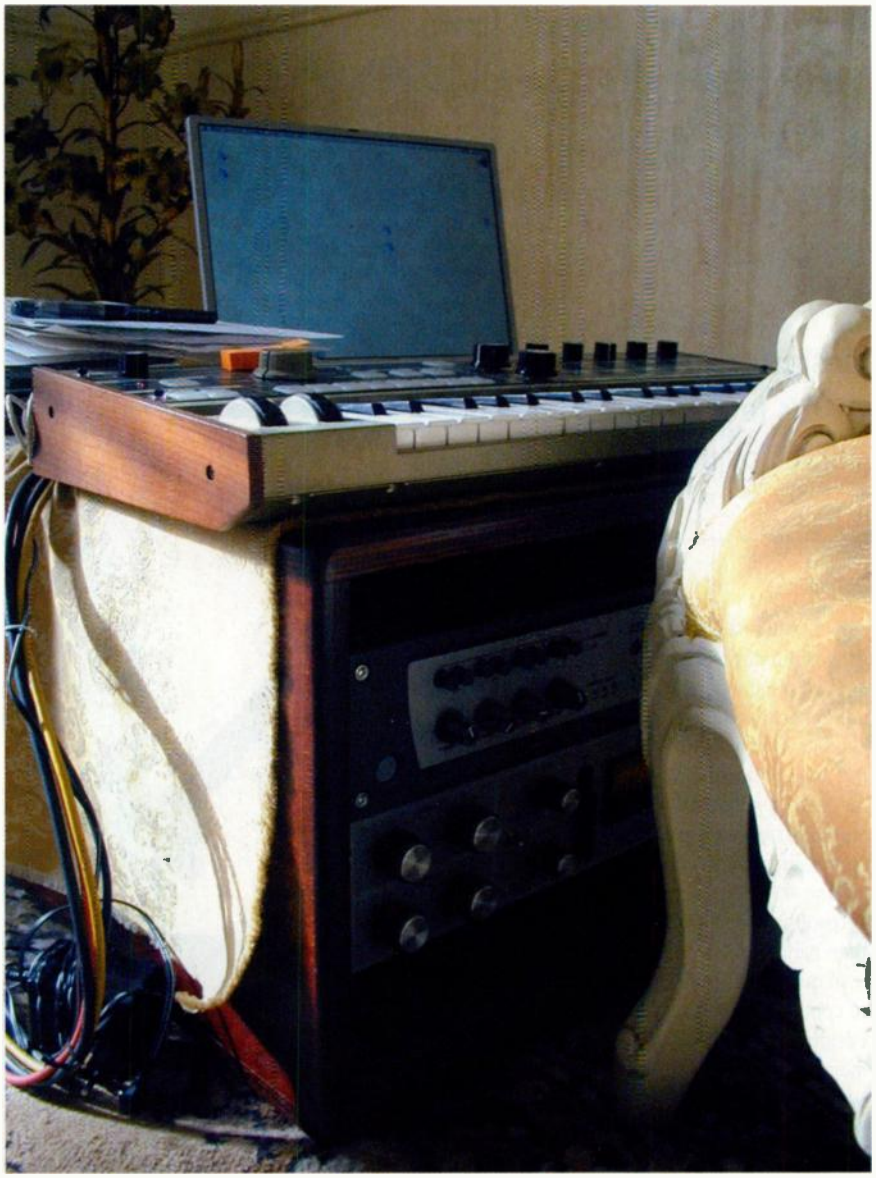
something like that.

"I also wanted a more eclectic sound than the last three Coconut albums. When I found it necessary for the flow of the part I would make the samples blend in, but in general that was not very important. I tried to adapt some of the new recordings to sound like the old samples using amp simulations, and *vice versa*. Sometimes a sample is obviously a sample, other times it's not. I wanted to play with that idea, so it's a bit like running through different historical moments of Latin music. The idea was not to stick with certain sounds but to emulate different sounds in different songs or sections of a song."

Part of Uwe's plan for giving *Yellow Fever!* an eclectic feel was to encourage vocal contributions from a variety of international artists, instead of using a single lead and fixed backing combo. "I asked other musicians to participate, and that gave me lots of little bits and pieces to deal with and incorporate. I emailed them little musical sketches, or sometimes only words, and they sent me their interpretation. Most of them stuck very closely to what I told them. They emailed it back to me and I combined it with what I had and what others had sent. One song features both Marina from Nouvelle Vague and Towa Tai, but they never heard each other's contribution. For me, one of the most interesting things was getting all the different parts and giving them coherency."

Mixing & Processing

Remarkably, the entire album was mixed on Uwe's laptop using a pair of Meyer Sound HM1 nearfields for monitoring, and a collection of Pro Tools plug-ins for processing. "I mostly used the Sony *Oxford EQ* and *Dynamics* for processing and the version of *Amplitube* that comes free with Pro Tools for effects," explains Uwe. "I didn't really use any delays other than for the odd little dub-style effect parts, but if there are any they will also be from *Amplitube*. Sometimes I used very cheap reverbs, like the *D-Verb*, which comes free with Pro Tools, and I used a lot of reverbs from *Amplitube*. Then there is also a bit of Waves *Renaissance Reverb*. I usually put two reverbs on each song; one small, close one and a distant one. I wanted the quieter instruments like the saxophone section to sound close and strong, and the loud instruments, like the trumpet and trombone, to be far away at the back of the room as if there was one mic standing there listening to them all. Having said that, I based the sound mainly on records from the '70s where some instruments were recorded together, some separately, then the reverbs were added afterwards to create a particular space. I didn't want to overdo the reverb so it's all relatively dry, and there are also a lot of unprocessed sounds in the mix.



Apart from the Digi 002 Rack, the only outboard in Uwe's studio is a Urei 1178 compressor (bottom).

"On the last three records, working from my sample libraries, it wasn't possible to give the same sound to a set of instruments, but having a well thought-out arrangement gave me a clear picture of how the band looked in terms of how many instruments there were and how they played, so it was much easier to imagine how each one should sound and where to place it in the virtual room.

"I mixed it down and did the pre-mastering on the same computer, but the final master was done in a separate studio in Germany. My pre-mastering is about fixing little bugs that are impossible to mend during the mix. Mixing a song may take a week or two so it's easily to get lost, and one's hearing adjusts to the sound of the song so you just don't know if it's too bright or dull. If you leave it for a week you realise what it is like, but going back to such a complex patchwork of a mix to fix it is impossible so I prefer to do it in the pre-mastering. By listening to all the tracks in a row I can hear the mistakes, so I balance all the songs using a touch of EQ and Waves' *C4* multi-band compressor to make it a bit more even."

Recovering From The Fever

Uwe admits that making *Yellow Fever!* was much more difficult than *El Baile Aleman*, summing up the editing process as "very anal and obsessive"! As yet, no other Coconut projects have been planned and Uwe is enjoying a well-deserved rest. It does seem certain through that, having achieved such an impressive result with his newly perfected recording and editing process, there will be more to follow. SOS

AKG Perception 200

Condenser Microphone

Designed in Europe and built in China, AKG's latest studio mic aims to deliver value for money.

Paul White

Austria's AKG have a long-standing reputation for building and designing high-quality recording microphones, a number of which have become studio classics. However, they also acknowledge the need to compete with the avalanche of far eastern microphones in the project studio sector, and that's tough to do with European manufacturing costs. To get around this, AKG designed their new Perception models in their Vienna facility, then had the mics built in China under their own quality control regime, the idea being to combine the advantages of Austrian design and Chinese manufacturing to maintain an attractive price.

Features

Conceptually, the Perception 200 reviewed here is a fairly conventional cardioid-pattern, large-diaphragm (one inch) true capacitor microphone, running from 48V phantom power. The Perception series includes a second model — the Perception 100. Both mics share the same capsule design, basic technical spec and construction style, the main difference being that the Perception 200 is equipped with two additional switches, for a low-cut filter and a -10dB pad. The Perception 200 also comes in a neat metal carry-case, complete with a heavy-duty spider-type shockmount. The shockmount is very nicely designed using a metal frame with sensibly large securing hooks to hold the elastic suspension bands. A threaded ring at the base of the shockmount screws into the

SOUND ON SOUND

AKG Perception 200 £169

pros

- Tough, stylish construction.
- Clean sound with a hint of flattery at both the low and high end.
- Includes case and shockmount.

cons

- None at the price, but make sure the mic's character suits the principal voice being recorded.

summary

AKG would appear to have succeeded in building a very attractive and warm-sounding studio microphone at a price most project studio owners can afford.

base of the microphone to fix it securely to the mount, so you can use the mic upside down without any fear of it working loose.

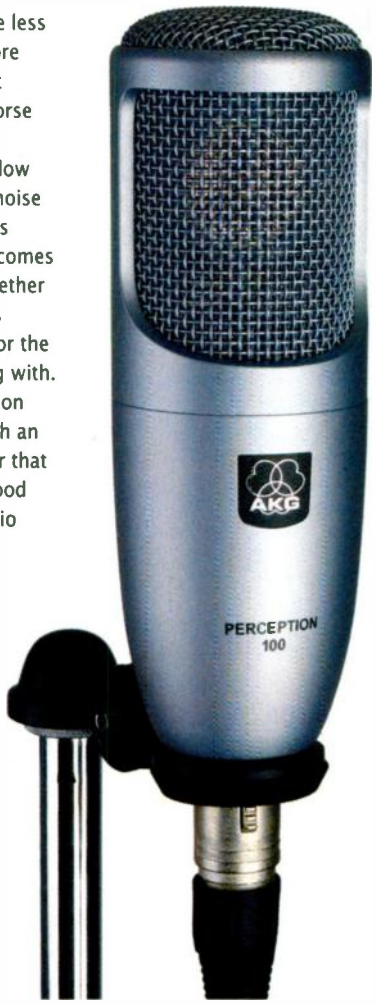
The frequency plots supplied by AKG for both models are nominally flat above 100Hz, with a gentle roll-off below that, and there's also a very gentle presence hump up at around 10kHz, which should add a nice airy quality to the sound. The overall response is quoted as 20Hz to 20kHz.

With an open circuit sensitivity of 18mV/Pa, the Perception 200 has an A-weighted noise figure of 16dBA, or a signal-to-noise ratio of 78dBA referenced to 1Pa. This is very typical for a microphone of this type, and although you can buy quieter microphones the specification in this respect is more than adequate for typical close-miked vocals and other common studio applications. The maximum SPL without the pad switched in is 135dB (the same as the cheaper Perception 100) and the low-cut filter rolls off at 12dB per octave below 300Hz, to compensate for the proximity effect or to exclude low-frequency sound when recording instruments that have no significant low end.

Physically, the Perception models have a distinctive truncated bullet shape, the 200 model being differentiated by a light-blue metallic finish and two recessed toggle switches for the filter and pad. The body is solidly made from cast alloy, weighing about half a kilogram, and there's a robust dual-mesh steel grille protecting the capsule. Unlike most Chinese-made mics, which can be accessed by unscrewing a locking ring at the bass of the body by hand, these models use a recessed locking ring that requires a special tool to undo. Not having such a tool, I couldn't take a look inside to comment on the internal construction of the mic, but I don't anticipate any untoward surprises there.

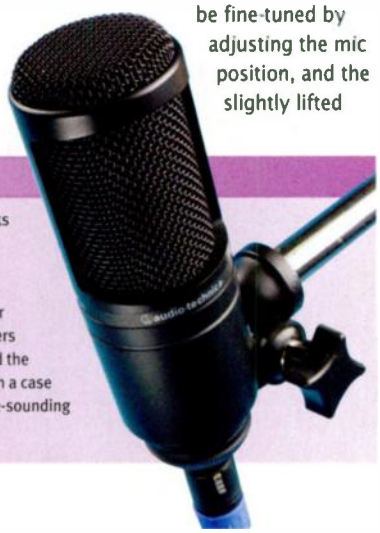
In The Studio

As a vocal mic, the Perception 200 combines a warm low-end response with a subtle top-end airiness, which is what I'd expect from the shape of the frequency plot. For comparison, I tried it alongside the low-cost Audio-Technica AT2020, which I felt hyped up the low end a little less and also sounded more focused, but it wasn't a case of better or worse — just different. Both produced a similarly low level of background noise with the preamp I was using, so, as ever, it comes down to deciding whether the mic in question is a good tonal match for the singer you're working with. Certainly the Perception 200 doesn't have such an overbearing character that it wouldn't make a good general-purpose studio mic, but at the same time it still has a slightly flattering sound that will inevitably suit some voices better than others. If you're working with a voice that needs both gentle low-end support and help with high-end projection, the Perception 200 should do well, but, by the same token, if the singer you're working with already has a strong low end, you might need them to back off from the mic a bit, to prevent the low end from becoming over-emphasised. Of course, you can also use the low-cut filter on the Perception 200, although with some voices this may thin out the sound too much. The call is definitely an artistic one.



The AKG Perception 100 is cheaper than the 200 but doesn't have a pad or low-cut filter.

As an acoustic guitar mic, the Perception 200 works fine, providing you take the time to find the best spot to position it. Any tonal flavour the mic has can be fine-tuned by adjusting the mic position, and the slightly lifted high end helps bring out the string articulation without making the result sound scratchy. Although the mic has a cardioid pattern, it still picks up other sounds from around the room, so you need to use the mic in a controlled environment where possible, but this is true of most sensitive capacitor microphones.



Conclusions

Given that the Perception 200 is priced to compete head-on with other Chinese-made microphones, it has to be considered great value for money, especially taking its excellent build quality and bundled shockmount into account. As my tests showed, it has its own tonal character, which is true of the majority of large-diaphragm cardioid studio mics, but this character isn't so pronounced that it restricts the mic to working with only a small number of voice types or instruments. If you buy a mic to record yourself or a regular band member, you should still compare it with other similarly priced models, in case one suits the voice better (see 'Alternatives' box).

Whichever way you look at it, the Perception 200 is a very nicely designed and solidly built studio microphone that has the makings of a good all-rounder. If you're in the market for a sub-£200 studio mic that's built to last, you should definitely consider the 200, but if your budget won't stretch that far, you could check out the lower-cost Perception 100, which will give you essentially the same results, unless you need to pad down very loud sound sources. SOS

Alternatives

It really does pay to take a little time to track down the mic that works best for you, and there are lots in this price range to choose from. Of those that I've personally tested, I'd suggest looking at the Audio-Technica AT2020, which I think is a real gem for the price. It's cheaper than the Perception 200, but it doesn't have a built-in pad or high-pass filter and it doesn't include a shockmount. Other contenders include the Rode NT1A, which has a very clean and classy sound, and the slightly more expensive SE Electronics 2200A, which also comes with a case and shockmount. MXL and Studio Projects also make some very nice-sounding mics in this price range.

Information

£ Perception 200 £169.99; Perception 100 £119.99. Prices include VAT.
T Harman Pro UK +44 (0)1707 668222.
F +44 (0)1707 668010.
E info@harmanprouk.com
W www.harmanprouk.com
W www.akg.com

Studio SOS

Simon's original equipment setup left a lot to be desired in terms of ergonomics, with the two main computer screens off to one side of an already asymmetrical monitoring position.

This month's task involves stripping down Simon McEvoy's entire studio to improve the acoustics and ergonomics.

Paul White & Hugh Robjohns

By day Simon is a computer specialist in the IT industry, but when he gets home he spends his time writing and recording music, much of it for musical stage productions. His studio setup, in a converted garage at the bottom of his garden in the Buckinghamshire town of Princes Risborough, is based around *Logic Pro* v7.2 running on a dual-2GHz Mac G5 with Wharfedale Delta 70 two-way passive monitors powered by a Samson Servo 250 power amplifier. On the day of our visit, a second pair of Sony three-way monitors powered by a separate amplifier were set up at the rear of the room for when louder playback listening was required. He uses a Roland A80 as a master keyboard and in addition to software instruments Simon also has a Korg Triton keyboard synth, a Deep Bass 9 synth module with integral MIDI-to-CV conversion, and an old Korg Mono Poly running from the Gate and CV outputs of the Deep Bass 9. An Edirol UM880 provides the necessary MIDI ports, while a MOTU 828 MkII looks after the audio interfacing. An M-Audio Trigger Finger furnishes a convenient means of tapping in drum parts, and a small Behringer mixer was being used as a master level control.

Paul and Hugh quickly decided to reorient the studio along one of the shorter walls, so Paul set about removing all the cabling and equipment so that they had room to rearrange the furniture.

Simon called us in because he was unsure as to the accuracy of his monitors and monitoring environment, and he also felt that the ergonomics of his setup could be improved to make it easier to start working on a song without having to deal with lots of technicalities first. We arrived to be greeted by no fewer than four packets of chocolate Hobnobs, so this was clearly going to be a good day! We took our coffee and biscuits down to the studio, which was essentially a rectangular room lined with plasterboard on studding with Rockwool insulation behind. The rear wall of the studio was spaced further away from the outer wall to improve the degree of sound isolation. None of the plasterboard was plastered, and there was no acoustic treatment anywhere, so as soon as we walked into the room we could hear that it sounded very live, but this was easily dealt with, as explained in the 'Acoustics Fixes' box elsewhere in this article.

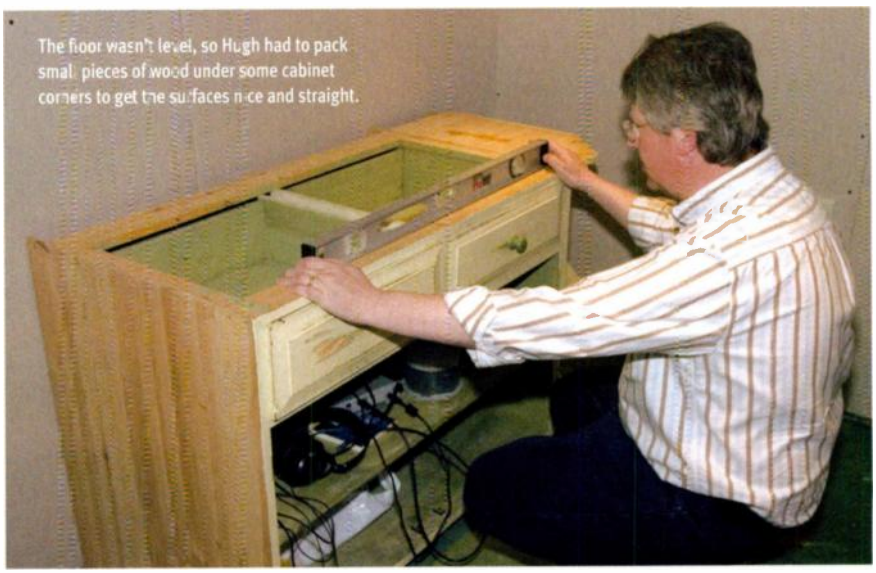
Rebuilding The Studio Setup From Scratch

Simon had his equipment set up on benches and worktops scavenged from his kitchen refit, but the setup was pretty cramped and was all pushed into one corner of the room, which made the monitoring very

asymmetrical. The Wharfdale speakers and Trigger Finger were set up in front of a window looking through to the other half of the garage, but the two huge 21-inch CRT monitors he was using with his G5 were set up on a bench to his left, as they were too deep to fit on the worktop in front of him. This meant he had to turn his neck to the left every time he wanted to look at the computer screen, which must have been very fatiguing.

The Roland A80 keyboard was set up on an impromptu shelf supported on piles of concrete blocks just below the main worktop. To get the Mac's Airport wireless system to connect reliably with the base station in the house, Simon had also put his G5 on the desk right in front of him, which put him very close to the cooling fans. These are reasonably quiet, but still noisier than ideal, especially when they're only inches from your face! A separate work surface was set up on wooden trestles bought from Ikea, and there were a couple of keyboard stands for the Triton and Mono Poly.

We quickly came to the conclusion that the system would be much better if it was set up centrally across the shorter wall of the room, with the computer monitors in their correct place directly in front of the mix chair. To achieve this we dismantled absolutely everything and then moved it out of the room while we rejigged the cabinets and worktops. This proved to be fairly straightforward, though we did have to cut some wooden packing pieces to compensate for an uneven floor. To accommodate the deep CRT monitors, we simply set up the worktop around 10 inches from the back wall so that the rear of the monitors could overhang. This left room in the front half of the worktop for the Trigger Finger, the QWERTY keyboard, and the Behringer mixer. ▶



The floor wasn't level, so Hugh had to pack small pieces of wood under some cabinet corners to get the surfaces nice and straight.

Banishing the crackles from a couple of pots on an old Korg Mono Poly involved just removing a couple of screws to whip the top open and then squirting a little contact cleaner into the underside of the offending controls.

The power amps, interfaces, and Deep Bass 9 were left installed in a flightcase rack and placed to the right of the main work surface.

We rebuilt the concrete-block shelf supports for the A80 controller keyboard under the main worktop, but dispensed with the separate Ikea trestle table completely. We also swapped the keyboard stands around, putting the Triton on a small keyboard stand Simon had picked up on Ebay. We then used an 'X'-frame keyboard stand behind it to support the Mono Poly a few inches higher. These were just in front of the rack case, over to the right of where Simon would now sit, but that wasn't a problem, as he normally plays them via his Roland A80 most of the time — and they were still near enough for

Acoustics Fixes

Because Simon hadn't plastered the walls, and we didn't want to mess up the surface to prevent him doing so in the future, we decided against gluing foam panels to the walls, and instead opted to use some spare Ready Trap samples from Ready Acoustics that we'd had sent over from the US for review. Ready Traps are, in fact, very nicely made fabric bags with concealed zippers that take standard two by four-foot Rockwool slab panels up to four inches thick.

Simon had a couple of packs of rigid Rockwool available, so we filled the bags with double thicknesses of 30mm slab taped together at the ends. As Rockwool tends to stick to everything it touches, we put plastic bin liners over the Rockwool, slid on the Ready Trap bags, then pulled out the bin liners. Straps are sewn onto the backs of the bags for hanging them, so we simply installed threaded plastic plasterboard fittings in the walls where we wanted to hang the bags, and screwed large cup hooks into them, then hung the bags directly onto them. The arrangement of the straps means that the bags tend to hang with the top angled away from the wall by a couple of inches, which actually makes them more effective by providing a small air space behind them.

We positioned three bags on the front wall, two on the right-hand wall and one on the left-hand wall. We'd have liked to be more symmetrical, but Simon had built a window into the left wall and we didn't want to obscure it. A further panel was hung above the mixing position and a final one propped up on the rear wall. Currently Simon has an organ and piano at the back of the room, so there wasn't much exposed wall space to treat.

Because the room was panelled with plasterboard, there was a lot of natural bass trapping anyway, and the Ready Traps effectively mopped up the mid-range and high-frequency reflections. The result was a room that sounded reasonably dead to talk in — very different to how it was when we first arrived — and the stereo imaging was markedly better. Simon was surprised at how much difference the acoustic treatment had made, and agreed that the monitors now sounded surprisingly good.

Some Ready Traps bags which Paul had brought with him allowed him and Simon to cover some Rockwool slabs and hang them from the studio walls from cup hooks as acoustic treatment.

PLUGSOUNDPRO.

COMPLETE WORKSTATION

massive grooves and instruments

The first software workstation combining a massive 8 GB core library, easy groove manipulation, preset management and universal compatibility.

- New Hybrid UVI Engine, seamlessly combining instruments and grooves
- 8 GB library, including the complete award-winning Plugsound Box, plus more instruments and loops
- Expandable with UVI Soundcards, Rex files, Apple Loops, WAV & AIFF
- Disk streaming for larger instruments
- Unlimited polyphony / ultra low software latency
- Smart integrated browser with realtime preview for grooves
- 64 parts & 64 MIDI channels (create stacks and layers in seconds)
- Expert mode with zone splits and key-switches
- Comprehensive sound shaping tools and effects
- Direct sync to host tempo & transport
- State-of-the-art realtime time-stretch
- Drag & Drop MIDI & audio files into the host sequencer's tracks
- Mac: AU, VST, RTAS & MAS • PC: VST, RTAS, DXi as well as Mac/PC stand alone versions

UltimateSoundBank www.ultimatesoundbank.com

Exclusively distributed by Time+Space Distribution Ltd.
www.timespace.com sales@timespace.com +44 (0)1837 55200

TIME+SPACE

Simon Says...

"Thank you both so much for the Studio SOS. I only had a brief time in the studio after you left last night, but still managed to knock out some new material. The new setup with its improved sound has already given me added inspiration for my work, not unlike the rush of excitement accompanying the purchase of a new piece of kit. The studio now feels much larger due to the logical positioning of equipment, while the new positions of the monitors directly in front has given my neck a long-deserved rest. Whilst the sound bags provide excellent sound absorbency, they also look good too, and thanks to the Auralex Mo Pads I can now hear a good level of bass. The *Logic* template Paul created in minutes is a life-saver. Having put a copy in my Startup Items I no longer waste valuable time waiting for samples to load and working out the correct synth inputs on the MOTU before starting to write."

E music@smcevoy.co.uk

▶ him to manipulate the analogue controls on the Mono Poly when required. Simon pointed out that the volume and resonance pots were very noisy, so we agreed to take a look at them later if we had time.

Having reorganised the furniture, we took the opportunity to tidy up the wiring and to rationalise the use of mains connector blocks to minimise visible wiring. The Behringer mixer was also rewired so that the monitoring output from the MOTU 828 MkII came in via the two-track returns. By pressing the Two-track To Control Room switch, the master section of the mixer could be used as a monitoring-level control while leaving the input channels free to be used for recording or rehearsal at any time. However, if Simon wishes to keep the Sony monitors in the system, a dedicated monitor controller with speaker switching would be more practical. We found room for the G5 under the right-hand side of the desk which reduced the noise to the point where it was negligible. To our surprise, it still picked up a strong enough signal for the Airport system to continue working reliably.

Testing, Testing...

With all the wiring sorted and all the equipment placed ergonomically, we came to test the system, but we were greeted by no sound other than some high-pitched whistles and a slight hint of instability coming from the speakers. Earlier we had disconnected a pair of optical cables connecting the Mac and MOTU, once we'd explained to Simon that the

The completely overhauled studio setup, complete with acoustic treatment.

Mac's S/PDIF interface isn't compatible with the ADAT format used by the MOTU 828 MkII — and Paul guessed the cause of the problem right away having come across it on his own system some weeks before. For some reason, the current Mac OS and current MOTU drivers only seem happy when you switch on the ADAT optical inputs and outputs in the MOTU control panel, even if you aren't using them. As soon as we did this everything sprang to life.

Using Hugh's very familiar BBC test CD, the monitors sounded clean, but a little bass light. Fitting Auralex Mo Pads beneath the speakers helped firm up the bass a little and we also used the Mo Pads' in-built angle adjustment function to ensure the tweeters were pointing directly at Simon's head when he was seated in his normal mixing position.

To help Simon get off to a quick start Paul set up a default Song file template for him with the three external synths coming in via the extra MOTU 828 MkII inputs to Aux Audio objects within *Logic*. As the Deep Bass 9 drives the Mono Poly, both synths play at once when addressed, but by using Aux objects the unwanted synth can easily be muted. It is a simple matter to route either synth to an audio track for recording when the time comes. The new default template comprised 12 audio tracks, 12 instrument tracks, individual tracks for the Deep Bass 9 and Mono Poly, plus 16 further tracks for the 16 parts of the Triton. Paul also managed to find an Environment object on-line with all the Triton factory patches already entered, although it transpired that Simon had changed all the factory sounds, so we didn't actually get around to using it!

Screen sets were then created so that Simon could quickly access his most-needed screens or screen combinations on the two monitors, and Paul also set the Preferences to open the Event List when double-clicking on an object in the Arrange page, as this is the window Simon uses most for editing. Two post-fade send busses were set up to feed a *Space Designer* Reverb and a 'tape' delay, respectively, while Simon's sampled Bösendorfer piano was set up on one of the virtual instrument tracks so he'd always have a piano to work with as soon as the Song loaded. Simon tried out the new layout and agreed that it was much more comfortable to work with, and the newly created default template got him up and running with the bare minimum of fuss.

Cleaning Crackly Pots

That just left his crackly Mono Poly to investigate. He'd already bought some suitable pot-cleaning spray, but he wasn't sure how best to get it into the pots. Access to the inside of the Mono Poly was gained by unfastening a handful of cross-head screws, after which the whole top panel could be lifted up and tilted back. The circuit boards are fastened to the rear of the pots, but if you use the plastic pipe that comes with the cleaner, it is possible to direct the spray into the pots via the slot from which the three contact legs emerge. A couple of sprays followed by some vigorous twiddling of the offending controls soon cured the crackling.

In all we spent less than six hours at Simon's studio, but managed to pull it all apart, reassemble it, and improve the ergonomics and acoustics very significantly. Paul even got to try out his vacuum cleaner to tidy up before we did our 'after' photos! The new layout is a vast improvement on the original setup, and the wiring is largely out of sight. Simon was worried that he might have to budget for some better monitors, but having treated the room they seem fine. The stereo imaging was much improved and the bass end was reasonably even. SOS

IK Multimedia Classik Studio Reverb

Modelling Reverb Plug-ins For Mac OS X & Windows

The convolution revolution has left a gap in the market for a high-quality reverb based on more traditional room-modelling techniques. IK's suite of plug-ins looks to fill the void.

Paul White

Though there are some extremely good hardware and DSP-card assisted digital reverbs available to the project studio owner, there are relatively few really good host-powered reverb plug-ins other than those based on convolution. Convolution is a fantastic tool for recreating real room sounds or snapshots of hardware reverb patches, but it has limited editability and tends to use a lot more computing power than traditional synthetic reverbs. IK Multimedia set out to address this shortcoming by designing *Classik Studio Reverb*, which is a suite of four synthetic reverb plug-ins dedicated to Plate, Hall, Room and Inverse effects. This is a different approach from putting all the reverb models into a single plug-in, but it has the advantage of making the individual modules easier to use, as there are no redundant controls. Though the reverb isn't marketed as being based on any specific hardware device, IK Multimedia have clearly taken note of the classic reverb treatments we all know and love in designing their algorithms.

CSR is authorised by means of the included USB key, which is similar to those used by BIAS and Steinberg. Licences may be transferred between compatible keys where necessary. The plug-in suite is available for both Mac and Windows users and supports the RTAS, VST and AU plug-in formats. Mac users need to be running OS X, while Windows users need to be working with a version of Windows XP.

On The Outside

I've owned hardware reverbs that have around 100 editable parameters per patch, and as Lexicon did with some of their more advanced reverbs, *CSR* has been designed for ease of use by giving the user a fairly simple interface augmented by the option of more advanced editing features 'under the hood'. These remain hidden in Easy mode but are available to those who wish to explore further in Advanced mode. By entering Advanced mode, you can indeed twiddle 100 parameters including a very flexible modulation matrix, LFO section and

Alternatives

In terms of sound and features, the most obvious competition to *CSR* comes from DSP-assisted reverbs such as TC's *Classicverb* Powercore plug-in and Digidesign's *Revibe* for HD Accel systems. In the native universe, Pro Tools users could consider Sony's *Oxford Reverb* (reviewed in *SOS* March 2005); Waves' *Trueverb* is another option, though it's now rather long in the tooth, while Wizoo's *W2* (reviewed in February this year) combines algorithmic and convolution reverbs in one plug-in. Some sequencers also come bundled with good-quality algorithmic reverbs, such as the Lexicon *Pantheon* plug-in supplied with *Sonar Producer*.

Classik Studio Reverb's four component plug-ins: *Classik Hall* (left), *Classik Room* and *Classik Plate* in Easy mode (above), and *Classik Plate* (right) showing some of the controls available in Advanced mode.

configurable Macro controls. The modulation section is useful in emulating those hardware reverb patches that include some element of parameter modulation within the algorithm to smooth out the reverb sound and create a sense of movement. However, if you just need a classy-sounding reverb with the ability to edit the essentials, such as decay time, brightness and so on, you can stick with Easy mode and still get great results.

As we now seem to take for granted in the plug-in world, *CSR* offers full on-screen editing control and, where the host supports it, control automation. The reverb types offered attempt to cover pretty much all bases from small rooms and ambiences, via plates, to large halls and chambers. Additionally, the Inverse module can serve up all the classic gated and reverse treatments. Whether you're doing rock, pop, folk or classical music, there's a reverb designed to fit the situation.

Reverb Colours

Each of the four algorithm types opens with a similar user interface but all are coloured differently for identification. The main controls are located towards the centre of the front panel, with the Macro sliders off to the right. The idea behind these is that you can arrange for up to eight parameters to be changed simultaneously using one slider so, for example, you could easily set up a reverb that gets brighter as it gets shorter. You could also create your own 'distance' control where one slider would increase pre-delay, darken the reverb tail and increase the diffusion.

At the bottom of the panel are dedicated buttons to switch between Easy and Advanced modes. Once Advanced mode has been activated, you see more buttons that provide access to extra pages dedicated to specific areas of the effects, such as reverb, echo, colour, reflections, modulation and macros. There's also an A/B compare function so that you can quickly switch back and forth between a stored effect and your new edits to that effect. All the parameters show up as pseudo-LED numeric displays above the controls, and there's a good selection of factory presets to get you started. I'm not a fan of presets for things like compressors and equalisers, but presets are really very useful in a reverb plug-in, and are easy to fine-tune to your own needs in this case.

The modulation matrix is a bit more advanced; in essence, up to eight modulation sources can modulate up to eight destinations, allowing the character of the reverb to be shaped by LFOs or by an envelope. In addition to allowing the user to set up audibly swirly reverbs or ones that get shorter or darker in response to a modulation source, the two envelope generators enable the shape or tonal character of the reverb decay to be modified in a number of creative ways, though they only provide attack and release controls, and there's no graphical envelope display.

Sight And Sound

I was very impressed with these reverbs, not just because they sound convincing, but more importantly, because they also integrate well with the dry sound they are treating. The Plate algorithm gives away its synthetic nature if you extend the decay too far, as the reverb tail takes on a slightly cyclic tonality, but used at sensible decay times it sounds absolutely fine with just the right amount of metallic coloration. Perhaps the most impressive module is the one dedicated to Room reverbs, as this has the ability to create some very realistic, short-decay ambiences that work well on drums, acoustic guitars and so on. The Inverse section offers nothing radically new, though it also includes a reverse tape effect, but it covers all the normal requirements with plenty of adjustment range.

Operationally, the Easy mode is extremely straightforward, yet the Advanced mode isn't at all scary once you know what the extra parameters actually do. Even setting up macros and modulation is a piece of cake. I think it was unnecessary to lock pre-delay away with the Advanced features, but other than that, the designers have got the balance about right. I also noticed that very little processor overhead is taken up in comparison with my convolution reverbs, so using two or three instances of *CSR* will add negligible load to a modern system. SOS

SOUND ON SOUND

IK Multimedia
Classik Studio Reverb **£229**

pros
- Dense, convincing sound.
- Easy and advanced editing modes.
- Macros and modulation matrix add to the flexibility.
- Low CPU overhead.

cons
- Perhaps more graphical elements could have been used to help inexperienced users visualise the changes their edits make.

summary
Like most IK Multimedia products, *CSR* takes a workmanlike approach with no unnecessary glamour or frills, but it gets the job done with style. It's also light on your CPU.

information

£	£229 including VAT.
T	IK Multimedia UK +44 (0)1223 234414.
F	+44 (0)7005 968006.
E	uk.sales@ikmultimedia.com
W	www.ikmultimedia.com

Win Focusrite

Focusrite has been a name to reckon with in the audio industry for more than a quarter of a century, famous for products including high-end mixing consoles, top-notch outboard processors, acclaimed preamps such as the ISA430 and, recently, the Saffire and Saffire LE computer audio interfaces. But alongside their own equipment, Focusrite also distribute complementary products from Novation and KRK, and this month they've dipped into their entire range to offer an amazing bundle of prizes to the lucky winner of this latest *Sound On Sound* competition.

The centrepiece of this highly desirable setup is the new Focusrite Liquid Mix processor (pictured above), worth £499, aided and abetted by a Focusrite Octopre LE eight-channel preamp, worth £349, complete with its £129 digital option card. Synth specialist Novation's new Remote Zero SL USB hardware controller, worth £229, will also find its way to the winner's studio. Finally, to help the winner hear in its full glory the sonic magic that this great kit will no doubt

questions

A pair of which KRK monitors is up for grabs in this month's competition?

a. Rokit 5 ☐
b. Rokit 6 ☐
c. Rokit 8 ☐
d. Rokit lolly ☐

What product have Focusrite NOT made?

a. Mixing consoles
b. Preamps ☐
c. Guitars ☐
d. Audio interfaces ☐

How much does the Focusrite Liquid Mix usually cost?

a. £599 ☐
b. £1999 ☐
c. £14.99 ☐
d. £499 ☐

Focusrite tie-breaker

One of this month's prizes is a pair of KRK Rokit monitors. If you could build a real rocket, where would you go and why? Answers in 30 words or less, please.

.....

.....

.....

.....

.....

Name .....

Address .....

.....

Daytime tel. no: .....

Email: .....

Would you like to receive more information on Focusrite products? If yes, please tick or cross this box. ☐

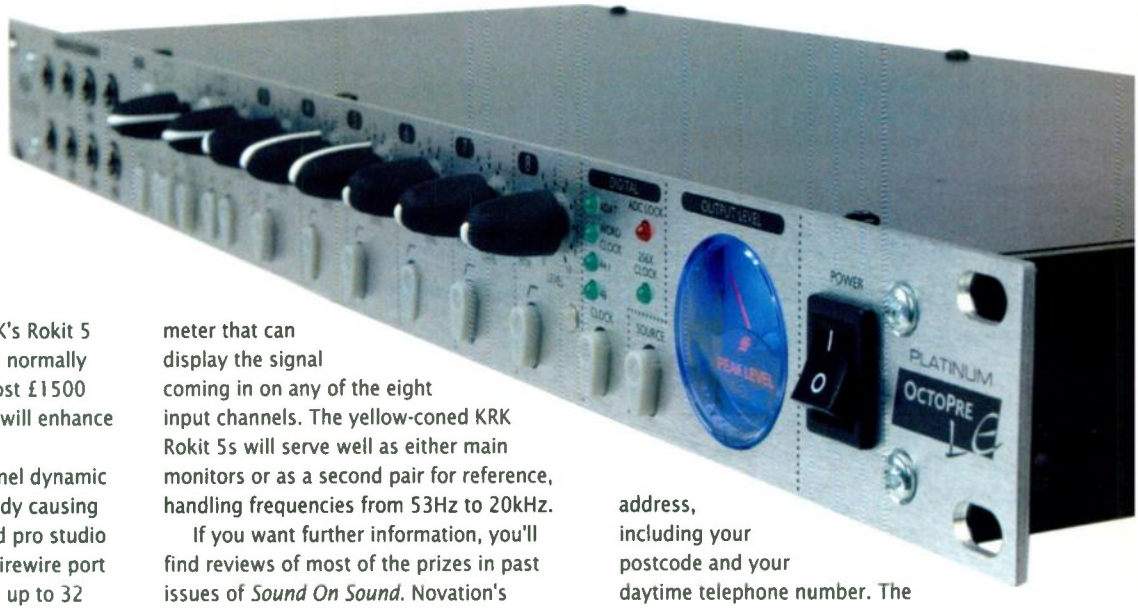
Post your completed entry to: Focusrite Competition August 2006, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge CB3 8SQ, England.

the small print

1. Only one entry per person is permitted. 2. Employees of SOS Publications Ltd, Focusrite, Novation and 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.

studio bundle

Worth £1465



be working, there's a pair of KRK's Rokit 5 active nearfield monitors, which normally cost £258. It all adds up to almost £1500 worth of cutting-edge gear that will enhance almost any studio.

The Liquid Mix, a multi-channel dynamic convolution DSP engine, is already causing quite a stir among the home and pro studio markets. Simply plug it in to a Firewire port on your computer and you have up to 32 channels of dynamics and EQ processing, offering the signature sounds of a range of classic studio processors, in a neat desktop unit that gives hands-on control over every parameter. Continuing the hands-on control theme is Novation's Remote Zero SL, a dedicated hardware controller that offers 16 assignable knobs, eight faders, transport control and useful LCD displays. It connects to a Mac or PC using USB but also has standard MIDI ports for use with other MIDI equipment. The included *Automap* software — which supports *Cubase SX 3* and *Nuendo 3*, *Live 5*, *Logic Pro 7* and *Reason 3* — maps the functions of the software currently selected to the control surface.

To help you get the best quality signals into the computer, there's the Octopre LE, featuring eight top-notch Focusrite preamps, plus ADAT I/O. It has mic, line and instrument inputs, plus an attractive blue meter that can display the signal coming in on any of the eight input channels. The yellow-coned KRK Rokit 5s will serve well as either main monitors or as a second pair for reference, handling frequencies from 53Hz to 20kHz.

If you want further information, you'll find reviews of most of the prizes in past issues of *Sound On Sound*. Novation's Remote Zero SL is so new that it hasn't been reviewed yet, but we featured it in the news section of *SOS* June 2006, while the Liquid Mix was previewed in *SOS* July 2006 (www.soundonsound.com/sos/jul06/articles/liquidmix.htm). We reviewed the Octopre LE in *SOS* July 2005 (www.soundonsound.com/sos/jul05/articles/octoprele.htm) and the Rokit 5 monitors appeared alongside the Rokit 8 in the August 2004 issue of *Sound On Sound* (www.soundonsound.com/sos/aug04/articles/krkrokit.htm).

For a chance of winning these great prizes, simply fill out the entry form and post it to the address on the coupon. Alternatively, enter via the electronic form on the *Sound On Sound* web site, www.soundonsound.com. Please make sure you answer all the questions and complete the tie-breaker. We also require your full address, including your postcode and your daytime telephone number. The closing date for entries is 31st September, 2006. SOS

Prize kindly donated by Focusrite
T +44 (0)1494 462246
W www.focusrite.com

CEDAR Audio *Retouch* winners

Congratulations to Lisette Davidson and Peter Medland, who both won a copy of CEDAR Audio's *Retouch* software after entering our competition in *SOS* March 2006. Lisette (pictured) commented "I have been pretty high all week, not least when I discovered how much the software was worth! Not that I'm materialistic, you understand". Thanks again to our friends at CEDAR for supplying those fantastic prizes.

Creamware B4000 ASB

Modelled Tonewheel Organ

Having served up convincing hardware emulations of the legendary Minimoog and Prophet V analogue synths, Creamware turn their attention to the mighty Hammond B3.

Dave Stewart

The past is all around us, and it won't go away. Virtually every popular keyboard from the '60s and '70s seems today to be available in software, sampled, or emulated hardware form — Vox Continental, Farfisa, and Hammond organs; Fender Rhodes and Wurlitzer electric pianos; Mellotron; Hohner Clavinet and Pianet; Yamaha CP70 electric grand; Minimoog, Moog Modular, ARP Odyssey, ARP 2600, Yamaha CS80, Prophet V, and PPG Wave synths; and even the daftly-named, UK-designed Oscar!

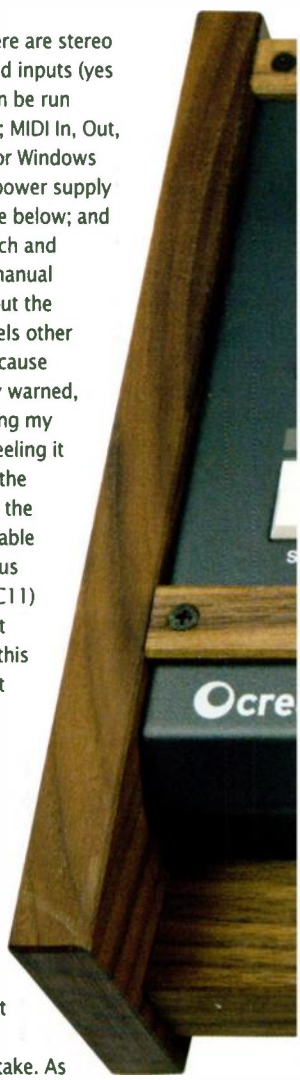
Having played most of these vintage items in their heyday, I well remember that they required regular and costly maintenance, tuning, and repair, and, as my shredded back muscles will testify, some of them weighed a ton. While I'm grateful to be spared the bills and the backache, I'm still suspicious that behind the flashy graphics and hip retro stylings of the modern replicants, the software-driven sound engine might not be doing justice to the original instrument.

On that score, German company Creamware recently allayed a few of my fears with their Pro 12 ASB, a very creditable hardware recreation of Sequential Circuits' fabulous 1978 Prophet V analogue synth. Apparently their Minimax ASB Minimoog emulation does the business too. (You can read the reviews in *SOS* May and January 2006 respectively.) So when I heard that Creamware were planning a digital clone of the mighty Hammond B3 organ, my ears pricked up. This instrument towers over the 20th-century music scene like a colossus: since it first appeared in 1935, it's been played by legions of legendary organists, is still widely used by touring bands, and can be found in recording studios throughout the world. The big, rich, churning, but incisive sound we all know comes from hooking the B3 up to a Leslie rotary-speaker cabinet, which means that anyone who attempts to emulate the famous Hammond sound has got to get the Leslie effect right as well.

Economy Of Scale

Creamware's B4000 ASB (short for Authentic Sound Box) is a sturdy, well-built hardware unit in a wooden case which, along with its various front-panel controls, boasts a set of nine full-sized Hammond-style drawbars which feel smooth and solid in operation. It must be said that Creamware do the miniaturisation thing very well — from a distance, their petite instrument (32.5 x 21 x 9.5cm) resembles a little 1970s rhythm box, and it's only on coming closer that one notices the set of drawbars on the front. If you placed it side by side with a real B3 (a huge, intimidating beast with two manuals and a large, two-octave pedal board) it would be like seeing a kitten standing next to a baby elephant. But small is beautiful — unlike the original, you can take this lightweight box to a gig in a shoulder bag, and if the gig happens to be in Yokohama or Oklahoma you won't be looking at an excess-baggage bill of thousands of pounds.

On the back panel there are stereo analogue jack outputs and inputs (yes folks, external signals can be run through the Leslie effect); MIDI In, Out, and Thru; a USB socket for Windows and Mac connectivity; a power supply socket, about which more below; and two jack sockets for switch and expression pedals. The manual makes a terrible fuss about the latter, insisting that models other than a Yamaha FC7 may cause permanent damage. Duly warned, I refrained from connecting my Korg expression pedal, feeling it might be better to finish the review before destroying the unit. I also figured I'd be able to use the MIDI Continuous Controller number 11 (CC11) Expression controller, but the B4000's response to this control is inverted so that high controller values make the signal quieter! Until Creamware fix this fault you'll have to use MIDI CC7 (Volume) instead. I also discovered that the B4000 ignores sustain-pedal and volume-control changes on all MIDI channels apart from channel one, which I presume is another mistake. As



SOUND ON SOUND

Creamware B4000 ASB £749

pros

- A well-designed, solidly built Hammond clone with a good-quality, full-size set of drawbars.
- The sound is authentic and replicates most characteristics of the Hammond B3.
- The Leslie effect is faithfully reproduced and sounds very nice.
- Creamware's editing software offers a high degree of control over the various Leslie settings.
- Fairly priced in the UK.

cons

- The UK mains adaptor supplied with the unit is rather flimsy and, when attached, very clunky.
- Various MIDI implementation problems are a handicap.
- The 'key click' effect is a little weak and varies in volume.
- There's currently no way of changing the Leslie speed over MIDI.
- No digital audio output.

summary

An authentic Hammond sound with real drawbars and a lush Leslie effect at a good price — what more could you want? Fewer niggly faults than I found in the review unit for a start, but once those are fixed the B4000 ASB will be set to take on the world.

for the 'switch' socket, I've no idea what it's for and the manual gave no clue. I tried plugging my sustain pedal into it — it didn't work, but caused no apparent harm.

Drastic Plastic

Unfortunately, the B4000's power supply is the wall-wart type with a European two-pin plug. When I reviewed Creamware's Pro 12 ASB in *SOS* May 2006 I criticised the cheap-looking, thin brown plastic UK mains adaptor supplied with it — the main problem was that it fitted too loosely over the power supply so that when plugged into a wall socket, the power supply was prone to fall out. Creamware now supply a virtually identical black plastic adaptor with a locking screw which holds it in place when tightened. However, the new adaptor is of no better quality than the original, and when fitted over the European plug at right angles it forms an ungainly and unbalanced L-shape. If I were taking the unit on the road I would definitely look for a sturdier, less clunky alternative — the best solution would be to buy your own UK 12V power supply, but it's annoying that Creamware don't supply this in the first place.

"The basic sound is very clean and pure and, either by accident or design, nicely reproduces the little 'patting' noise Hammond organs make when you release a key."

A more marked improvement over the Pro 12 is that the B4000's manual appears to have been properly proofread, and its English version is mercifully free of German technical words!

A Hammond B3's upper and lower manuals and pedals each have their own set of drawbars, so it's possible to have three different sounds going on at the same time. The B4000 has the same capability: for multitimbral work you can set a different MIDI channel for the three components, or to play all three simultaneously from a single keyboard, you can set them to the same channel and create a three-way keyboard split. The upper manual's MIDI channel (which doubles as the B4000's global channel) is not stored as part of a preset, and its last setting is not always retained on power-up. The volume controls for the manual, pedals and audio input are always 'live' and their settings are not stored within presets.

The unit ships with 127 'global presets' (each comprising the settings for both manuals and the pedals) which can be edited and overwritten. In addition, the manuals and pedals each have 100 editable user drawbar settings. Up to five presets can be assigned to the five 'encoder buttons' and instantly called up — the same goes for the drawbar settings. This flexible system makes the B4000 suitable for live use, and the good news is that sound changes occur instantly, although (as with most sound modules) a preset change tends to cut off any notes you're holding down at the time. When you turn a panel rotary control, its value appears on screen in a nice big red LED display. To aid comparison between a stored and edited sound, a small yellow 'sound changed'

▶ LED lights up whenever you alter a stored setting. In theory the LED should go out again when you manually match the setting's value, but on the review unit this function was not working on the second and third drawbars, rendering any sound-matching impossible.

Basic Sound

The Hammond drawbar system is faithfully reproduced by the B4000 (see the table below for an explanation of drawbar pitches). The basic sound (best demonstrated by pulling out the bottom drawbar with no effects) is very clean and pure and, either by accident or design, nicely reproduces the little 'patting' noise Hammond organs make when you release a key. Real Hammonds also produce a pronounced key click on the front of notes, a sonic artifact which was first regarded as a nuisance, but gradually came to be seen as an asset. (As well as amplifying his Hammond with a bank of Leslies, Keith Emerson used to run it through a Marshall 100W stack to bring out the click!) Creamware's key click simulation is a little weak — it has its own volume control and (unlike the real thing) can be heard even when no drawbars are pulled out, but I found it a bit thin-sounding, slightly synthetic, and too quiet in relation to the drawbar volume. The click also seemed to vary in level somewhat, which occasionally made it vanish in the mix.

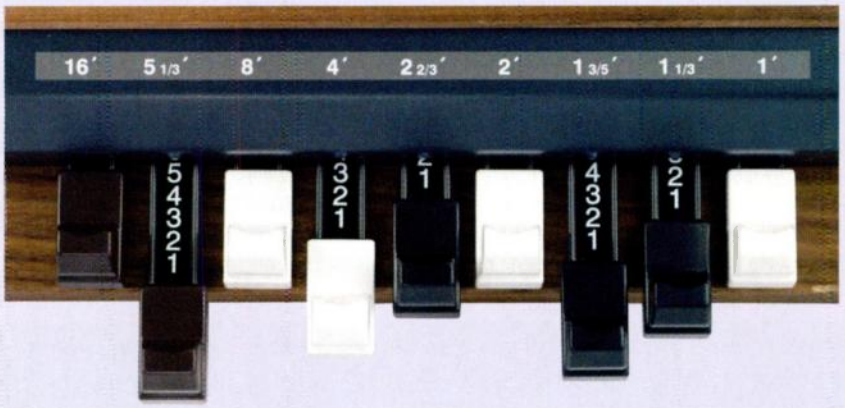
You can make the unnaturally pure organ tone less pristine by turning up the front panel 'condition' control — this successfully simulates the drawbar 'leakage' which occurs in out-of-condition Hammonds and adds upper frequencies and a bit of rumble and body to the sound. Turn the knob beyond 63, and you begin to affect the organ's tuning; setting it to maximum makes the tuning so horrid that if this were a real B3 you'd be ringing your local organ repair shop to book a service! There's also a subtle 'crunch' control which recreates the slight drawbar distortion which creeps in on old Hammonds.

I always used to steer clear of the B3's vibrato, because it reminded me of the cheesy theatre organists that held my parents' generation in a weird kind of thrall. (Hours of radio time were given over to this terrible throbbing noise — God, it was depressing!) Understandably, Creamware have ignored my silly prejudices and totally nailed the Hammond's vibrato sound, which is selected (as on the original) by a six-way wheel control offering three types of vibrato and the same number of chorus settings. For what it's worth, my favourite is Chorus 1, simply because it's the most subtle, but you can emulate the sound of the great Hammond jazz players by choosing Chorus 3. I feel Creamware have missed a trick by not providing a 'vibrato depth' setting; it's nice to be able to add just a hint of vibrato to a plain, churchy organ sound, but with the B4000 you're stuck with full vibrato, or none at all.

Hammond Drawbar Terminology

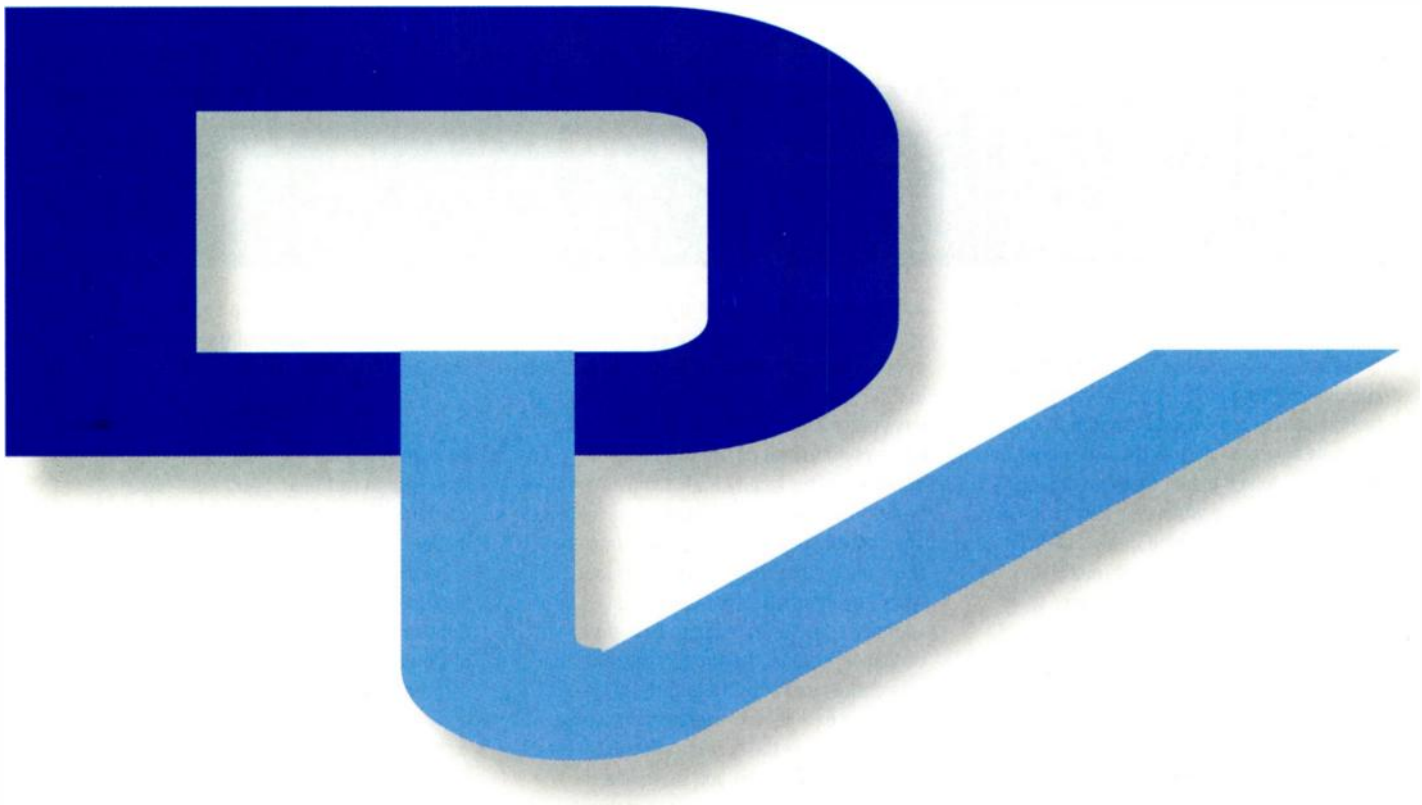
Traditionally, Hammond drawbar pitches are described in pipe-organ terminology (16', 8', and so on, referring to the pipe length in feet) or in terms of the acoustic harmonic series (second harmonic, third harmonic, and so on.) Neither seems appropriate or particularly helpful when applied to an electronic instrument, so to clarify matters the table below shows the relative musical pitches of the drawbars along with their traditional descriptions. Note that the second drawbar is pitched higher than the third, a Hammond organ idiosyncrasy.

Drawbar	Pipe Length	Harmonic	Interval	Pitch (for MIDI note number 60)
1 (Brown)	16'	1st	Sub-octave	C2
2 (Brown)	5 1/3'	3rd	Fifth	G3
3 (White)	8'	2nd	Fundamental	C3 (Middle 'C')
4 (White)	4'	4th	Octave	C4
5 (Black)	2 2/3'	6th	Octave + fifth	G4
6 (White)	2'	8th	Two octaves	C5
7 (Black)	1 3/5'	10th	Two octaves + major third	E5
8 (Black)	1 1/3'	12th	Two octaves + fifth	G5
9 (White)	1'	16th	Three octaves	C6



Percussion & Pedals

No complaints about the B4000's percussion sound — it's perfect. The default percussion pitch is two octaves above the lowest drawbar's notes, producing a pleasant, but piercing high-pitched chime which adds rhythmic definition to the Hammond sound ▶



Get More...

DIGITAL VILLAGE

www.dv247.com

and helps it compete with cymbals and guitars. Setting the percussion to '3rd' on the front panel raises the pitch by a fifth, which, in conjunction with the percussion's fast decay setting, makes the sound you hear playing the bass line on Kylie Minogue's 'Can't Get You Out Of My Head' and the current National Lottery theme tune. (Please note that I offer these examples purely for illustrative purposes, not as indicators of my musical taste...) Further percussion options can be accessed via the B4000's editing software, which I'll get to in a moment.

As you may know, Hammond percussion sounds only when notes are played with a detached fingering style, hence its original name — 'touch response' percussion. This means that once the percussion has sounded on the first note of a passage, it's up to you to lift your fingers smartly if you want to trigger the effect on every subsequent note. (Don't complain — it's good for your keyboard technique.) On tonewheel Hammonds, one of the buss-bar contacts was used to trigger the percussion, which meant that one drawbar became inoperative whenever percussion was enabled. On a Hammond B3, the 'sacrificial drawbar' is usually the highest (ninth) one; on my Hammond L122 (a smaller, spinet style organ), the eighth drawbar gets the chop. Creamware have diligently replicated the B3's behaviour by muting the ninth drawbar when percussion is activated, but while this is undoubtedly authentic there's no longer any electronic reason to do it — I would have preferred to have the option of keeping all nine drawbars sounding in percussion mode.

Alternatives

The Hammond organ's long-standing popularity has inspired many digital soundalikes — the Korg CX3, Roland VK8, and Voce Key 5 keyboards all sport Hammond-style drawbars, while Clavia's Nord Electro 2 uses a set of LED's to display virtual drawbar positions. Hammond continue to manufacture their own pricey digital clones, including the single-manual Hammond Suzuki XK3 and XK1 models. In the software domain, Apple's *EVB3* and Native Instruments' *B4 MkII* (which has its own B4D hardware drawbar controller) carefully replicate the Hammond/Leslie sound. Ultimate Sound Bank's *Charlie* is not be sniffed at, and those operating on a zero budget can download the free VST instrument *ORGANized Trio* at www.soundfonts.it. But if you want a hardware Hammond-in-a-box equipped with real drawbars, the closest alternative to Creamware's B4000 ASB is a Roland VK8M module. Voce's V5+ unit is also a contender, but it has no Leslie rotary-speaker effect.

A real B3 has just two pedal drawbars, but the B4000 dedicates six drawbars to the creation of pedal sounds. However, since Hammond pedals are characterised by their big, booming bass (a sound so powerful that it's often been used instead of a bass player in jazz combos) you won't need the upper frequencies provided by the higher drawbars if you want to recreate this effect! The B4000 does a reasonable job of simulating that low end power, but it doesn't match the strength of a real Hammond pedal board.

Soft Leslie

The front-panel controls are sufficient for most purposes, but to access the B4000's hidden depths you need to connect it to a computer via USB or MIDI and open up the instrument's software editor utility. There you'll see individual drawbar settings for the upper manual, lower manual, and pedals, plus (on the Main page) a handy extra control that lets you set the pitch of the percussion to correspond to that of any one of the drawbars. (This facility is also implemented in Native Instruments' B4 — quite a coincidence!) The control doesn't have any markings to show the pitches, but you can work out what's going on by using your ears (a pair of auditory accessories made of flesh and cartilage, positioned symmetrically on either side of the head and DI'd into the brain — I find them excellent for stereo auditioning).

The stereo Leslie speaker effect has several useful controls that let you customise its sound beyond the rather limited settings found on a B3. The most significant of these are separate speed controls for the 'slow' and 'fast' rotation speeds — with the real thing I always found the former too slow and the latter too cheesy, so it's nice to be able to set a speed somewhere in between. Aided by the editor, it was easy to get a lovely rich, floaty chorale Leslie sound, and I appreciated the fact that you can set the rate of change between slow and fast speeds in both directions. Separate controls for the treble and bass rotors combined with Spread, Distance, and Balance settings for the two virtual mics (again duplicating features found in NI's B4 — funny, that) are also welcome. It's theoretically possible to use aftertouch to switch the Leslie speed, but this facility didn't work. This means there's no way to change Leslie speed over MIDI, which is bad news if you're planning to sequence the B4000 in a track.

Test Spec

- Creamware B4000 OS v1.29
- Creamware B4000 DSP software v1.19
- 2GHz Athlon PC with 2GB RAM running Windows XP Professional

Distortion, controlled or otherwise, has always been an important part of the Hammond sound, and the edgy sound of an over-driven Leslie is a classic, exciting rock-organ timbre. Creamware say that their version of this is 'faithfully digitally modelled on the sound of an original overdriven Leslie cabinet amplifier', and although it has no controls beyond basic level and balance, it does sound pretty lifelike. However, if you want to explore a wider range of distortion timbres for your Hammond sound, you'd be advised to look at some of the excellent amp simulation plug-ins out there.

End Note

By giving me knobs to fiddle with and drawbars to pull in and out, the B4000 unchains me from my computer and makes me feel like a musician again. With this piece of hardware, Creamware have got the important things (not least, the basic sound) right, and have also done a good, faithful job of simulating the various characteristics and eccentricities of a real B3 — even, in the name of historical accuracy, the undesirable ones!

On the downside, the numerous niggly faults I found in the review unit somewhat take the shine off this product. The box has the potential to establish itself as a viable Hammond clone, but until Creamware sort out these bugs (none of which, I have to say, are particularly serious) that potential won't be fully realised. I look forward to these problems being fixed, at which point I'll be able to recommend the B4000 ASB to my fellow keyboardists as a truly desirable piece of professional kit, fit to uphold the reputation of Laurens Hammond's immortal invention. SOS

information

£ £749 including VAT.
T Sonic8 +44 (0)8701 657456.
F +44 (0)8701 657458.
E info@sonic8.com
W www.sonic8.com
W www.creamware.de

...Much More

pricebeat@digitalvillage.co.uk

We guarantee the lowest prices in Europe. However, if you find a more competitive price from another UK or European authorised retailer* within 7 days of purchase, we will refund the difference.

Same Low Prices In-store, On-line and Mail-order

Unlike some UK retailers, DV are proud to offer the same low prices in-store, on-line and via telephone mail-order. Why pay more in-store?

The UK's Largest Chain of Pro Audio Stores

Unlike some UK retailers, we specialise in Pro Audio. Visit a DV store and find out why DV are the UK's leading Pro Tools dealer and the UK's No.1 supplier of professional recording equipment.

DVPC & Apple Systems

For your ultimate computer system, contact a DV store for friendly and unbiased advice. Digital Village are Apple's No.1 Audio Solution Expert.

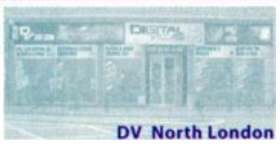
25 Years of Experience

Digital Village has been trading for over 25 years - depend on us for expert advice when configuring your studio system.

Latest Deals and Advice

Sign up to receive the latest deals and product advice via e-mail and you could win a year at the SAE Institute in Australia (prize includes tuition fees, flights and accommodation - worth over £10,000). Contact your nearest DV store or visit www.dv247.com for more details.

Why buy from anyone else?



DIGITAL VILLAGE

www.dv247.com

Southampton
023 8023 3444 3 Kingsgate Centre
St. Mary Street S'hampton SO14 1NR
southampton@digitalvillage.co.uk

Bristol
0117 946 7700 21 The Mall
Clifton Village Bristol BS8 4JG
bristol@digitalvillage.co.uk

N. London
020 8440 3440 141 High Street
Barnet Herts EN5 5UZ
northlondon@digitalvillage.co.uk

S. London
020 8407 8444 562 Brighton Road
South Croydon Surrey CR2 6AW
southlondon@digitalvillage.co.uk

DV Installation
01708 771906
Fax: 01708 771901
installation@digitalvillage.co.uk

Cambridge
01223 316091 86 Mill Road
Cambridge Cambs CB1 2AS
cambridge@digitalvillage.co.uk

Birmingham
0121 687 4777 2 Dakota Buildings
James Street Birmingham B3 1SD
birmingham@digitalvillage.co.uk

E. London
020 8510 1500 10 High Road
Chadwell Heath Essex RM6 6PR
eastlondon@digitalvillage.co.uk

W. London
020 8992 5592 14 The Broadway
Gunnersbury Lane London W3 8HR
westlondon@digitalvillage.co.uk

Educational Sales
01708 771900
Fax: 01708 771901
sales@mveducation.com

*item must be in stock. Applies to regular DV stock items only. Other offers may not be available when price beating.

WRH

Cakewalk Rapture

Software Synth For Mac OS X & Windows XP

The main *Rapture* screen. Tabs bring up the next Element or the Global page, while a simple mixer runs along the bottom.

Cakewalk's latest virtual instrument takes traditional subtractive synthesis as its starting point, but offers a degree of flexibility that's unimaginable in any hardware instrument.

Alan Tubbs

Cakewalk have been making all kinds of progress on their synth front. First, they brought on René Ceballos from RGC Audio. He worked on Cakewalk's VA synths that were included in their DAW packages, and they are distributing his *Z3TA+* (reviewed in *SOS* October 2005: www.soundonsound.com/sos/oct05/articles/pluginfolder.htm). He designed the *Dimension* sample-based synth, of which the newer *Pro* version was reviewed in May's *SOS* (www.soundonsound.com/sos/may06/articles/cakewalkdimension.htm), and his latest effort is *Rapture*, which looks like the lovechild of those two soft synths. Visually, *Rapture* resembles *Dimension*, but it sounds and works more like *Z3TA+*. All three synths share the same SFZ multisample file format; this is relevant to several types of synthesis, but *Rapture* concentrates on wavetable synthesis. Not the PPG, Prophet, Waldorf or Ensoniq kind of swept wavetable, but single-cycle waveforms built from a table of data, producing a starting point for subtractive, analogue-style synthesis. Cakewalk are mighty proud of their so-called Expressive Engine and claim it is similar, in some respects, to an analogue oscillator. Be that as it may, *Rapture* doesn't look or operate like some of the more strictly retro analogue emulators, but does sound impressively free of digital artifacts.

Rapturous Reception

Many aspects of *Rapture*'s design are similar to *Dimension*, so I'll concentrate on the ways in which the two synths differ. First, there are six Elements, or voices, instead of *Dimension*'s four. Each Element remains a complete synth voice in itself, with a full complement of keyboard mapping and sample controls in the Oscillator/Editor section; DSP for filters and drive; Modulators for envelope generation, LFOs and step-sequencers; a three-band parametric EQ; and your choice of insertable effects complete with yet another filter.

There are few differences in the control mapping section: since *Rapture* concentrates on wavetable playback, some of the more sample-oriented options are missing, but a Phase amount is added, so you can twist the starting phase of a sample, and a Quality option which lets you switch between standard and high. *Rapture*, being more 'synthy', also includes a Multi function where you can choose between three, five, seven and nine unison voices per Element. These fattening oscillators are simply switched on and automatically ▶

SOUND ON SOUND

Cakewalk *Rapture* £149

pros

- An easy-to-program entry into the sound of subtractive synthesis.
- Flexible step generators offer lots of possibilities.
- Anti-aliasing engine allows you to use sample-based waveforms as a basis for analogue-style synthesis.

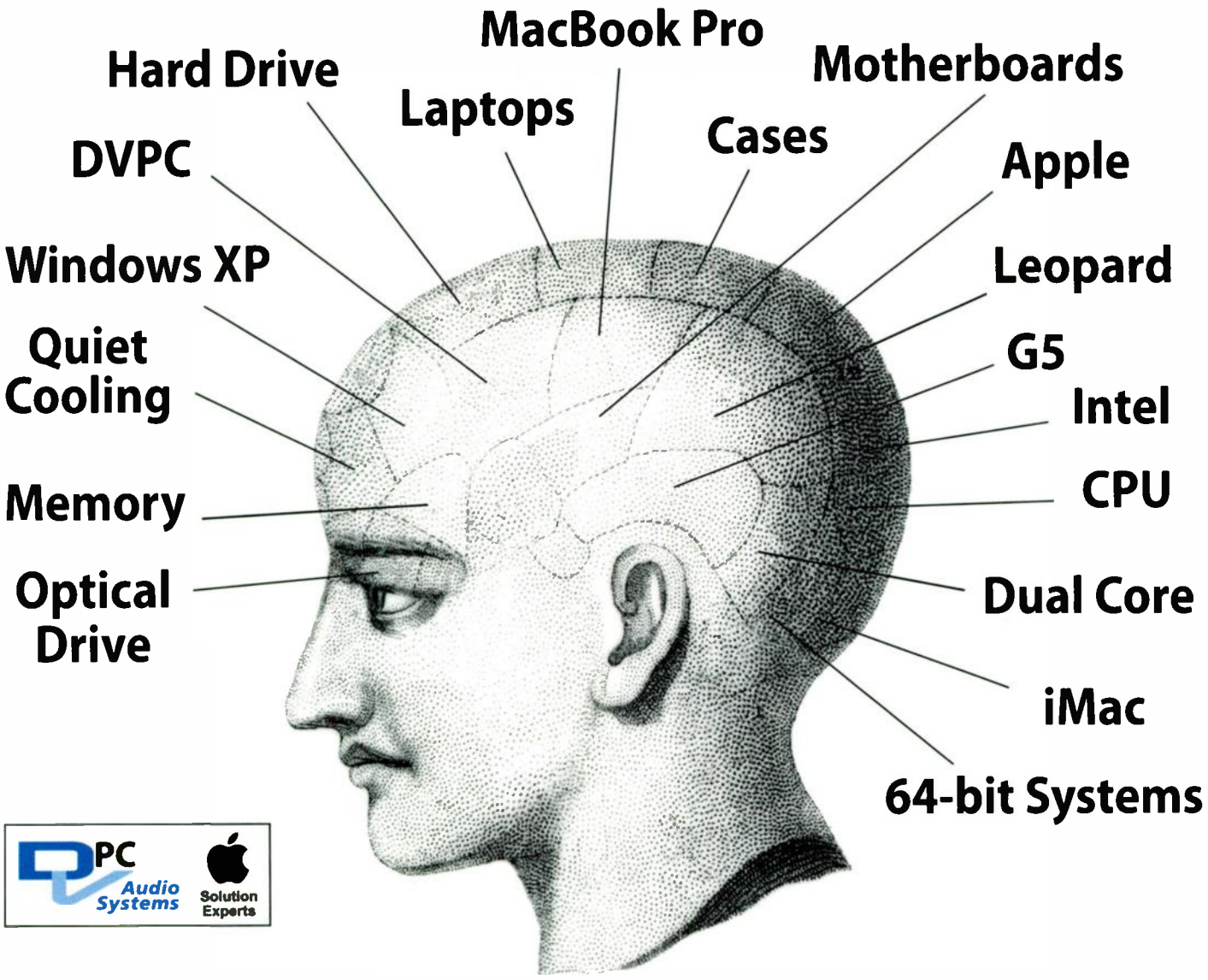
cons

- Though programming is mostly easy, there are still a few areas that could be improved.
- More global modulators could streamline the sound-creation process.

summary

Cakewalk's companion soft synth for their *Dimension* sampler looks and operates much like its cousin, but the emphasis is on synthesis, not replicating acoustic instruments. The sound is very clean, and the design strikes a nice balance between ease of use and depth of control.

Unable to buy the computer you need? Frustrated by inflexible specs?



DPC Audio Systems

Solution Experts

Don't be!
Create your PERFECT machine

online configurator
www.dv247.com

instore configurator
DIGITAL VILLAGE

▶ panned across the stereo field for extra oomph from a single Element. Think of each step as adding two extra patties (with cheese!) to your single-voice burger order. For even more obesity, there is a Detune control which operates by cents — up to 100. Many synths include a 'super-saw' setting; *Rapture* does the same sort of thing with any sample you load. And it can do it with each of the six Elements. Talk about supersizing your sonic menu.

An alternative to the Multi function is the Ring Modulation setting, which creates a second oscillator, with Detune controlling the amount of the effect. If you must have Ring Modulation and Multi, you can right-click on an Element button and choose Ring Modulate Previous Elements. This doesn't allow a single Element to do both, but uses one entire Element to ring modulate the previous ones. Most ring modulation uses a simpler system where one oscillator modulates another prior to the envelope generators, filters and so on, so creation of very rich and evolving timbres is possible with the *Rapture* method.

The final difference between *Rapture* and *Dimension* is portamento. You could always program in portamento within an SFZ file, but *Rapture* has a controller for it. Unfortunately, however, this isn't polyphonic portamento — you have to set the Polyphony to zero for it to work. Still, it is something I sorely missed in *Dimension*.

Modulation Madness

The Filter/Drive section of *Rapture* contains the same components as its counterpart in *Dimension*, but is more flexible, with two filters in series instead of *Dimension*'s single filter. The decimator, bit reducer and drive each have an on/off switch (plus drive type), and their order can now be cycled through by right-clicking. I suppose it would be more flexible to be able to drag the order and if the filters could be switched to parallel, but what you have certainly already provides a wealth of choices. The filters are very good — René has always designed very analogue-sounding filters — and there are certainly enough types. They start with low-pass, one-pole filters and progress up to high-pass, six-pole. In between there are one-pole all-pass and two-pole band-pass, band-reject and peak filters, plus 'pink' and comb filters.

The Modulation section takes up most of the screen real estate in *Rapture*, and most of the work gets done here. Keyboard shortcuts can facilitate getting around between the modules.

The Modulators can control both cutoff and resonance for each filter, as well as pitch, pan and amplitude. The envelope generators and LFO are basically taken from *Dimension*; you create envelope breakpoints by right-clicking in the graphical display, while the LFO contains all the usual controls. You can also add your own LFO shapes simply by naming mono WAV-format files consecutively from 20 to 99 and putting them in the LFO folder. These files can be of any sample rate and length and can certainly add texture to your bag of LFO tricks. However, the LFO rate control barely gets into the audio range (its fastest setting is 40Hz), so clangorous FM sounds can't be achieved this way. Keytracking works the same highly ergonomic way in *Rapture* as in *Dimension*: dragging either side of the segment maps the effect to the destination according to MIDI note numbers. There are a few tricks here, but it mostly works as you would expect.

Finally, each modulation destination has its own step sequencer in addition to the more standard EG and LFO. You can specify a number of steps from two up to 128, choose whether you want it to sync to host tempo, apply smoothing at the step boundaries and control the depth to which it modulates its destination. One trick with the latter control is to use the Ctrl key (on a PC) to move by hundreds, rather than single digits — since the range of the Depth control is in the thousands, this can save a lot of wear and tear on your wrist and mouse. Setting up a sequence is easy: not only can you 'draw' in your steps, but right-clicking on the step area brings up other choices allowing you to randomise, reverse, invert and mirror the steps. You could, for instance, come up with a pattern for the filter cutoff, copy it and paste it into the step sequencer that controls the same filter's resonance, then reverse the steps, creating an instant geometric relationship between the two. Last, but too cool to be least, you can use a keyboard to enter in MIDI notes for the pitch step sequencer.

The three-band EQ is the same as in *Dimension*, with an on switch, buttons for low-, band- and high-pass, and full parametric controls. It is, of course, great for adding that final touch to a sound. Finally, Cakewalk have added a few extra insert effects. All the *Dimension* effects are there, and *Rapture* adds more choices in the size of halls for reverbs, as well as two kinds of distortion. By default, each Element has its own EQ and effects, but if you want to use the same effects on several Elements, you can go back up to the Element buttons and, instead of choosing to ring modulate the previous Elements, choose Chain To Next Elements and share the effects. This also, of course, saves on CPU cycles. Finally, instead of hitting an Element button at the top, you can click on Global and bring up the final page. There are the two global send effects and a global step generator, plus master EQ and insert effects. The global step generator is actually two sequencers, one applied to each stereo channel's volume, so you can have lots of fun using it for auto-panning effects. ▶

Minimum System Requirements

Windows
- 1.3GHz processor, 512MB RAM, 256MB hard disk space, screen resolution 1024 x 768, 16-bit colour.
- Windows XP.

Mac
- 1.2GHz G4 processor, 512MB RAM, 256MB hard disk space, screen resolution 1024 x 768, 16-bit colour.
- Mac OS 10.3.9 or higher.

Alternatives

The world of soft synths is pretty crowded these days, although *Rapture* offers more flexibility than most. Among the innumerable other options, one that stands out is Virsyn's *Tera 3*, which now allows you to use your own samples as oscillators, and offers lots of other nice features including the additive Spectrum Oscillator.

www.scan.co.uk

t: 0870-755-4747

SCAN COMPUTERS

computer music

SCAN 3XS - ATHLON 64 Dual Core
"This Scan 3XS system is a truly high spec audio workstation. Top notch components are used throughout and it really shows."
Computer Music - December 2005

SOUND ON SOUND

SCAN 3XS - Dual-core Athlon 64 PC For Music
"This scan system turns in a stunning performance for a very reasonable price. I think we can safely assume that dual-core systems are going to be extremely popular with the PC musician, and that Scan themselves should expect plenty of customers for their 3XS audio PC systems."
Sound On Sound - January 2006

Hendrix, the true performer, in a league of his own...

reincarnated to build systems...

systems built by enthusiasts

SCAN 3XS SYSTEMS

About Us...
Scan's principal business is to build and supply specialised high performance systems and provide PC hardware/ software through mail order and the E-tail market.

Heavy investment in Scan's backend systems combined with a great work force has allowed us to achieve an award winning service level whilst still being competitive in a near perfectly competitive market.

AMD Athlon™ 64 X2 Dual-Core processor.

Satisfying your demand for more digital media.

Usher in the next generation of digital media software for amazing high defi nition video and photo editing, content creation, and audio mixing. With an AMD Athlon™ 64 X2 dual-core processor, your computing performance can improve up to 80%* on power-hungry digital media software applications.

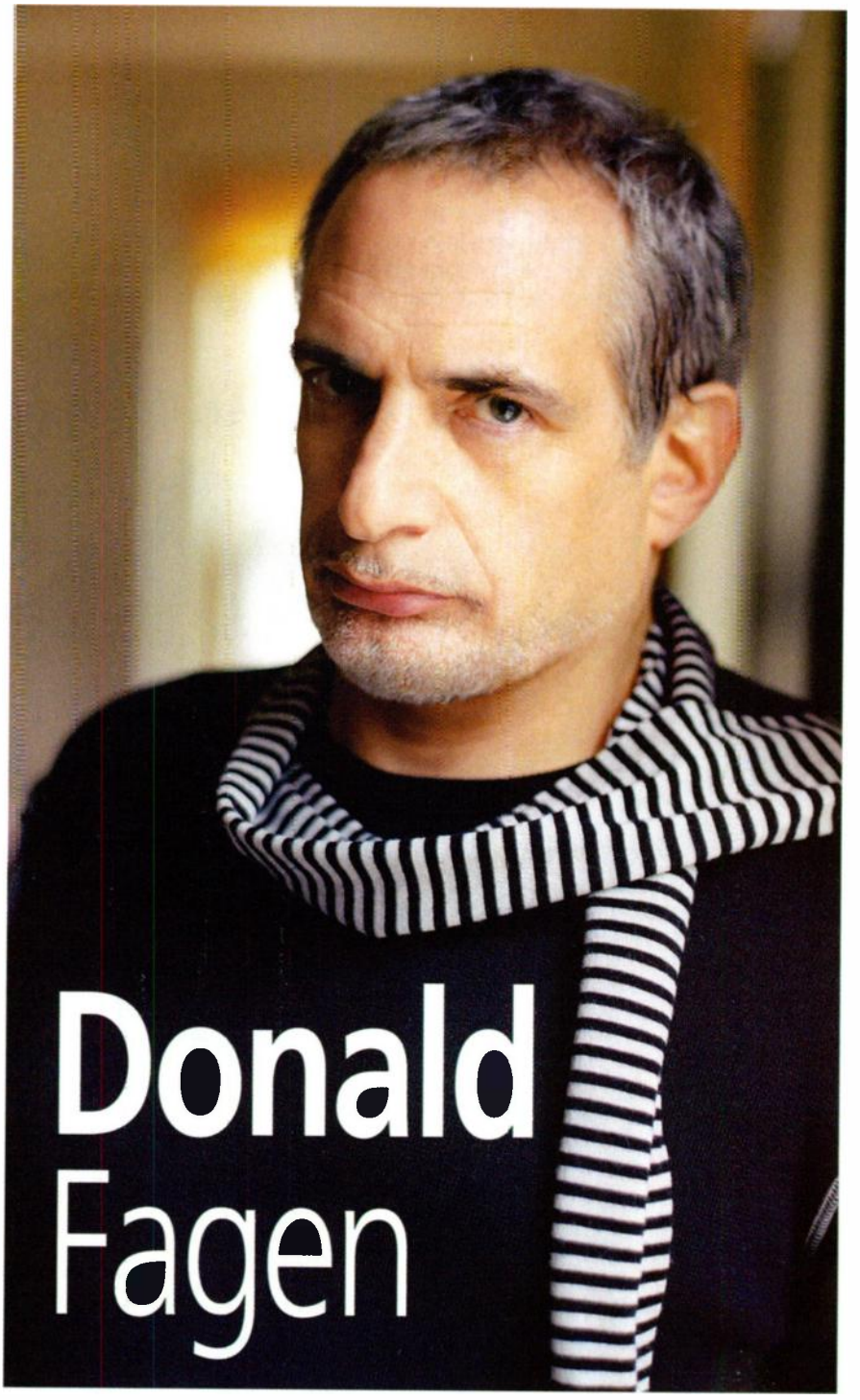
M-AUDIO
Where do you want to take your music?

Axiom 49 Advanced 49-Key Semi Weighted USB MIDI Controller

If you need total MIDI control and playability in a compact package, check out the Axiom 49. This advanced 49-key USB mobile MIDI controller delivers semi-weighted action plus assignable aftertouch, along with eight rubberized trigger pads that are perfect for hands-on drum programming and performance.

Morph The Cat, Donald Fagen's third solo album in 24 years, sees Fagen and engineer Elliott Scheiner continue their quest for the best possible sound quality — which, it seems, comes only from analogue recording.

Paul Tingen



Having wowed the music world during the 1970s with seven studio albums full of their signature hyper-intelligent mixture of rock, soul, jazz, R&B, blues and whatever else took their fancy, Steely Dan went AWOL for more than a decade in 1980. Their silence was only punctuated by Donald Fagen's best-selling 1982 album *The Nightfly*, his rather less successful *Kamakiriad* (1993), and Walter Becker's *11 Tracks Of Whack* (1994). However, the Dan duo returned to the live stage in 1993, and eventually recorded the Grammy-winning *Two Against Nature* (2000), their first studio album in two decades, followed by a second, *Everything Must Go* — the recording of which was the subject of an *SOS* article in May 2003.

And now Fagen is back on the solo path again with a new album, *Morph The Cat*. When queried about the reason for releasing a solo album at a time when Steely Dan are still standing, Fagen simply offers that he's not as good at taking holidays as Becker is. Given that Fagen lives in New York City and Becker on Kauai, Hawaii, this is, perhaps, not surprising. Fagen has also gone on record stating that his three solo albums form a trilogy, with *The Nightfly* charting the outlook of a young man, *Kamakiriad* a portrait of the artist as a middle-aged man, and *Morph The Cat* an expression of a man in the last stage of his life. (Fagen is only 58, but describes the album as an attempt to "forestall whatever there is to forestall".)

Imaginary Friends

Walter Becker was not in any way involved in the making of *Morph The Cat*, making this Fagen's first-ever DIY production, and allowing him to follow his own instincts. "I did miss Walter at times," comments Fagen, "but when I had a question I just imagined that he was there, and that worked pretty well." He was, however, aided by the familiar presence of Elliott Scheiner, a five-time Grammy-winning living legend of the American recording industry, who has worked with virtually everybody, and with Fagen since Steely Dan's fifth album, *The Royal Scam* (1976).

"Elliott is usually there for the beginning and the end," says Fagen. "He recorded the basic tracks and I used other engineers for overdubbing — which is the part of the recording that takes the longest — and he later comes in to mix the album. Elliott doesn't like to do overdubbing, because he's too impatient. He prefers to do the easy stuff! But he did a great job recording all the instruments as full as possible, which is strictly due to his expertise. I'll come into the control room during tracking and will give some comments, like 'It sounds good,' or 'The snare could be better,' or 'Perhaps

Recording *Morph The Cat*

the bass needs a dB extra at 250Hz or something.' But basically Elliott knows what I like, and we have very similar taste."

Although Steely Dan wore their love of studio technology on their sleeves, it seems that this was driven more by Walter Becker and engineer Roger Nichols than by Fagen. "Roger and Walter were always more interested in technology and in what the latest thing was," explains Fagen. "Walter's father was a hi-fi nut in the late '50s and '60s, and Walter is a science prodigy who went to Stuyvesant High School in New York, a specialist school for kids who are really good at science. I also got into high fidelity and I like good sounds, but I was never as much into the technical side of things."

Elliott Scheiner, who also worked on *The Nightfly* but not on *Kamakiriad*, agrees, but finds that things have changed. "Donald has become much more savvy as far as what takes place in the studio is concerned," he says. "He now knows what the technical issues are and what can and can't be done, whereas a dozen years ago Donald didn't know or didn't care, he just wanted to get things done. For this new album it was a great process to be working just with him, and he definitely made a lot of comments, but he is not specific about certain things. As far as EQ is concerned, he'll say 'I want a bit more top end there, or low end on the voice,' general comments like that."

Clinton Administration

The memories of engineer and artist occasionally diverge in the telling of the story of the recording *Morph The Cat*, but they agree on the basic facts. Tracking began in August 2004, at Clinton Recording Studios in New York. Fagen, second keyboard player Ted Baker, guitarists Wayne Krantz and John Herrington, Freddie Washington, drummer Keith Carlock, and Scheiner gathered there for two weeks to lay down all the backing tracks. The only difference from the *Everything Must Go* tracking line-up was that Krantz had replaced Hugh McCracken.

"I'm too lazy to look for other guys," comments Fagen. "These guys are really good anyway, and it's nice to have a band that you can communicate with. Walter and I took a long time to find this band, and I really enjoy playing with them. I'd done demos for some older songs on a computer that died [*a Mac Quadra running Opcode's* Vision *software*], and I had a hell of a time getting the information off a disk and into *Logic*. I also arranged some new songs on an Apple computer, using the *Garage Band* program, and I played these demos for the musicians. I always write on the piano, but I like to create this little mock-up in the computer.

"I'll quickly put down a bass part in *Garage Band* using a keyboard and a fake guitar using a keyboard, and the simplest basic drum beat, and it gives the musicians an idea of how the song goes. I usually also write out the keyboard part, and sometimes the bass part, though the guitar parts are worked out by the guitarists, and I'll modify what they're playing. We go back and forth until it sounds good. All kids know how to use the *Garage Band* program, and it's good to write like that, because all you need is a tiny keyboard that you plug into your computer. Keyboard magazines hate me when I say things like this, because I'm a threat to their existence. Basically I'm saying 'You don't need any of that fancy stuff.'"

"I think Donald sent CDs of the demos to the guys," elaborates Scheiner, from his studio in Connecticut, "and before we started each song he'd play the demo for the band, to refresh their memories and talk about it. The demos were just a sketchpad — I'm pretty sure there weren't any vocals on them, just a keyboard playing the melody part. He'd done a couple of rehearsals with the band before coming into the studio, but I don't think that every song was rehearsed; there were at least three or four songs that I'm pretty sure the band first heard when they came into the studio. Then, while recording them in the studio, it's a work in progress. During tracking Donald would always play with the band, calling out certain changes."

The Sound Of The Room

The backing tracks for *Morph The Cat* were recorded during a two-week period at ▶

The Decline Of Keyboards

Donald Fagen's interest in the ins and outs of recording technology might have grown in recent years, but when it comes to the tools of his trade — keyboards — the opposite is true. "From an instrument point of view, I find that the technical developments in keyboards since the '70s are not worth talking about. I experimented with all sorts of synthesizers at the time. I recall that my first synthesizer was an ARP Odyssey, which I used on the early Steely Dan records. Somebody gave me a Synergy and that had some interesting sounds that I used on *The Nightfly*.

"I don't use many synthesizers any more. Basically I'm too impatient to be using computers and synthesizers, and end up with something that's not as good as what live players do. I find that the sounds you get out of most synthesizers are basically degraded. Especially if you're playing a full-sized keyboard, there are a lot of tuning problems, with wrong harmonics that annoy me. They're also not stretch-tuned properly so that the upper notes are a little flat and the lower notes are a little sharp, so for the most part I play only tuneable instruments, like acoustic piano, Fender Rhodes, Wurlitzer and so on. I do have a Triton and a Kurzweil K2500, and use them for special effects and when I can get away with only using part of the keyboard, like if I want to mimic a flute or a mallet instrument or an organ. Organs aren't tuneable, and are flat, weird-sounding anyway."

On both Fagen's first two solo albums, *The Nightfly* and *Kamakiriad*, mention is made of the use of sampling technology, while *Morph The Cat* has none of it. What has changed? "We started using sequencing and stuff on [*Steely Dan's*] *Gaucho*," replies Fagen, "out of desperation really. We were having trouble laying down 'Hey Nineteen'. We tried it with two different bands and it still didn't work, so one of us said something like 'It's too bad that we can't get a machine to play the beat we want, with full-frequency drum sounds, and to be able to move the snare drum and kick drum around independently.' Roger [*Nichols*] replied 'I can do that.' This was back in 1978 or something, so we said 'You can do that???' To which he said 'Yes, all I need is $150,000.' So we gave him the money out of our recording budget, and six weeks later he came in with this machine and that is how it all started."

The pioneering machine was the now-legendary Wendel, reportedly based on a CompuPro S100 computer with an CPM/86 operating system. It was capable of replacing already recorded sounds and moving them around, rather than constructing a drum track from scratch. "This was in the days when digital was still very primitive," recalls Fagen. "Roger's machine did not even have any switches, it only had a regular computer keyboard and he had to type all these bytes out, huge lists of numbers, which took him 20 minutes, and at the end he would hit Return, and we heard this one snare a beat. It took so long. It got a little better during *The Nightfly*, but it was so horrible, I have tried to figure out how to get out of sampling ever since."

Roger Nichols has continued to develop his drum replacement technology, which he has now made available as a plug-in called *Wendeliser* (see www.rndigital.com).

Fagen (right) with Elliott Scheiner, who engineered *Morph The Cat* as well as previous Fagen and Steely Dan projects.

Clinton Studio A, which has a live room that's large enough to hold 85 players, with wooden floor and wooden wall-panelling ('one of the last big rooms on the East Coast' claims their web site). In this case it held six. "Everybody was set up in the same room, except for the acoustic piano," explains Scheiner. "Sometimes Donald played the piano, sometimes Ted. We built little enclosures around the guitar amps, the bass was DI'ed, as were the electronic keyboards in the room, and the drums had some low baffles with fibreglass tops and a canopy over them. We had no problems with leakage whatsoever."

According to Scheiner, he didn't do anything deliberate to achieve the more hard-hitting sound of the album. "I think a lot of the sound of the album is the room that we recorded in. I didn't set out to do anything different. I simply did what I always do, which is to capture what the musicians are playing. I seldom try to make things sound different from what is played in the recording room. I go for the assumption that the guitar player gets the sound out of his amplifier that he wants. So I go out into the studio and listen to the sounds the guys get, whether guitars or drums, and just try to get that sound. With bass it's a different story, because 50 percent or more of the time you record the bass direct. The same with electric keyboards."

Scheiner adds that he 'seldom' uses EQ during the mix, and that *Morph The Cat* was recorded via Clinton's Neve 8078 directly to analogue 24-track. Straightforward recording to analogue without much processing is now Fagen's favoured approach, says he. "It's the sound I like. It's not necessary to have the latest equipment. Today I think that I could use any studio, and any equipment, and all I need is good players and it will sound good. I like the sound of jazz records recorded in the late 1950s. I love the sound of Rudy van Gelder's records for Prestige. I can't imagine anything sounding better. Van Gelder's jazz recordings definitely influenced the Steely Dan recording and mixing style."

Three years ago Scheiner, and to a lesser degree Becker, went into fairly great detail about the analogue versus digital debate, while Fagen only let slip that he felt that

Scheiner's Mic Choices

According to Elliott Scheiner, the following mics were used on the *Morph The Cat* recording sessions.

- Kick drum: AKG D112.
- Snare: SM57 (only on top).
- Hi-hat: Neumann KM81 or 84.
- Toms: Audio-Technica ATM25.
- Overheads: Neumann U67.
- Room mics: Electrovoice RE20.
- Electric guitar: Shure SM57 right on speaker cone.
- Piano: 2x AKG C12 mics, about 12 inches from the strings.
- Trumpet and trombone: Coles ribbon.
- Tenor sax: Neumann U67.
- Baritone sax: Neumann FET47.

cakewalk

SONAR
POWER STUDIO

It's all in the box.

The most powerful, yet portable 24-bit/96kHz USB and FireWire Music Production solution packages.

SONAR Power Studio is a no-compromise combination of EDIROL's best selling audio interfaces perfectly matched with the latest Sonar music recording software. Plus, you get the extras that give you an edge in the studio. And with easy installation and auto-setup, you'll be producing professional recordings in minutes.

It's all you need. And then some.

SONAR POWER STUDIO 25
Full version of SONAR Studio Edition
Additional Roland® V-Vocal™ & AmpliTube LE
Edirol UA-25 USB Audio Interface
USB Cable
Printed User's Guide

SONAR POWER STUDIO 66
Full version of SONAR Studio Edition
Additional Roland® V-Vocal™ & AmpliTube LE
Edirol FA-66 FireWire Audio Interface
FireWire Cable
Printed User's Guide

☎ 0870 350 1515
www.edirol.co.uk

EDIROL by Roland

"digital sound loosens the fillings in your teeth". So three years on, with the dramatically fast developments in digital technology, has anything changed for Scheiner, and how are Fagen's teeth? Did they survive his work on *The Nightfly*, which was one of the first best-selling albums recorded to digital, and for years a popular demonstration record in hi-fi stores across the globe? Surely it didn't sound that bad? And what does he make of digital today?

"I haven't listened to *The Nightfly* since I made it," replied Fagen, "but the people in these hi-fi stores must have liked something about it. I think most of the way a record sounds is independent of whether it was recorded digital or analogue. So much has to do with the miking, the material, the studios, and the engineer. Having said that, I do think that digital has improved a lot over the years. It doesn't have that weird scratchy high end any more, and the bass sounds a little better too. But frankly I don't hear that much of a difference between the two media. As long as bass and drums are recorded to analogue you're OK. So we recorded the basic tracks to analogue, and for convenience's sake we loaded them into Pro Tools for overdubbing. To use analogue for overdubbing is just too much of a pain in the ass."

Many would agree with Fagen on these points, but strikingly, Scheiner's attitude appears to have hardened in the last three years. "I don't think digital will ever catch up with analogue," he says uncompromisingly. "Digital is convenient and it is good for doing trench work, but as far as sound is concerned, it's definitely analogue. I recorded the basic tracks to Quantegy GP9 tape, 15ips, +3dB operating level, Dolby SR. All edits on the backing tracks were done in analogue, and we then digitised everything, transferring stuff to Pro Tools HD at 24/96."

The Finishing Touches

The overdub sessions for *Morph The Cat* were done by TJ Doherty (who also worked on *Everything Must Go*) and Brian

Nuendo And The Rest

Given Scheiner's view that "digital will never catch up with analogue", it's a little surprising to see his studio in Connecticut filled to the brim with digital equipment, including a Yamaha DM2000 desk and MSP1 monitors, Alesis HD24, and Steinberg's *Nuendo* DAW. The producer/engineer states that he went for digital equipment because of economical reasons, but given his preferences the recent acquisition of a Studer A827 24-track was pretty much inevitable. He does, however, wax lyrical about one piece of digital equipment.

"I very much enjoy working on *Nuendo*. It's probably the best digital workstation available. It is the closest to analogue of any digital equipment I've heard. *Nuendo* sounds much better than Pro Tools, for instance. The two aren't even on the same page. I don't know why it is, I am just trusting my ears. A friend of mine, Frank Filipetti, did a comparison test between Euphonix, Pro Tools and *Nuendo*. It was all digital, there was no analogue-to-digital conversion, and Pro Tools did not even compare to the other two.

"I work on Pro Tools in commercial recording studios, but that's because I don't have a choice. I owned a Pro Tools system when it first came out and found the company impossible to deal with. They never returned any phone calls, if you had a problem there was no help. I spent more time rebooting the system than anything else. This is not necessarily Digidesign's fault, it was probably a Mac problem, but as a result I stopped using digital for a few years, until I got into the *Nuendo* system in 1998 or 99. I don't know whether it is Steinberg, or the combination of *Nuendo* and the PC I run it on, but so far it has never crashed."

Montgomery, at Avatar Studios and Sear Sounds in NYC, and Sugar Sound in Kauai, Hawaii, when Fagen visited Becker. Instruments overdubbed included guitar solos, vibes/marimba, harmonica, Fagen's lead vocals, backing vocals, and percussion. At Avatar Scheiner then overdubbed the horns in June and November of 2005, just before mixing, also at Avatar.

"Donald had done his other overdubs there," comments the engineer, "and was comfortable there. They have a Neve VR and all the analogue equipment that we needed, so it made sense to mix there. The mixes went from Pro Tools through the Neve VR and then to two-track half-inch analogue. Since I hardly use EQ during mixing, any EQ that I do will be applied during the mix. So I did some EQ-ing, and added some reverb, using the EMT 140 plates at Avatar, and also a Lexicon 480 and a TC3000. I only used room mics on the drums. I also used a Fairchild 670 on the bass and kick drum. There was no compression during recording."

It appears that mixing is a plug-in-free zone for Scheiner. "Most plug-ins are a joke," he says. "I don't think there's a reverb plug-in that comes close to an EMT 140. A plug-in that claims to make your SM57 sound like a C12 is a load of shit. It just doesn't. The only plug-ins that I use are the Universal Audio emulations of the 1176 and LA2As. And the Nuendo EQ is incredible, I love it."

"The mix was pretty much straight ahead," continues Fagen. "There are a few effects, but overall a little bit of reverb is basically it. When we are mixing, I will go out of the room, and Elliott will set up a basic track mix. I will then come in and do some serious alterations. I usually listen to the mix very carefully. I'll start with the bass and drums. My work is mainly to do with level adjustment, like the balance between kick drum and snare drum. The thing that I'm good at is the balance of instruments, so I will do a lot of that and also adjust vocals EQs, things like that."

Scheiner's version of events complements Fagen's, but there are some differences, suggesting that the engineer should, perhaps, watch his back. "I will normally come in during the morning," says Scheiner, "and get the mix to the point to where I like it. Donald then comes in and he will want to listen to the drums and then the bass and the relationship between them. Maybe the snare needs to be a bit fatter or louder, whatever. We go through this procedure and get a mix going. Donald is pretty focused when we mix, but he never touches the faders. Number one, I won't let him touch the faders, and number two, he knows that he may be deleting information, so he won't want to touch them. If he was actually sitting at the console, and wanted some low end taken off, I don't think he'd know what to reach for."

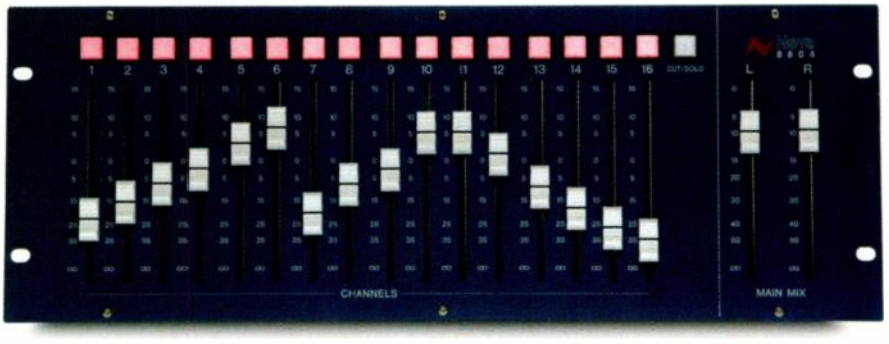
Is it possible that Fagen sneaks into the control room while Scheiner takes breaks? Whatever way they worked, the end result, *Morph The Cat*, kicks up a storm. So with Fagen in such barnstorming form these days, also have embarked on his first-ever solo tour, will the wait until his next solo album be relatively brief, or will the gap be as long as that between his previous solo efforts, fast-forwarding us to 2019 or thereabouts? "Probably," laughs the New Yorker, "but whenever it will be, it will be something completely new and it won't be part of this trilogy." SOS

mixed with the main stereo buss. This is quite separate from the main buss insert, point which functions in the traditional way. The Insert Mix path effectively runs in parallel with the main insert point, the return being mixed with the normal path at a level controlled by the Insert Mix Return control under the button (again, spanning a range from off to +10dB). This facility allows additional signals to be mixed into the final output, or for parallel compression to be set up, for example.

There are two more illuminated buttons under the right-hand meter, the first of which switches in the normal mix buss insert return, and lights yellow to indicate the active insert. However, the insert return can also be operated in a second mode which incorporates sum and difference amplifiers before and after the insert point, changing the insert signal to the Middle and Sides format instead of conventional stereo. This alternative mode is achieved by pressing in the blue Mix level rotary control under the button, at which point the insert button lights red. The Mix level knob is effectively the main output fader, ranging from off to 0dB. If the optional fader pack is connected this control is bypassed completely and only the switch function still operates.

The next button is labelled '<W>' and switches a 'spatialiser' circuit into the signal path (post the mix level control). Once activated, the stereo width can be adjusted with the red knob under the button, and this facility allows the image width to be adjusted from mono through to normal stereo and then on to extra-wide stereo.

The last control knob on the panel is the monitoring level control, and this is accompanied by four LEDs immediately above. With no LEDs lit the monitoring signal is the main stereo output, but pressing the monitor level control cycles through the alternative monitoring sources, starting with the two-track return, then channels one and two (pre-fader), and then the iMon front-panel mini-jack input, before returning to the main stereo output. By pressing the stereo width knob, the stereo mix is combined with the selected monitor source, allowing the internal and an external sources to be auditioned simultaneously. This mode is indicated with a yellow LED labelled 'Sum'.



The 8804 fader pack provides 18 long-throw faders and 16 aux outputs.

Metering

Two VU meters are fed from the main outputs. The metering is as complex as everything else on this unit, with multiple scales and built-in LEDs. The latter flash red if the signal reaches +25dBu, which is 1dB below the analogue clip level, and yellow if the optional A-D converter clips. Sensibly, the A-D clip warning takes priority over the analogue clip indication. The meter ballistics are based on a pseudo-PPM format, but with a unique white-on-black scale marked from 0 to 10. Like standard PPMs, there is 4dB between scale increments, and the meter spans the range from -16 to +26 dBu.

In addition, a yellow scale marking indicates reference levels, relative to +4dBu. Hence, the yellow zero mark (aligned to 5 on the main scale) equates to +4dBu. The 14 mark (aligned to 8.5 on the main scale) equates to +18dBu, and the 18 mark (9.5 on the main scale) equates to +22dBu.

A silver button above the right-hand meter switches the 8816 on and off, glowing red through the Neve symbol when the unit is on, while one last illuminated button above the left-hand meter selects the sample rate and clock source if the optional A-D card is fitted.

Rear-panel Connections

The rear panel of the 8816 is much simpler than you might expect, mainly because most of the analogue I/O is implemented via five 25-way D-Sub connectors, wired according to the familiar Tascam protocol. One socket carries balanced +4dBu inputs for channels 1-8, while a second handles inputs 9-16. The main outputs are presented on a third D-Sub, catering for the main mix outs, main and alternative loudspeaker outputs, and the stereo cue mix outputs.

Insert sends and returns, plus the external inputs, are carried on the final two D-Subs. The main stereo insert sends, mix insert sends and the aux output are provided on a D-Sub labelled Main Insert Outputs. The corresponding returns, plus the two-track return and stereo mix buss input are carried on a D-Sub labelled Main Insert Inputs.
A recessed button next to the Insert input connector is labelled 'Slave Bus' and connects the mix buss input directly to the mix buss. It

Digital Output Options

The 8816's optional A-D card is a high-quality stereo design based on a Burr Brown PCM1804 converter chip, supporting sample rates up to 192kHz. The DSD format is also supported. When the A-D card is installed, the analogue signal feeding it is normally taken from the mix buss output, but a pair of TRS sockets on the rear panel allow an external balanced line-level signal to be over-plugged to feed the A-D separately. The nominal input level is +26dB for 0dBFS, but this can be adjusted with a pair of recessed headroom switches to give 14 or 18 dB of headroom above +4dBu (in other words, +18 or +22dBu for 0dBFS).

A button on the front panel allows the sample rate to be selected from any of the six standard rates between 44.1 and 192 kHz, followed by the DSD mode. Normal operation is based on internal crystal clocks, but an external AES or word clock reference can also be accepted — both inputs having a bi-colour LED associated with them. The unit selects the best reference automatically, and if no external clock is present, or if an external clock is at a different rate to the selected sample rate, the internal clock is used. If both AES and word clock references are present and at the correct sample rate, the word clock source is given priority.

Word clock in and out are catered for with BNC sockets, and a recessed button allows a 75Ω input termination to be applied, if required. Three XLR connectors provide an external AES reference input and two AES3 outputs. These can be used as separate identical outputs, or for dual-wire operation, although this mode requires an external clock reference. If an external clock is provided, and the sample rate is selected to twice the external rate, the AES output is automatically formatted for dual-wire operation. For example, a 48kHz clock can be provided when the unit is switched to a 96kHz sample rate, or a 96kHz clock for a 192kHz sample rate. In this mode the first XLR carries the left channel data and the second XLR carries the right channel data.

The DSD mode is always referenced to a 44.1kHz clock, and the DSD output is provided on two BNC connectors, switchable from the rear panel for SDIF 2 or SDIF 3 formats. The former carries only the audio data and a separate word clock reference has to be routed along with it, while the latter has embedded clock signals.

GAK PRO AUDIO

PART OF THE GUITAR, AMP & KEYBOARD CENTRE

01273 671971 EXPERT ADVICE

0% FINANCE AVAILABLE OVER THE PHONE

NEXT DAY DELIVERY*

VISIT OUR BRIGHTON STORE.
HUGE RANGE OF STOCK ON DEMO.

79-81 NORTH ROAD, BRIGHTON, EAST SUSSEX, BN1 1YD
FROM LONDON VICTORIA TO GAK IN LESS THAN AN HOUR.

JUST 5 MINS FROM BRIGHTON STATION

** THIS MONTHS FINANCE OFFERS **

PC AUDIO SYSTEMS

GAK supply a full range of PC's, fully configured & tested for optimum audio performance. Simply choose the system you require add hardware/software & let us take care of the rest!!! All our desktop & rack PC's come with a 3 year warranty!!!

FINANCE AVAILABLE

ROLAND

SH-201

Roland continue their synthesiser legacy with the brilliant SH-201. Built in F/X, USB and Roland's renowned supersaw waveform.

£499 24x20.79 0% Finance NO DEPOSIT REQUIRED.

ROLAND

Juno G

Stunning "Fantom-X Quality" sound engine, audio/MIDI multitrack recorder, USB, SRX expansion board, & D-Beam controller. All for an incredible

£799 24x33.29 0% Finance NO DEPOSIT REQUIRED.

BOSS

BR-600

New from Boss the ultimate notebook studio. 8 tracks, built-in drum machine, built-in COSM F/X, USB, built-in mic and battery powered!!!

£299 12x24.92 0% Finance NO DEPOSIT REQUIRED.

APPLE

APPLE POWERMAC & POWERBOOKS NOW AVAILABLE FROM GAK...

Please call and let us put together a complete audio solution to suit your individual requirements. From beginner packages to pro suites we'll find the ideal solution for you.

FINANCE AVAILABLE

BEHRINGER

DDX 3216

No other digital desk gives you this many features at such a low price!

FREE TDF-1616 with every desk!

£449 12x37.42 0% Finance NO DEPOSIT REQUIRED.

NORD

Stage 88

The ultimate performance keyboard! Unrivalled authenticity of all the classic electro-mechanical instruments. The best keyboard we've ever heard.

£2199 36x61.08 0% Finance NO DEPOSIT REQUIRED.

ALESIS

Fusion 6HD

Unbelievable new workstation from Alesis. Full spec'd synth and sampler engine, analogue modeling, 40GB HD, and 8-track simultaneous recording.

£799 24x33.29 0% Finance NO DEPOSIT REQUIRED.

M-AUDIO

Axiom Series

New semi-weighted controllers from M-Audio. 8 pads, 8 rotary encoders and 6 transport buttons.

Axiom 25 £149
Axiom 49 £209
Axiom 61 £239

M-AUDIO

ProjectMix I/O

Record directly into Pro Tools M-Powered 10-bit touch sensitive faders, 8 phantom powered mic pre amps, assignable rotary encoders and scrub wheel.

£949 24x39.54 0% Finance NO DEPOSIT REQUIRED.

KORG

Oasys

The ultimate workstation. Now on demonstration in store. Call now and talk to one of our dedicated OASYS experts! *FROM...*

£5149 0% Finance is available! NO DEPOSIT REQUIRED.

KORG

Radias

Korg push the boundaries yet again with this unique and innovative synth.

£1129 36x31.36 0% Finance NO DEPOSIT REQUIRED.

SE ELECTRONICS

SE 2200A Bundle

Exclusive GAK bundle includes the brilliant SE 2200A large diaphragm studio condenser ideal for a wide range of applications. *Plus...* XLR cable, boom stand, & pop shield.

£179 12x14.92 0% NO DEPOSIT

ALESIS

Monitor One 520

The third generation of the M1 active series, the new 520s offer incredible definition and flat frequency response. Ideal for the bedroom DJ/project studio.

£249 12x20.75 0% Finance NO DEPOSIT REQUIRED.

YAMAHA

AW-1600

8 simultaneous inputs, 16 track playback, 8 mic pre's, 40GB HD, USB 2.0, and more.

£699 12x58.25 0% Finance NO DEPOSIT REQUIRED.

UNIVERSAL AUDIO

UA-6176

Warm up your recordings with this luxurious front end. The ultimate combination of the 610 valve mic pre and the classic 1176 compressor.

£1749 24x72.88 0% Finance NO DEPOSIT REQUIRED.

SEEN A CHEAPER PRICE? PLEASE DO NOT HESITATE TO CONTACT US.

FOR OUR FULL PRODUCT RANGE GO TO...

WWW-GAK-CO-UK

HITECH@GAK.CO.UK

CHECK OUT OUR PRICES... ...WE WON'T BE BEATEN

SECURE ONLINE SHOPPING FROM THE UK'S LARGEST INDEPENDENT RETAILER

*NEXT WORKING DAY, SUBJECT TO STOCK. - WE ARE A LICENSED CREDIT BROKER. ALL PRICES INCLUDE VAT. ALL MONTHLY REPAYMENTS ARE BASED ON NO DEPOSIT & ARE PAID OVER 6,12,24 OR 36 MONTHS AT 0% APR. CREDIT SUBJECT TO STATUS. WRITTEN QUOTATIONS AVAILABLE ON REQUEST. PRICES CORRECT AT TIME OF GOING TO PRESS. E&OE. - PLEASE NOTE GAK.CO.UK LTD IS A SISTER COMPANY & AS SUCH PRICES, STOCK & DELIVERY MAY DIFFER. 0% FINANCE IS NOT AVAILABLE ONLINE.

▶ is intended to allow two 8816s to be connected together as master and slave, allowing up to 32 inputs to be mixed together.

Another D-Sub is provided for connecting the optional 8804 Fader Pack, and quarter-inch sockets provide unbalanced outputs for the main mix and both the main and alternative monitor speakers. The maximum level on these unbalanced feeds is +14dBu instead of the massive +26dBu for the balanced outputs. A slide switch adjacent to the power inlet enables the audio ground to be lifted from the mains safety earth to help cure ground-loop problems, if necessary.

There are numerous internal jumpers provided to reconfigure various aspects of the machine. To re-route the signals to the Fader Pack, jumpers have to be moved for each of the 16 channels, the two outputs and the aux buss, and two more jumpers allow the output monitoring to be taken pre or post the A-D option card's external input sockets.

The 8816's technical specifications are impressive, as you would expect, with a bandwidth that is 3dB down at 60kHz, noise below -80dBu with all 16 channels routed, and distortion below 0.02 percent.

In Use

I'm glad I read through the 40-page PDF handbook early on in my time with the 8816. So much of its functionality is unmarked on the front panel and concealed behind rotary controls which double as push switches! However, having read up on the unit, I found it a doddle to use. Its main purpose — combining analogue DAW signals into a stereo mix — is very simple and straightforward to achieve using the mute buttons, level controls and pan pots. It might sound strange, but I was particularly impressed with the linearity of the pan pots towards their extremes. Some lesser systems seems to become quite abrupt and imprecise as the control nears the ends of its range. The ability to create mute groups and to monitor solo-in-place was also very handy.

Alternatives

There are several dedicated outboard summing mixers available now, although they are all fairly expensive devices. Some, like the Audient Sumo, include mix buss compression or limiting and optional A-D cards, some provide stereo width processing or other tweakery, and pretty much all can accommodate external mix buss inserts to facilitate external processing. However, I don't know of any other unit that combines the functions of a 16-channel mix buss and a well-furnished monitoring controller, nor any designed to work with an external fader panel. This makes the 8816 uniquely versatile and comparison with direct equivalents virtually impossible. However, simpler alternatives include the Audient Sumo, SPL MixDream, API 8200, Nautilus Commander and Phoenix Audio Nicerizer — what a fantastic name!

Recall Software

The 8816 uses the same *Recall* software as Neve's other 88-series outboard units. It's compatible with both Macs and PCs, which are hooked up to the 8816 via a USB socket on its rear panel. The software application required to access this facility is included on a CD-ROM, and the only user intervention is to select master or slave status if two 8816s are connected to the same recall system (via a USB hub).

The software interface is pretty straightforward and intuitive. Pretty much all switch, rotary knob and fader positions can be stored, with the exception of the A-D sample rate settings. When you load a *Recall* file, all switch positions reset automatically and immediately, while a graphical display (pictured below) shows both the current and required positions of any rotary controls that don't match the stored values. An enlarged graphic is provided of each control in turn to make matching the settings as accurate as possible, and as each control setting is matched, that control is removed from the overall display.

This is a straightforward and intuitive facility that will be valued by anyone who might need to rebuild a mix using the 8816 at any future point. It's not quite as slick as using the automation and settings recall facilities of a DAW, but it is the next best thing, and probably quicker and more reliable than writing the settings down! The only thing it lacks is dynamic automation...

The included *Recall* software. As each control is returned to the correct position, it disappears from the software's GUI.

The cue mix system works well, and it is nice to have well engineered talkback facilities built in. Likewise, the aux buss is handy for generating a reverb send when overdubbing vocals, and even more flexible when the optional fader pack is used.

Despite its very unusual scale, the metering works well enough in practice, as does the whole monitoring section — the facilities, while basic, cover at least 80 percent of most people's requirements. My only real complaint is that the control markings, being below the knobs, can be hard to see if the unit is placed on a table. More familiarity with the unit would help resolve this problem, of course.

The stereo width control is a nice facility to have, but the ability to introduce sum-difference matrices into the mix buss insert is actually a lot more useful and creative — a very nice feature indeed. I suspect a lot of people will find the dedicated iPod monitor input useful too, although I can't help feeling that it's akin to fitting remoulded tyres to a Ferrari!

The critical issue, though, is not so much what the unit is like to set up and use, but what it sounds like. The answer is simple: it sounds fabulous and sublime. Its huge headroom and the classic transformer mixing topology endow the mix with an effortlessly silky quality that just screams 'analogue' at the listener. I can't explain what is going on here — any more than Paul could when he reviewed the Audient Sumo — but this unit alters the sound in some extremely subtle way that is, quite simply, extremely nice.

I tried remixing a variety of eight-track material — rock and pop, classical and choral — both internally in a SADiE System 5, and externally through the 8816. Technically, I couldn't really differentiate between the mixes, but sonically I favoured the 8816 every time! Annoyingly, though, mixing internally in the DAW and routing the stereo output through the 8816 seemed to have much the same effect and sonic benefit. It makes no sense to my head, but my ears like it all the same!

After playing with the 8816 I guessed at a price of £3500-4000 — it sounds that expensive. The reality is that it retails for only a little over £2000, which is quite remarkable, especially when you consider the monitoring functionality built in. The A-D and fader options will obviously add to that price, and although neither is essential, they are certainly nice options to have. Were you to also throw in a rack of Neve 1081R mic preamps, you would have the basis for a very nice portable recording outfit. Now there's a thought... SOS

Information

£ 8816 Summing Mixer £2173.75; 8804 Fader Pack £1169.13; A-D option £470. Prices include VAT.
T AMS Neve +44 (0)1282 417281.
F +44 (0)1282 417282.
E sales@ams-neve.com
W www.ams-neve.com

the ultimate hook-up.

EUREKA

Professional Recording Channel

Class A transformer-coupled
microphone/instrument/line preamp
Selectable input impedance and
"saturate" tube warmth control
Fully variable ultra fast compressor
w/ separate make-up gain
3-band parametric EQ
with bandwidth "Q" control

Electronic Musician 2005 EDITORS CHOICE

AD192

Optional Digital Output Card for the Eureka

24-bit resolution, up to 192K sample rate
Simultaneous SPDIF and AES outputs
Analog line input enables one AD192 for two Eurekas

Hook-up two Eurekas to your FIREPOD via AD192 SPDIF output for the ultimate signal path. Hook-up your FIREPOD to your computer via standard IEEE1394 FireWire connection and use the eight onboard Class A preamps for recording drums, guitar cabs, pianos, ensembles, etc., for the ultimate FireWire computer interface. Need more inputs? Hook-up a second FIREPOD for eight additional microphone preamps. Finally, hook-up your Central Station to your FIREPOD for ultimate communication and monitoring control.

FIREPOD

24Bit/96k 10x10 FireWire Recording System

Eight PreSonus microphone preamplifiers
Two instrument inputs, eight analog line outputs
SPDIF and MIDI input / output
Cubase LE 48-track recording software

Mix CERTIFIED hit

CENTRAL STATION

Studio Control Center

Flexible stereo input/output
switching & routing
Talkback, mute, dim, mono;
separate cue and main outputs
24-bit/192K D to A converters
(SPDIF and Toslink)
Passive signal path (no op-amps)
for ultimate stereo imaging and depth

TEC

CSR-1

Optional Remote Control for the Central Station

Main volume, input/output switching, dim, mute, mono
Talkback microphone on remote
Rack your Central Station with your gear

freedom to record

PreSonus

NEW:
Browse our interactive catalogue
on the internet. Only available at:
www.musicstore.com

The Music Store .....over 12000m² service, storage & demo space

100x MEGADEAL!

Casio WK 3700
The ambitioned 76 keys from CASIO!
32 polyphonic voices / 16 part Multimode,
232 sounds GM standard, Stereo out/
Midi,120 Styles, 20 piano orientated patterns/
10 memories for your own styles

£266

20x

Kurzweil SP 76
32x polyphonie, 32 Sounds, 9 Preset Effects,
8 different Reverbs, Chorus, 32 User Setups,
6,3 mm Jack (L/Mono, R), Headphone-Out,
Midi-In/Out, Dual Switch-Pedal,
weight: SP76 12,5kg

£383

30x

Alesis Fusion 6HD/
Alesis Fusion 8HD
88 weighted key (Fusion 8HD) or
61 half weighted, touch sensitive keys
(Fusion 6HD), up to 180 voices.
From the idea nonstop into the charts

£739

Alesis Fusion 8HD
£999

20x

Alesis QS 8.2
A new 88 keys Synthesizer with
hammer weighted action. 640 Sounds
in 16 MB ROM, a good Stereo Grand
Piano Sound and the 500 Multisounds.
64 voices, 16 part Multimode,
4 independent effect processors.

£515

HIT PRICE!

Yamaha MO 8
The MO6 delivers Yamaha's powerful
MOTIF synth engine functions and
sounds in a compact, affordable
workstation.

£1125

100x

Alesis Photon 25
Equipped with 25 touch sensitive
keys in normal size, this handy
USB/MIDI Controller is great for
both stational and mobile use.

£66

100x

Alesis Photon X25
Photon X 25 is a spacy design
USB/MIDI controller with integrated
Audio-Interface and extraordinary
real time controls.

£119

70x

Model SP1
Handy, easy to transport
stage piano with intergrated accompany
sequencer and speaker system for an
unbelievable price

£333

1000x

Music Store Midilink
1 x 1 USB Midi Interface.
Works without drivers on Win XP
& Mac OSX

£13,99

100x

Tascam US 122
USB 24-bit audio MIDI-interface
for Windows and Mac computers.

£105

Korg PA-50
The Pa50 features 2 dual speakers
in a reflex cabinet and twin
15 Watt amplifiers, 304 editable Styles.
Included with its 660 editable sound pro-
grams is a set of very expressive solo instru
ments, sampled with a natural vibrato.

£629

Roland

Roland FP-2
This affordable instrument comes with
an 88-note Progressive Hammer Action
Keyboard, superb piano and instrument sounds
and a refined new look with onboard speakers.

£729

Kurzweil PC1X
Aside from the diverse library of the
Kurzweil PC2 series with 512 top sounds,
the PC1 X has been expanded with the
high quality Orchestral ROM.
88er Hammermechanic keyboard. Through 64 v
and expressionful classic sounds, the PC 1 X is ideal for Pop & Rock

£799

Roland VK-8
New generation of the legendary
VK-7 with improved Sinus Sound
(taken from the flagship VK-77)
61 keys, Waterfall, 128 stimmig polyphon,
D-Beam Controller, Vibrato, Reverb, Overdrive

£1039

Solution Expert

Germanys
No.1 Apple Audio
Solution Expert

t.c. electronic

TC Electronic Konnekt 24D
FireWire Audio-Interface, 14 I/O,
Konnekt 24D Integrated, incl. Cubase LE.

£342

TC Electronic Konnekt 8

£239

MOTU

Symphonic Instrument
Plugin for Windows & Macintosh,
Symphonic Instrument 8GB
of orchestral sounds

£179

Ethno Instrument
The Ethno Instrument offers
expressive, ethnic instrument sound
with authentic world music loops in
an easy to use window.

£184

Music Store Garage Key
GarageKey is the ideal supplement
for the GarageBand software
by Apple. Easy to use, small and
compact. It will find room on every
desk. The perfect MIDI keyboard
for the Mac.

£46

Bundle Price!

TC Powercore
MK II + VSS3

£617

Bundle Price!

TC Powercore
MK II + MD3

£892

Motu
UAD-1
Ultra Pak
DSP Card
incl. 24 Plugins

£789

Motu
UAD-1
Project Pak
The UAD-1
Project Pak
operates with
most of the
DAW-programs
under WIN98
to XP, aswell as
Mac OS9.x
and OS X.

£267

US-2400 DAW-REMOTE
The Tascam US-2400 is the first DAW
fader controller with enough faders
to handle a typical recording session
and feel like a real mixing console!
24 touch-sensitive moving channel
faders and a touch-sensitive moto-
rized master fader, each with 10-bit
resolution for precise level control.

MEGADEAL!

£586

MOTU

NEW!

MOTU
Ultralite 3D
The UltraLite is compact,
great for mobile use and
studio alike and is power supplied
via FireWire. Half the size of a 19" device,
it has two microphone-/instrument inputs, and illuminated display with
level display of all in- and outputs, a programmable CueMix DSP, SMPTE-
sync and much more

£409

New! Also available as USB 2.0 Version!

MOTU 828mkII FireWire/USB 2.0
The 828mkII is an expandable, powerful 24-bit, 96kHz digital audio
interface with 20 inputs and 22 outputs. It provides 10 channels of pris-
tine 96kHz analog recording and playback, 8 channels of ADAT digital
I/O and stereo S/PDIF and a CueMix DSP near-zero monitoring system.

£499

MOTU Traveler
Powerful 24-bit and 192kHz. For mobile use.
The MOTU Traveller.....The Traveller provides
8 channels play/record in high professional quality
up to 192kHz. Therefore he owns all digital connections you'll need:
ADAT optical, AES/EBU and S/PDIF(optical/phono-lead).
Expand your system through connecting other
Traveller, 828 or 896 FireWire Audio Interfaces.

£599

Sony Oxford Bundle
Oxford Bundle for PowerCore EQ,
Dyn, TransMod, Inflator

£755

PreSonus

DEAL!

Presonus Inspire
FireWire Audio-Interface, 2 In-/ 2
Outputs, incl. Cubase LE + Samples

£135

Presonus FireBox
FireWire Audio-Interface,
6 In-/ 10 Outputs, 24 Bit/ 96Khz.

£225

Digidesign
Icon D-Command

Call for BEST PRICE

SYNTH / PC-HOTLINE & ORDER SERVICE: +49/221/92 57 91 -3301 · pc@musicstore.de · synth@musicstore.de
MAC-HOTLINE & ORDER SERVICE: +49/221/92 57 91 -306/ -805 · Fax -0221/92 57 91 -809 · mac@musicstore.de

One of Europes biggest Mail Order Companies

TRANSPORT FLAT CHARGE ONLY £7 (10€)

MUSIC STORE professional

www.musicstore.com

20x TASCAM DEAL!

£130

Tascam 414 MKII/Fame Micro Bundle
4-Track Portastudio inclusive Fame DM 330 Microphon!

100x LOW PRICE!

£89

Behringer V-Verb REV 2496
Reference-class reverb modeling processor with high-quality 24-bit/96 kHz A/D and D/A converters.
2 separate effect processors, 10 different routing possibilities, AES/EBU-, S/PDIF, high resolution graphic LCD-display,

PRICEDROP!

£609 £1239

Yamaha AW-1600 **Yamaha AW-2400**
The AW1600 and AW2400 have everything needed to record your music anytime, anywhere in compact, road-worthy designs. New features include 8 combo inputs (all with Phantom power), USB 2.0 drag and drop capabilities for moving files onto your computer, 24-bit recording and a new Pitch Fix algorithm for correcting the pitch of vocal tracks.

40x

£688

Behringer MX-9000 Eurodesk
Behringer's largest console features 24 fully inline channels

HIT PRICE! incl. Slimline CD/RW Writer

£344

Fostex MR-8 HD Bundle
8-Track HD-Recorder, 4Track simultan recording, 4 Mic/Line Inputs with 48v, 40GB HD, incl. USB 2.0 Slimline CDRW-Recorder

100x

£60

ART Tube MP Studio V3
Tube mic preamp with limiter & VU, hand selected 12AX7A tube, +48V Phantom Power, PhaseReverse, OPL output protection limiter, XLR & jack I/O,variable valve voicings, perfect for hard disc recording or as a tube DI

£481

TC electronic Voice Live
4-voice vocal harmony processor with scale based pitch-correction and effects (3-Band EQ with Low CUT, compressor/gate, resound + to echo).

PRICEDROP!

£185

Mindprint TRIO
Desktop Monitor Controler Mic/Line Amp/Mixer/Monitor-IF. TRIO combines all needed outboard components for PC-recording in one device for the first time.

70x MEGADEAL!

pair £688

Tannoy Ellipse 8
50 kHz, 8" Dual-Concentric-Chassis in highpass section. 3-way speaker with a huge variety of sounds and authenticity. 3 high quality poweramps, symmetric inserts, adaptions for level and HF/LF switches in a speaker.

TANNOY

Bundle Price!

£229

Fame 1050AM/SUB-08 Bundle
2x Fame 1050AM studio monitors paired with the uncompromisingly good Fame subwoofer for a full sound for the complete frequency range. Includes 2x 3m XLR-XLR cables!

£370

Avalon U-5
Mono Instrument/DI PreAmp

Bundle Price!

pair £395

Genelec 8020 APM Set
Active Monitor-Set incl. orginal Genelec bag

pair £219

Behringer 2031A
Active 2 way studio monitor with 8.75" bass speaker, wave guide, limiter

£1490

Avalon VT-737
High quality tube preamp

£1690

Avalon VT-747
Highend mastering processor

£455

Genelec 7050B
70 w, 25-85Hz ±3 dB (LFE 85/120Hz

£666

SPL Channel One
The Channel One is a complete channel strip with all the tools onboard for recording voices, instruments and digital or analog audio sources with the highest sound quality

NEW!

pair £549

Adam Audio A7
2-way active bass reflex speaker 6,5"Rohacell/Kevlar sandwich woofer A.R.T. tweeter, frequencyresponse 46Hz - 35 kHz, power 100W (sin), 150W(rms)balanced XLR + unbalanced RCA Inputs.

pair £877

Adam Audio P11
Nearfield Monitor, 2-way active bass reflexspeaker, 2x80Watt, Woofer: Ø 182 mm (7.25 „) Homex®cone , A.R.T. tweeter, adjustable highfrequenzresponse +/- 2dB und +/-4dB, +/-3Dbshelve Filter <150Hz und >6khz

pair £1098

Dynaudio BM-6A
2-way Active Nearfield Monitor System, Frequency Response (+/- 3 dB) 41 Hz - 21 kHz, Amplifier power Tweeter: 100 W / Woofer: 100 W.

£759

Neumann KM 184 mt Stereo Set
The KM 184 (cardioid) in particular has become the classic high-quality miniature microphone. Now also available in an attractive stereo set.

£639

Brauner Phantom AE
Black Finish!! Special Edition Brauner Phantom C. World-Wide only 1000 pcs. Incl. Case, elastic suspention, high quality 5mtr. VOVOX cable

£1639

Brauner Valvet
The Valvet is quite possibly the best value in high end microphones available.

£229

Sennheiser MD0421-II
The MD 421 II is one of the best known microphones in the world. Its excellent sound qualities enable it to cope with the most diverse recording conditions and broadcasting applications

£99

Sennheiser HD-25
Reference headphone for movies and television, for homerecording and DJ's. Frequency range: 16-22.000 Hz, 120 dB, 70 Ohm 140 g,

DEAL!

£449

AKG Heard 999 Audiosphere II Set
Wireless highend-Headphone-System with Dolby Surround Pro-Logic and Dolby Digital Decoder

TC electronic C-300
Dual engine compressor/limiter gate/expander. 16 compressor/limiter presets and 16 gate/expander presets optimized for selectable audio sources Mix features for new style compression

£149

SPL Transient Designer 2
The firstdynamic processor from SPL with the new DifferentialEnvelope Technology, that allows the envelope of the sound to be changed bringing very musical results.

£299

Presonus Central Station
THE ULTIMATE STUDIO CONTROL CENTER
19" monitor matrix, diverse connection possibilities with 5 stereo inputs, 2 of which are digital with 24Bit/192kHz converters and a dynamic response of 121dB,3 seperate stereo outputs.

£379

Fame Studioamplifiers

A-200
2x 100W, 4ohm, mono bridge, fanless cooling, XLR-, jack- and speakon

£84

A-400
2x 200W, 4ohm, mono bridge, fanless cooling, XLR-, jack- and speakon

£106

Benchmark Media DAC-1
2-Ch. DA Converter, 24b 192KHz, AWC-Gen. 1HE 9,5",

£685

£799

Alesis HD-24
With the development of the H 24 ALESIS created one of the top-selling hard disk recorders. 24 tracks at 24-bit/48kHz, 12 tracks at 24-bit/96kHz, 3 ADAT Optical I/O, Ethernet

£599

Universal Audio Solo 610
M-610 Mono Tube Mic-Preamp, 48v Phantom Power, Lo Cut filtering, Phase Reverse, and for evengreater tonal variety, maintains the 610's flexible dualimpedance selection for both Mic and DI inputs

£739

Fostex FR-2
The new Fostex FR-2 has been designed specifically for location sound (up to 24-bit / 192KHz) recording.

£75

Presonus Tube Pre
Microphone preamp with excellent sound! XLR Mic input, high ohm instrument input, Tube Drive: 0dB to +20dB, Gain: 3dB to +48dB, Phase switch, 20 dB Pad switch, 48V Phantom power, 80Hz low cut, XLR and Jack outputs

£739

Tascam HD-P2
Portable, high resolution stereo recordee. Recording resolution selectable between 16 and 24Bit from 44,1kHz to 192kHz

£76

Tascam Porta 02
This compact cassette player can record up to 4 tracks (2 of them simultaneously). 2 Jack ins and 4 Chinch outs, it is a great „notebook" for very fast and easy recording

70x MEGADEAL!

£92

TC electronic M-300
The M300 is a Dual Engine Processor with high quality Reverbs and Effects.
The Processor covers virtually any effects application, from PA and Live to Studio and Club Installations by combining a dedicated true Stereo Reverb engine and a Multi-purpose Effects engine.

incl. Firewire option!

£382

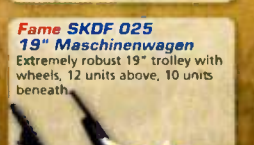
Mackie Onyx 1220
The smallest mixer in the family, the Onyx 1220 is an 8-channel mixer (12 inputs: 4x mono + 4x stereo)

£580

Mackie Onyx 1620
This premium 16-channel 4-bus format analog mixer features all-new circuitry including 8 boutique-class Onyx mic preamps with 4-band Perkins EQ

£982

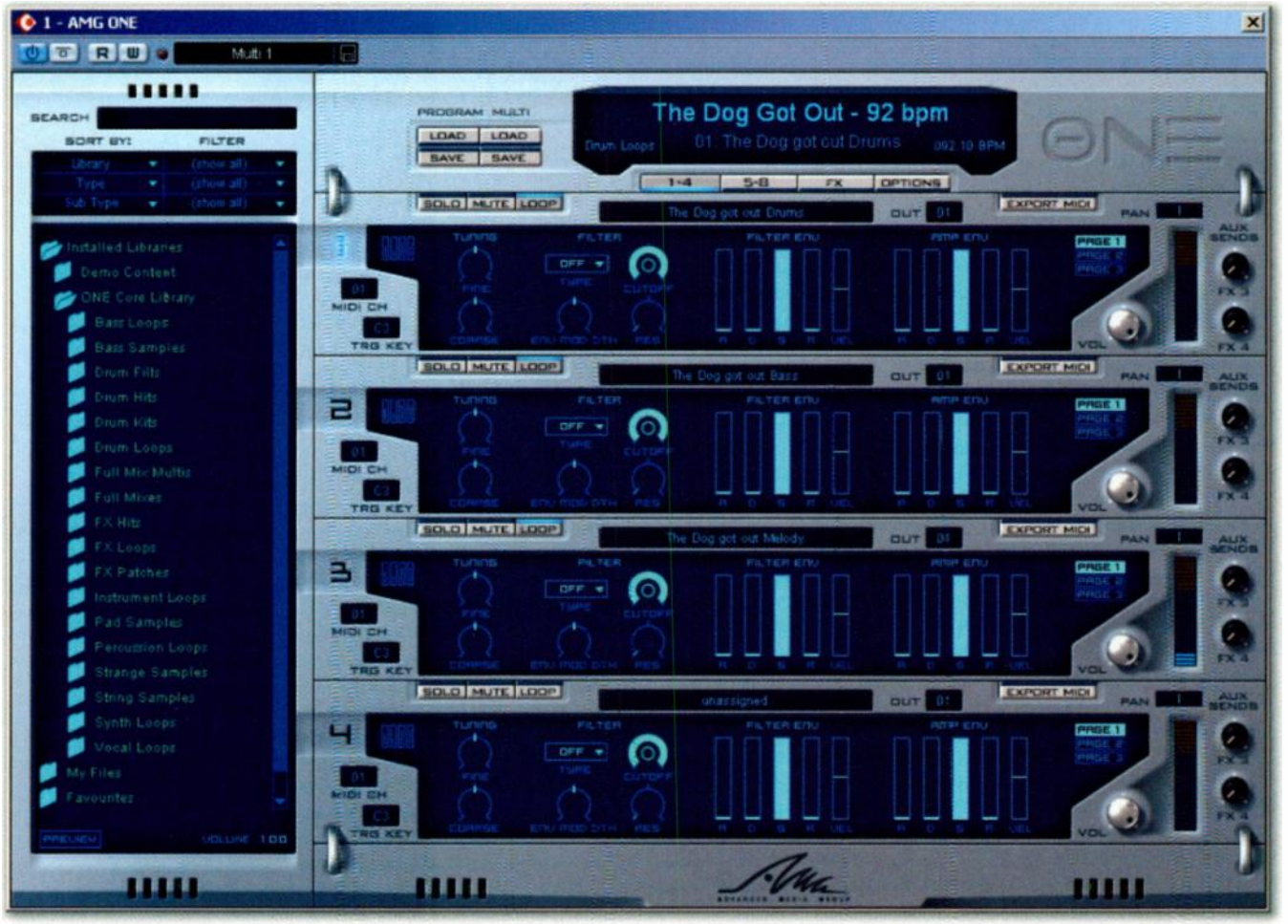
Mackie Onyx 1640
This premium 16-channel 4-bus small-format analog mixer features all-new circuitry including 16 boutique-class Onyx mic preamps with 4-band Perkins EQ



£68

Fame SKDF 025 19" Maschinenwagen
Extremely robust 19" trolley with wheels, 12 units above, 10 units beneath

MUSIC STORE · GROSSE BUDENGASSE 9-17 · 50667 COLOGNE · GERMANY
REC-HOTLINE & ORDER SERVICE: +49/221/92 57 91-3304 · SHOP DEPT.: +49/221/92 57 91 -3314 · Fax -869 · recording@musicstore.de



AMG One

Software Loop Player For Mac & PC

John Walden

Working with musical loops is now a well-established part of the musical mainstream. Could you do with *One* more tool to get the job done?

If your musical productions are predominately based around pre-recorded musical loops, applications such as Sony *Acid* and Ableton *Live* offer comprehensive and mature platforms to get the job done. However, while similar facilities for working with loops are now integrated into all the major sequencers, sometimes a dedicated tool is what is required. Enter AMG's *One*. AMG are well known to *SOS* readers for their sample-library products, but *One* represents their first software instrument. Available either with or without AMG's Core Library, *One* includes a comprehensive feature list for tempo matching and manipulation of loops as well as a well-specified Step Sequencer.

Round *One*

I tested using the VST plug-in within *Cubase SX* running on a PC system — the Mac VST and AU versions were, however, supplied on the same DVD. Installation of the plug-in was straightforward, and AMG use challenge-and-response copy protection for *One*, with a machine-specific ID number being generated by the plug-in. Once registered on AMG's web site, a licence file is returned via email to authorise the plug-in and allow access to the Core Library content. This all worked without a hitch on my test system. The Core Library itself is a hefty 4.5GB of sample material (see the 'Library Work' box for details) and AMG have other libraries in development. While the Core Library content can only be accessed via *One*, the plug-in can load loops or samples from other libraries, including *Acid*ised WAVs and REX files. Users can

SOUND ON SOUND

AMG One

pros
- Good set of features for working with loops.
- Simple to use.

cons
- A few minor niggles about the user interface.
- Core Library will only appeal to a limited number of musical styles.

summary
AMG have done a good job with their first virtual instrument, which manages to strike a sensible balance between features and simplicity of use. If you need a tool for working with loops that goes a little further than the features provided by your sequencer, *One* is well worth a look.

System Requirements

- **PC: Pentium III 1GHz processor with 512MB RAM running Windows (no version specified). 16-bit soundcard. VST 2 plug-in compatible host software. 4.5GB hard drive space for Core Library installation.**
- **Mac: 800MHz G4 with 512MB RAM running Mac OS 10.3.9 or better. VST 2 or Audio Unit plug-in compatible host software. 4.5GB hard drive space for Core Library installation.**

specify the location of the library via *One*'s Options page.

One's main screen is split into three main areas. On the left is the Browser, which includes search facilities to assist in finding loops or samples of a particular type from within the Core Library. The top strip of the *One* window contains the Global Controls and, in addition to the buttons for loading/saving either individual loops or Multis (collections of up to eight loops and their associated settings), it includes an information display that shows data for the currently selected loop or sample.

The four buttons at the bottom of the Global Controls section dictate what is shown in the bulk of the window. These buttons toggle between the eight Racks (two banks of four) and a single instance of *One* can therefore be used to mix and match eight loops. The other buttons call up the Options page and the FX page. *One* features four user-configurable FX units — two Insert units for processing an individual loop, and two Send-Return units for global processing.

On The Rack

One's Rack units are where all the serious work gets done. A loop or other sample can be loaded into a Rack via the Browser. The MIDI Channel and Trigger Key can be adjusted via clicking and dragging with the mouse. This works well enough, but can be a little fiddly — the ability to double-click and then enter a value via the keyboard would be a useful addition. These settings can be used to create a number of different configurations but, most obviously, they can be set for 'live' layering of loops, with all eight Racks configured to respond to the same MIDI channel, and with the individual Racks being triggered by successive notes on a master keyboard.

If single samples (as opposed to loops) are loaded into a Rack, they are simply mapped as a playable instrument across the MIDI keyboard. *One* then acts as a simple sample playback tool, although all the usual problems of samples speeding up/down to achieve the changes in pitch apply here — this is not a tool for sophisticated multi-sample instrument playback. In contrast, if a loop is loaded into a rack, the Loop switch toggles between two playback modes. With Loop mode on, the Trigger Note switch causes the whole loop to play back in sync with the project. With Loop mode off, a beat-sliced version of the loop is mapped across the keyboard starting at C1, and the individual slices can be triggered. The Export MIDI button creates a MIDI file that will trigger the loop slices in sequence to recreate the original loop performance. This file can, of course, be reloaded into your sequencer for further manipulation as required. The Out setting allows the loop to be routed to any one of *One*'s four stereo outputs or to one of the two Insert FX units. These in turn can be routed to any of the four main output pairs. The Solo, Mute, FX3, and FX4 Aux Send controls and the Pan and Volume controls complete the line up.

The options displayed in the central portion of each Rack can be switched between three main views — Synthesizer, Waveform, and Step Sequencer — using the selector button located at the top left (next to the Rack number). In Synthesizer mode (the default), three pages of controls are available (toggled via the Page buttons towards the right side of the Rack). Aside from Fine and Coarse Tuning controls, the screens are dominated by controls for the multi-mode resonant filter, filter envelope, amplifier envelope, and filter and pitch LFOs. These manage to combine plenty of sound-shaping possibilities, without being so complicated that a PhD is required in order to use them! Creating both subtle and extreme filter-sweep-type effects is a doddle, and real-time control is made much easier by a well-implemented *Reason*-style MIDI Learn facility for assigning a hardware controller to a particular *One* setting.

Test Spec

- AMG *One* v1.01.
- Athlon Dual core 4400+ with 4GB RAM running Windows XP Pro (SP2).
- Echo Mia 24 soundcard.
- Steinberg *Cubase SX* v3.1.1.

With a loop loaded, switching to Waveform view shows the beat-sliced version of the sample. While no control is provided over the slicing process itself, there are some excellent options for manipulating slices. These include the ability to shuffle slice positions and mute, reverse, or change the volume, pan, or pitch of individual slices. Slices can also be selected and then replaced with another sample, and a further twist to this is the very neat multi-layer option, where a second sample can be layered with a particular slice. The obvious application for both these features would be the replacement or beefing up of drum sounds within a loop and, while it is not without its limitations (for example, the length of a layered sample is fixed to the length of the slice from the original loop it is being layered with), it is easy to use and offers plenty of creative possibilities.

One Step At A Time

As mentioned above, the third option within each Rack is the Step Sequencer. Up to 32 samples can be loaded into the Sample ▶

The Rack panel display alters depending on the operation mode — here you can see the display in Waveform mode (top), Step Sequencer mode (middle), and in Synthesizer mode with pitch and LFO settings visible.

▶ Matrix. For each of these samples, a 16-step sequence can be constructed and steps can be accented. Sample volume, pan, and pitch can be also be adjusted as can the 'swing' and accent strength.

Interestingly, if the Rack also contains a loop in the Synthesizer mode, this loop and the samples in the Step Sequencer are layered, and both play back in response to the Trigger Key. For Step Sequencer fans, perhaps the most obvious limitation is that each Rack can only hold a single sequence — if you want to create variations on a Step Sequence, then the same samples have to be loaded into a second Rack, although this can easily be done by saving the Rack patch and re-loading it into a second Rack.

More Than *One* FX Option

As indicated earlier, *One* includes four separate effects processors — two Insert (FX1 and FX2) and two Send-Return (FX3 and FX4). The effects types available include all the usual suspects: reverb, delay, various modulation effects, EQ, compression, gate, distortion, and various filters. Usefully, *One* includes a large number of preset effects patches suitable for common tasks. For each FX slot, the functions of the six controls change to reflect the particular effect type selected.

The output destination of all four FX units can be specified, as can the 'mix' level. The routing options also allow a signal from FX1 or FX2 to be passed to the two Send-Return effects if required. Overall, the quality of the effects processing is respectable, with a good degree of control, although it would be unfair to expect miracles of the reverb algorithms given the price of *One* and its prime function.

One Versus The Rest

During my testing period, *One* behaved itself impeccably within *Cubase SX*, and was fairly undemanding on CPU resources — I was able to fully load several instances of *One* and the Performance Meter barely gave a flutter. On the whole, the user interface is straightforward, and AMG have struck a sensible balance between the creative options offered and the level of complexity. That is not to say that *One* is without quirks. Some of the controls can be a little difficult to adjust and (as with the MIDI channel and Trigger Key settings) it would be nice to have the option to enter values via the keyboard. It would also be preferable if the MIDI Export option could pass the data directly to a track within the host sequencer.

It is almost inevitable to draw a comparison between *One* and NI's *Intakt* (reviewed in *SOS* June 2004). The general looping options — with automatic tempo matching — offer similar levels of functionality, but there are also some differences. For example, *Intakt* provides a useful virtual keyboard for auditioning loops without you having to take your hand off the mouse, while *One* includes its simple (but effective) Step Sequencer. One trick missed by both products is the *Acid*-style automatic preview of loops from the Browser while other loops are playing. *One* does include a Preview button within the Browser, but this has to be clicked with the mouse to trigger playback — a Preview Always On option would be most welcome.

One's four internal effects units. The top two provide insert effects, while the bottom two offer send-return treatments.

Conclusions

So if you need a dedicated looping tool, is *One* the one? Although *One* is well thought out and offers plenty of creative options for those that like to work with tempo-matched, beat-sliced loops, whether it can challenge other well-established loop tools such as *Intakt* or *Phatmatik Pro* is another matter. I would guess that two factors will dictate the choice for most users: firstly, while each product offers similar core tools, the slightly different feature sets may appeal to particular users; and, secondly, for *Intakt* and *One*, the contents of the provided sample sets might tip the balance. Sensibly, AMG also offer just the plug-in at a lower price, and in this form it ought to compete well with its more established rivals. A demo version of *One* should be available by the time you read this, and for loop-heads everywhere, it is well worth checking out. SOS

Library Work

One can be purchased either on its own or bundled with the massive 4.5GB Core Library. This library provides a collection of both loops and individual samples that are fully formatted and tagged to work with *One* and the search facilities built into the Browser. In terms of instrument groups, these are split into the conventional section including bass, drums, strings, synths, vocals, percussion, pad, and FX types.

The quickest way to get a flavour of the material, however, is via the Full Mix Multis. These each comprise a *One* Multi patch, and the majority contain three related loops (drums, bass, and melody/chords) loaded into the first three Racks. There is some excellent material here covering a range of styles, but titles such as Black Hoodie, Save Hip-hop and NuGroove Soul give a clear indication that Hip-hop and R&B producers are well catered for. What is equally impressive, however, is the sheer number of Multis — over 300 in total.

AMG have other '*One* + library' bundles in preparation, with sample sets aimed at specific musical genres — *Metropolis* (Hip-hop) and *Infinite Groove* (acoustic drums) will be available soon with other content-only libraries (including a large World music loop set) to follow. If these contain as much good material as the Core Library, then they will certainly broaden the appeal of *One* itself.

information

£ Instrument and Core Library, £150; Instrument alone, £tbc. Prices include VAT.
T AMG +44 (0)1252 717333.
F +44 (0)1252 695680.
E matt@amguk.co.uk
W www.amguk.co.uk
W www.samples4.com

Fu·sion (fyōō'zhən) n. 1. The merging of different elements into a union 2. A union resulting from fusing: "A fusion of sound and technology emerged." 3. A complete music workstation

An 8-track hard disk recorder within a keyboard. Understated sex appeal included.

Hard Disk Recording
8 Tracks of direct-to-disk recording with a Built-in 40gig hard drive. Complete Song Production (MIDI and Audio).

Sampling
Sample your own sounds into the Fusion or take advantage of the high quality, built-in instrument samples.

Analog Modeling
Create the sounds of classic analog synthesis for huge pads and searing leads to go beyond the average keyboard workstation.

fig 1: Alesis Fusion 8HD

FM Synthesis
FM (Frequency Modulation) to recreate classic and new digital timbres with complete real-time manipulation of the synthesis parameters.

Physical Modeling
A whole new synthesis technology to add unique timbres and textures to your musical ideas.

fig 2: Alesis Fusion 6HD

NEW LOWER PRICING.

Typically £999 (8HD) and £799 (6HD)inc vat

Call 01252 341400

Please visit www.fusionsynth.com for MP3's and info

FUSION where great ideas converge

alesis.com

ALESIS®

Today The Bedroom, Tomorrow The World

Thanks to iTunes, on-line music distribution is a reality — but can you really use it to make money from your music?

Simon Trask

Distributing Your Music On Line

There can be no doubt that on-line music downloads are now an integral part of contemporary culture, and increasingly of the music industry. Apple's success with the iPod and the iTunes Music Store has been remarkable — over a billion tracks have been sold at the time of writing, and the Store is now open for business in 21 countries from Denmark to Japan, Australia to Sweden. And in just a couple of years, downloaded music in the UK has gone from having its own separate chart to forming a major component of the sales that determine the Top 40 singles chart each week.

New Tricks

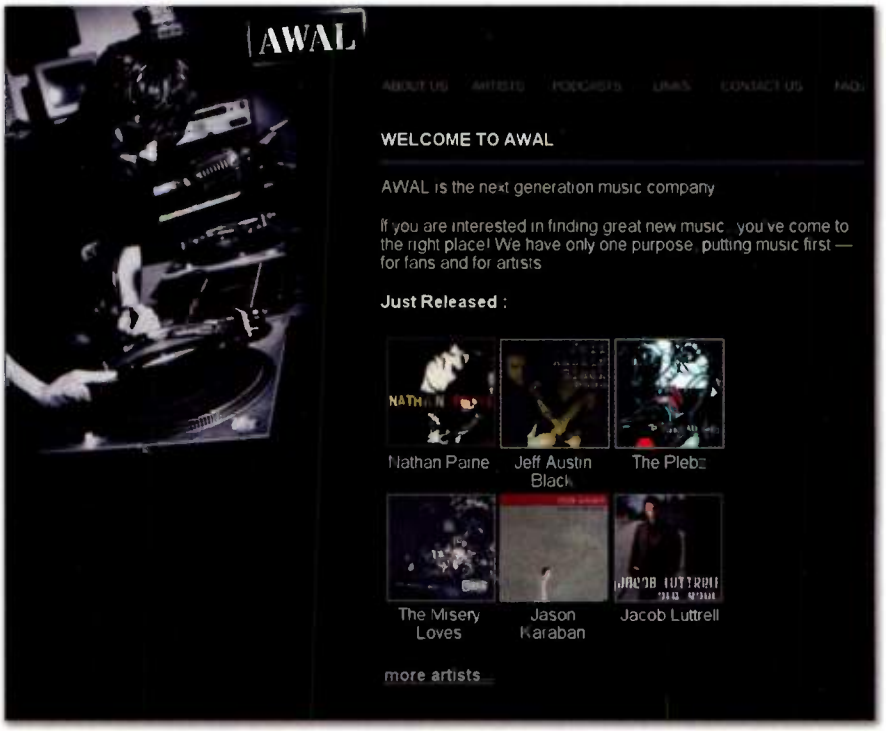
Of course, the downloadable sales model espoused by Apple isn't the only one — streaming and downloadable subscription models are used by the likes of Emusic, Napster, Playlouder MSP and Rhapsody. And while Apple uses the AAC file format and 'digital rights management' software, or DRM, as it's euphemistically known, on the tracks they sell at the iTunes Store (in other words, copy protection), many indie-focused services prefer to stick to high-quality MP3s with no DRM, preferring to trust fans to buy music by the artists they like rather than sharing it for free. Warp Records' site at www.bleep.com is a prime example of this approach, selling albums from over 200 indie labels in MP3 format for £6.99, or at 99p per track.

Meanwhile, the new breed of digital music distributors like CD Baby and The Orchard support unsigned artists not only by selling their CDs on line, but also by acting as so-called 'aggregators', and getting unsigned music onto a wide range of on-line music services that won't deal directly with individual artists (for more on the former, see my article in *SOS* February 2004 at www.soundonsound.com/sos/feb04/articles/onlinemusic.htm). The Independent On-line Distribution Alliance (IODA) is another body which has emerged more recently to enable independent rightsholders to get their music into the digital music outlets in return for a portion of the licence royalties paid out by the on-line services. IODA provides on-going rate negotiation, audio-encoding, data-management and reporting services, and of course handles the administration and distribution of royalties. By aggregating music from independent labels and musicians, it is able to provide collective bargaining power; the company works with over 750 indie labels of a variety of sizes, and now distributes a catalogue of over 200,000 tracks.

The likes of IODA are helping to level the on-line playing field for independent artists and labels who otherwise wouldn't have the resources and the ability to compete on-line with the majors. Over the past couple of years, podcasting has arisen as another way to get homebrew music out to people, and Apple's adoption of podcasting in its iTunes store has helped greatly in raising its profile. The term 'podsafe' has come about to describe music which is owned by the artist and released under a Creative Commons licence, thus removing any restrictions on the use of such music when it is broadcast non-commercially (for more on Creative Commons, see www.soundonsound.com/sos/jan05/articles/creative.htm). Such is the profile of music podcasting in the States that the US royalties body ASCAP recently introduced interactive and non-interactive podcasting licensing agreements.

Another musician-friendly success story has been the rise of Myspace, a 'social networking' web site which has become a popular place for signed and unsigned musicians to post their work. The music section of the site allows surfers to search by category, keyword,or musical genre, and site

Neil Leyton, boss of independent label Fading Ways.



Artists Without A Label, the on-line distributor that bought you the likes of Arctic Monkeys and Editors.

users can be notified of upcoming gigs and events relating to the bands they follow through Myspace. Such features have proved a hit with the under-30s, leading to a flood of trendy articles focusing on 'The Myspace Generation' and no doubt contributing to last year's successful sale of Myspace's parent company to Rupert Murdoch's News International for 580 million dollars.

The Musicians Speak

These are just some of the new ways musicians can use the Internet to obtain a wider audience for their music. But what do the musicians themselves think? Neil Leyton, label boss for forward-thinking indie label Fading Ways and an artist in his own right, has a typically pragmatic indie attitude: "Any new technology presents challenges to and opportunities for an existing business model. The Internet affords artists huge promotional potential and the prospect of new distribution models. It's best to work with it rather than against it, respecting the artists' wishes on how they want to be marketed."

But as it becomes easier for unsigned artists to promote, distribute and sell their music on line, are labels becoming less significant? Leyton: "That depends on the label. Artists still need someone to handle the business of selling pieces of plastic and distributing them, or, alternatively, marketing their on-line stores to drive fan traffic towards them. It's all a question of adaptation — and dinosaurs have trouble evolving. Labels and artists should become equal partners in the age of the Internet, as opposed to artists being owned or controlled by labels. There is no need for an artist not to own their copyright, for example."

One highly evolved organisation whose very name calls the need for record labels into question is AWAL, or Artists Without A Label. Like CD Baby, AWAL enables independent artists and labels to get onto iTunes and its ilk. AWAL UK now represents around 400 organisations ranging from unsigned artists to labels with whom iTunes won't deal directly. "Just think of the manufacturing muscle that would traditionally have been needed to reach, say, eight million people in Europe and North America" says Paul Bower of AWAL UK. Now you can do that by uploading the contents of a single CD-R. It's the smaller independents that will really flourish. While they have smaller budgets to play with, they can now shift more funds towards promotion rather than manufacturing."

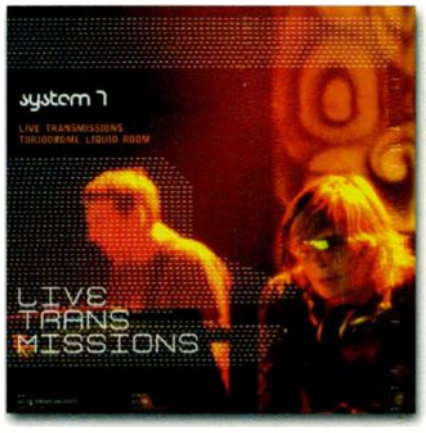
"The rapid development of a digital music supply chain, pulled by the success of iTunes, is enabling more recorded music to get into the marketplace than at any time in history,"

says Paul Sanders, co-founder of Consolidated Independent, a company which specialises in encoding music and submitting it to the growing number of on-line music stores. "And we have all discovered now that there's no future in pretending that a digital product is somehow the same as a CD, which almost everyone has up till now." Indeed, thanks to digital stores like iTunes, people can sample and often buy individual tracks, not just whole albums. More radically, the subscription model moves music purchasing beyond per-track or per-album pricing models to an all-you-can-eat service. Paul Sanders and Consolidated Independent are also involved in on-line music retailer Playlouder MSP, which offers a music subscription service of the latter type, in this case allied to a broadband ISP. Playlouder MSP has attracted many independent labels and musicians, but also has a deal with Sony/BMG. For Sanders, Playlouder is leading the way in an approach that will become much more common.

"As I see it, there's a growing realisation that applying a per-unit revenue model to on-line music leaves too big a gap to be enforced through legal action and copy-prevention technology," he observes. "The more appropriate model is a broadcast-style licence with royalties collected by either application or network providers, such as we're adopting with Playlouder."

Guitarist and composer Steve Hillage, who with his wife Miquette Giraudy operates as System 7, is a long-time user of the Internet. System 7 started putting up music clips on their site from the late '90s onwards, and began exploring the possibilities afforded by selling on-line downloads in 1999. "We've now got quite a clear strategy for selling our stuff as downloads" says Hillage. "We're putting our whole catalogue on line, including some exclusive stuff that's previously only been available on vinyl. CD singles are often uneconomic, but as a download people can burn it to CD or play it on their iPod. We're not going to stop releasing CDs and vinyl, we just see downloads as a new arrow to add to our quiver."

Paul Sanders of on-line music encoder and 'aggregator' Consolidated Independent.



Live Transmissions, the latest Steve Hillage album to be released under the name System 7, is available from a variety of sources, including the iTunes store and Steve's own web site.

Acknowledging that iTunes is the place to be for downloads, System 7 now have 14 albums available on Apple's Store, and recently released an iTunes-exclusive LP of remixes. "If you're not on iTunes, you're not digital" says Hillage. "You can quote me on that. It's had a significant effect on our revenues, though we're getting quite a lot of sales from Napster as well. In the past year we've signed up with IODA as our digital distributor, and I would highly recommend them to anybody. Within one month of signing up with IODA, our tracks were out in the USA. Pretty damn good!"

P2P, DRM, & Weed

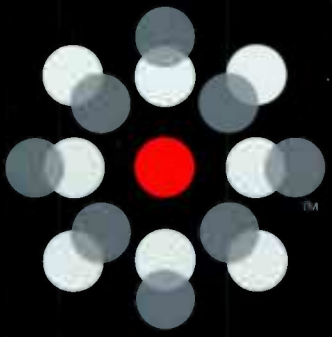
Peer-to-peer (or P2P) filesharing is the perennial thorn in the side of the majors, but indie musicians and labels have typically adopted a more relaxed attitude, preferring to see filesharing as free marketing. Steve Hillage: "People have been getting free music for nearly one hundred years. It's called a radio, you know what I mean? I don't think filesharing is a bad thing in moderation, because it helps you get your music heard. And everyone's happy when their music is played on the radio, so there's a certain amount of hypocrisy there. I try to have a balanced view of filesharing, because it's not going to go away."

Neil Leyton feels that filesharing is great for independent musicians. "Provided they're genuinely driven by cultural exchange and not a third-party moneymaker, these services should be allowed to exist. Anyway, filesharing is a fact of life, much as home taping was a fact of life in the '70s and '80s. It's cultural exchange, and artists should embrace it as such."

Leyton isn't suggesting that music should simply be free; selling more recordings is his ultimate goal. "If someone hears an MP3 file for free, they can then decide whether they want to buy it and support that artist. Not every transaction or exchange needs to be a financial one, nor do we believe that anyone is ripping our artists off by downloading their work. On the contrary, it's great promotion. If you're selling more records as a result of P2P, then it's just fine with us that those filesharers are not paying anything for the right to listen. Filesharing is *not* piracy, no matter how many major PR campaigns are launched to try and beat that into our social consciousness — and it allows people to sample more music than the radio can offer. Music fans are the life-blood of an artist's career. They are certainly not pirates, and if they want to hear music for free it is out of their appreciation of that artist's work. If they like it, they will continue to support and endorse that artist in

Useful Links

APPLE iTUNES MUSIC STORE W www.apple.com/itunes	CD BABY W www.cdbaby.com	GARAGE BAND W www.garageband.com	MYSPACE W http://music.myspace.com	RHAPSODY W www.rhapsody.com
AWAL UK W www.awal.co.uk	CI (CONSOLIDATED INDEPENDENT) W www.ci-info.com	IODA (INDEPENDENT ON-LINE DISTRIBUTION ALLIANCE) W www.iodalliance.com	NICK WEBB (SHOPSONIC/ FARFIELD RECORDS) W www.shopsonic.com	SHARED MEDIA LICENSING (WEEDSHARE) W www.weedshare.com
A WAVE (SYSTEM 7/ STEVE HILLAGE) W www.a-wave.com	EMUSIC W www.emusic.com	MAGNATUNE W www.magnatune.com	THE ORCHARD W www.theorchard.com	TOBY SLATER W www.tobyslater.com
BEATNIK W www.beatnik.com	FADING WAYS W www.fadingways.co.uk	MOBILE STREAMS W www.mobilestreams.com	PLAYLOUDER MSP W www.playloudermsp.com	WARP RECORDS W www.bleep.com



emusic®

Nº 1 Site for Independent Music

emusic.

Nº 1 Site for Independent Music

eMUSIC IS THE WORLD'S LEADING DIGITAL RETAILER OF INDEPENDENT MUSIC

- Number 2 Digital Music Retailer selling more than **4.5 MILLION SONGS** per month.
- With more than **1 MILLION TRACKS**, eMusic is the largest collection of independent music ever assembled.
- eMusic is the only major digital music service to provide music in the universally compatible **MP3** format; eMusic allows subscribers to transfer music to any desktop, portable music player or digital device, including the market-leading **iPOD®**.
- With more than 4.5 million tracks sold every month, eMusic provides labels with revenue they can count on. In fact, we generate **MORE ROYALTIES FOR INDIE LABELS** than MSN® + Yahoo!® + Napster® + Real® combined.
- Join eMusic for its **EUROPEAN LAUNCH** in 2006! eMusic will launch its service in the UK and the rest of the EU in the late Summer of 2006. Contact an eMusic representative for further details.
- Look for eMusic at **LONDON CALLING 2006!**

eMusic is a subscription-based service that offers acclaimed editorial content, a vibrant online community and music discovery tools, giving consumers the ability to explore great music they wouldn't otherwise find.

eMusic Europe Emma Collins / Label Relations, Europe / ecollins@emusic.com / +44. 7957.488.646

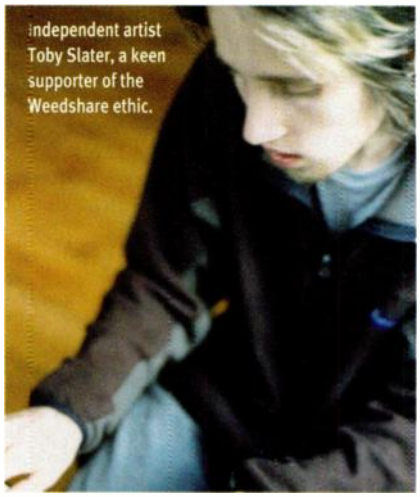
eMusic US Ray Farrell / Vice President, Content Acquisition / ray@emusic.com / +1. 212.201.9279

a number of different ways, from buying a record to wearing a T-shirt to going to a live concert."

DRM typically marks another distinction between major and indie approaches to selling music on line. Apple had to use DRM on its iTunes tracks (albeit a relatively light version) in order to get the major labels on board. But, as mentioned earlier, many indie sites sell DRM-free MP3s, seeing DRM as an unnecessary inconvenience. Another option, adopted on System 7's site, is to give fans the choice; DRM versions of tracks, at 79p each, are cheaper than the 99p non-DRM MP3s, reflecting DRM's inherent inconvenience. However, Steve Hillage says they're now moving to MP3 only, because the MP3 files have been outselling the DRM ones by a ratio of 15 to one, despite the latter's cheaper price. And the changes won't stop there. "Dance music downloads have blown up in the past year, and beatport.com, who are the leaders, have starting using WAVs" comments Hillage. "Once broadband speeds get a bit faster, MP3s are going to be history. It'll be WAVs or AIFFs, and that's the true nightmare of the record companies — or not. It depends how creative people in the music industry are."

"Generally, my feeling about DRM is that it penalises the people who purchase the track" says independent artist Toby Slater, an enthusiastic adopter of the Internet since the days of the original Napster. But not all DRM is created equal. Recently, Slater has found his music appearing as so-called Weed files thanks to a deal between CD Baby and Weed developers Shared Media Licensing. He points to the Weed system as a way in which DRM can work for artists and fans: "The fantastic thing about Weed is that it rewards the people who pay for tracks, and it rewards you for passing them on. That's the opposite of most DRM, which just limits you."

Weedshare is a system developed by Shared Media Licensing (SML) which challenges the assumptions of those who only see the negative side of DRM. Essentially, Weed uses a Windows DRM-based system to enable fans to be rewarded for sharing tracks by allowing them to share in the track revenue. The centrally-managed DRM limits the number of times an unbought track can be played (to three plays), and tracks the chain of purchases. Fifty percent of any purchase goes to the track's artist, 15 percent to SML, and the remainder goes to the sharers, divided 20, 10 and five percent through three stages. So if someone buys a Weed track that they got from you, you get 20 percent, while if someone then buys the track from them, you get 10 percent of that purchase, and on one further stage to where you get five percent. If a track costs you a pound, you get 20p, 10p and 5p respectively from every person who subsequently buys the track. How much money you get obviously depends on how many people download it from you in the first place and then buy it, and on what subsequently happens to the track. See www.weedshare.com/share/musicfans for a more detailed explanation.


Independent artist Toby Slater, a keen supporter of the Weedshare ethic.

The Shared Media Licensing web site (below), home of the Weedshare concept, and (right) co-founder John Leighton Beezer.

"It seems obvious to us that filesharing is, in effect, free promotion and distribution — so it can't be all bad," says Shared Media Licensing co-founder John Leighton Beezer. "That was the basis of the idea that it was OK to pay people to share. It was a friend of mine who proposed the idea to me; he called me up one day and said he'd figured out how to make DRM work for music. I was very sceptical at first, but he convinced me. We know that our files can be hacked, but we're trying to create a positive alternative to standard on-line distribution."

So-called Independent Content Providers (ICPs) are approved by Shared Media Licensing to do A&R, set up rightsholder accounts, verify rights, encode the music tracks, and work with artists to promote their music as Weed files if that's what they want. An ICP then gets all 35 percent when someone buys a track from their site, because they're the first provider. "In a lot of cases, record labels have an in-house ICP for working with us," says Beezer. Another feature of the Weedshare approach is that anyone can set up as an on-line music 'store' by hosting existing Weedshare tracks, and earn revenue through the 15/10/5 system. There's no need to set up an e-commerce capability, because SML handle all the sales and the distribution of revenues.

Site Inspections

Another site which typifies the open approach of selling MP3s is Magnatune, the on-line music retail site set up by John Buckman to find a better, fairer way to sell music after his wife fell foul of music industry dealings. Magnatune follow a largely traditional sales model, but don't lock files up in DRM. Buckman does non-exclusive deals with artists on a per-album basis and gives them 50 percent of each sale. When buying an album, people can choose how much they want to pay for it, from $5 to $18. Magnatune also encourage podcasting — because they make all of their 128k MP3s available under a Creative Commons licence, use of these files is 'podsafe' if they are podcasted non-commercially. In keeping with Magnatune's open attitude, people can also stream complete versions of tracks and albums from the label's web site at any time. There's even a well-developed licensing side, with a very open web-based licensing interface that allows potential licence-holders to see how much they'll have to pay for their desired use. One such use is for film soundtracks; the system allows filmmakers to try out Magnatune tracks for free and only pay for them when their film enters the

Music On Mobiles

While the Internet is maturing as a music sales and marketing medium, it's still early days for music in the mobile world. "Mobile music sales are still very much focused on the big seller, the chart hit, with the top 20 percent of overall content generating 80 percent of the total revenue" observes Steve Hayward of Mobile Streams, a company that sits between the music business and the mobile networks, liaising with both to make music available on your mobile. "But there is growing pressure for change. In the same way that the Internet music download market has given a new lease of life to jazz and classical music, I think we'll see mobile music diversify into more niche areas."

Heyward describes Mobile Streams as a media company with a production facility. As a formatting house, it has its own dedicated image-, video- and audio-formatting teams. "Our strategy is very much to work with multiple network operators in multiple territories. In the UK, we work closely with 3, Vodafone, O2 and Orange. Approaching the networks directly is pretty difficult; they're quite entrenched in terms of who they work with. And it's hard to reach the mobile mass market if you can't work with a network operator, so it's quite difficult for small independent labels; in fact it's hard even for the large indies. We work with Beggars, Sanctuary, Ministry of Sound, and a large number of independent music suppliers."

"The mobile network operators want a closed service, whilst it's in the best interests of the music owners and third-party service providers for there to be an open environment," says Jeremy Copp of mobile music-technology specialists Beatnik. "My belief is that in the interests of creating the biggest and fastest-growing market, mobile users need to be able to have access to music from a whole range of sources, not just those dictated by the operator."

So while the Internet is opening up opportunities for indie labels and unsigned musicians like never before, will they have to resign themselves to the fact that music for mobiles will remain the province of the major labels? "There's no reason that independents or unsigned artists cannot share in the opportunities that mobiles provide" Copp says. "The authoring of music for mobiles fits into the existing recording studio workflow, and is being made much easier due to the adoption of common, standard formats. So there are likely to be opportunities to distribute content either directly or through partnerships with operators, service providers or perhaps major labels." And other broadcasters may also provide an outlet, he adds. "I can imagine radio stations and TV channels picking up on mobile audio as a promotional channel, so those stations who today promote independent and unsigned artists on the Internet will be able to extend this to mobile phones."

commercial arena.

Buckman's feeling is that iTunes has stolen the show for the consumer side of the business ("the lack of any major non-DRM site means there isn't a real contender to iTunes"), and he's doubtful that any music service dealing in mainstream genres will make any headway against them. As for the most promising developments for the independent sector, Buckman sees these in the new marketing possibilities in smaller markets, "be it on Myspace, by artists self-releasing on iTunes through someone like IODA, or the new music sources such as the Magnatune and Garageband sites."

Nick Webb first discovered the value of the the Internet for selling CDs and downloads back in the late '90s. He began selling CDs from his own label, Farfield Records, and other ambient labels in 1998, and his URL, www.ambientmusic.co.uk, became a leading destination for those in search of ambient and chillout music. Meanwhile, he did deals with on-line music services such as Emusic, Rhapsody and Wippit. Early last year, he set up a new on-line site, Shopsonic, to sell CDs *and* album downloads.

The Magnatune web site (left), featuring the excellent mission statement 'We Are Not Evil', and (below) founder John Buckman.

Webb sees on-line CD retailing as the way forward for small labels like Farfield and the various other labels whose catalogue he stocks. "Getting your CDs into the shops isn't necessarily going to get you anywhere. You've got to have a budget to advertise nationally, and with a lot of large companies, even if they do stock something from an independent, it will be tucked away at the back of the shop where no-one can find it. Having the ability to sell downloads in the ▶

same shopping basket as CDs is quite unique to our site and allows us to offer free sampler downloads and albums when customers are making a CD purchase." Perhaps unsurprisingly, download sales have been increasing since Shopsonic agreed a joint on-line licence with the UK's MCPS & PRS music-licensing bodies in April 2005. According to Webb, downloads now account for approximately 40 percent of music sales, and of those, 30 percent are from the USA. He's definitely of the opinion that downloads have helped to bring in new customers to his site from overseas.

"Bricks-and-mortar distributors can make it very difficult for labels like mine," says Fine Arts Militia co-founder, Zenstone record-label owner and bassist Bryan Hardgroove, who leads Public Enemy's live band, "but the Internet provides a distribution channel. If you mass-produce records and put them in a store, then you have a loss if you don't sell them. But if you can expose people to your music via the Internet, you can see how much people like it in whatever region you're interested in, and then you can move records where you know they're going to be wanted."

Opportunities (Let's Make Lots Of Money)

Steve Hillage says he's 50/50 about on-line music as an opportunity for independent labels and musicians: "It's something that's important to do, and obviously idealistically it's a fantastic way of bypassing all forms of corporate

On-line retailer and distributor Shopsonic, brainchild of aptly-named musician Nick Webb.

AWAL's Top 10 Presentation Tips

AWAL UK specialise in getting independent music onto on-line music stores such as iTunes. Paul Bower is one of their UK team. "When you offer your music to an on-line distributor, presentation is just as important as if you were offering the music for sale on CD" he says. "Remember, you're entering the music business; thinking about a few simple points will help you to be taken more seriously."

1) CONTACT DETAILS
"Don't just send a CD-R with a scribbled email address. At the very least it should have the artist, title, and copyright information written neatly on the disc. Better still, invest in a thermal or inkjet disc printer."

2) POSTAGE
"Check your postage. A CD in a jiffy bag will weigh just over 100g, so putting a single first-class stamp on the package will result in your recipient having to collect the package from the sorting office and pay a one-pound fine."

3) PACKAGING
"If your CD artwork is printed on a laser or inkjet printer, try to include a CD-ROM of the original artwork, as these rarely scan successfully. And if you send in scanned artwork, always clean the artwork and your scanner before you scan. If the scanned artwork is not absolutely straight, even to within a tenth of a degree, it will look odd when reduced down to a thumbnail. To remedy this, scan the artwork with a border to allow subsequent readjustment."

How to present a CD to an on-line distributor (above) and (right) how *not* to present it!

4) MASTERING & MEDIA
"You are treating your music as a saleable commodity, so you shouldn't send it off until you are absolutely happy with it; this includes the mastering side of things. Similarly, do not send out multiple disks for the recipient to compile."

5) BIOGRAPHY & FURTHER INFORMATION
"Always include a biography in your package, and type out any technical information such as copyright details, ISRC numbers, and track titles if they don't already appear on the sleeve."

6) RIGHTS
"Make sure you have the rights to enter into the agreement! If you've previously signed a digital distribution deal, or your music has previously been released on CD, check your contract. You did keep a copy, didn't you...?"

7) TAKE THE INITIATIVE
"After initial contact, keep in touch by phone or email to push the deal through."

8) NO MP3s OVER EMAIL
"Don't email MP3s unless you're invited to."

9) CLEARANCES
"Make sure you have cleared — or removed — any samples in your recordings, and obtained permission from session players and featured artists to use their performances."

10) NO CROSSFADES
"If your album contains a lot of crossfades, think about recompiling a download version without them."

distribution and getting your music directly to the people that want it, which is a major plus. But as I said, it has to be seen as another arrow in your quiver."

Others are more convinced. "We're at a great time right now," opines Toby Slater. "There's access to the on-line stores, so you as an independent artist or label can get your stuff onto iTunes and Rhapsody and all those services pretty easily. You could sign up with someone like The Orchard or with CD Baby, and within a month or so you'll probably have your stuff up there and selling, which is really great."

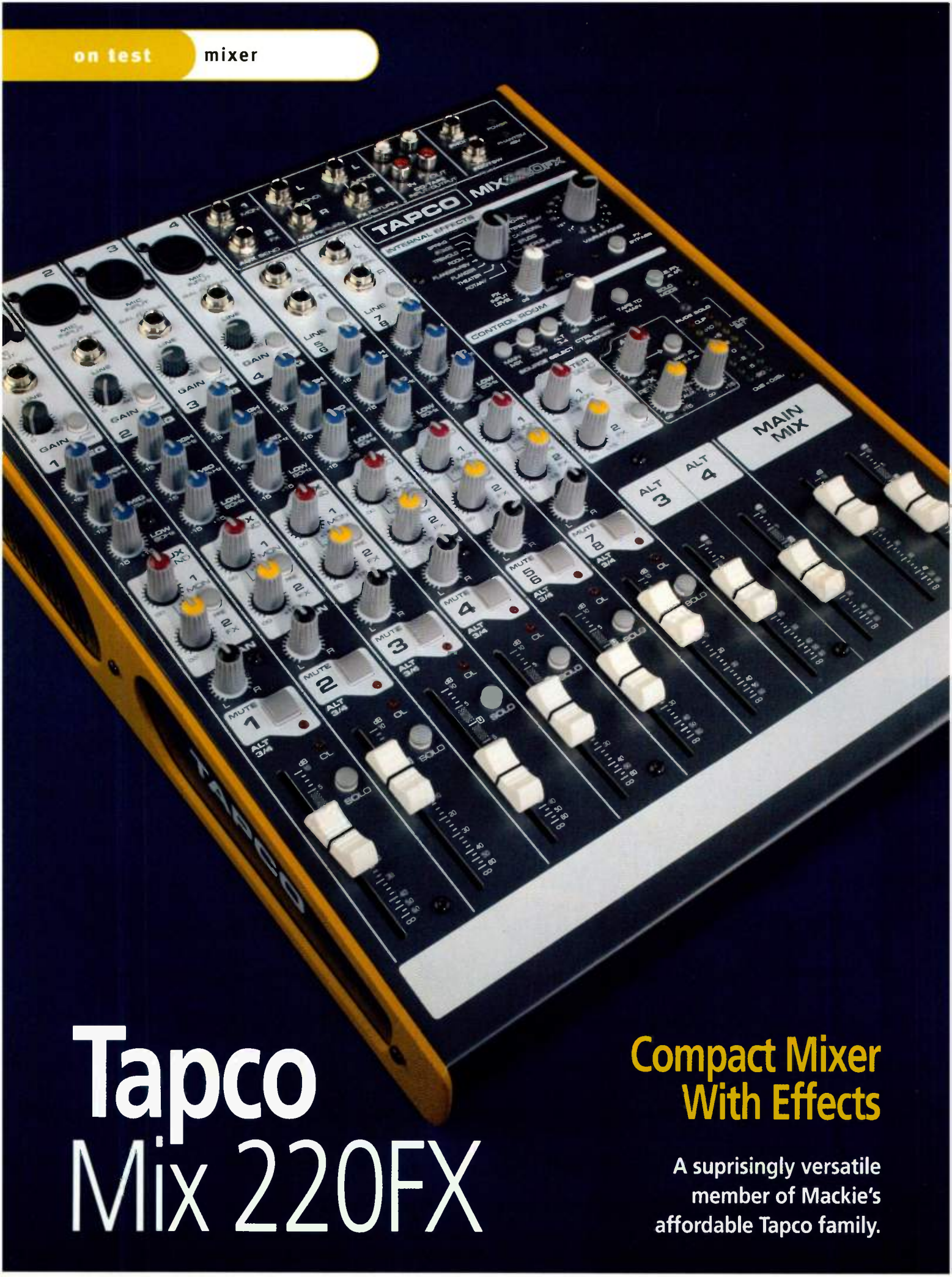
Consolidated Independent's Paul Sanders says he remains "infinitely optimistic" about the opportunities for independent labels and musicians, but that it's important to keep a sense of realism as well. "We have to keep working hard, but technology is opening a marketplace in which anyone who has created music can participate on the same basic footing, if not on the same commercial terms. When you're in the middle of such rapid change as we are now, there are all sorts of opportunities. The key to making the most of them is staying positive. If you can't get played on the radio, make your own station, and if you can't get into the shops, join a musicians' retail web site. The technology gets easier and more effective every year."

Fine Arts Militia bassist and Zenstone label owner Bryan Hardgroove.

"I'm optimistic," says Bryan Hardgroove. "The only way you can stop a smart independent is if some sort of legislation appears, some sort of worldwide restriction. At the moment, the success of on-line record sales is as available as your willingness to work. If you have good ideas, good talent and a good marketing plan, it doesn't take a whole lot of money."

"I think that we're in a window of opportunity right now for indies and artists," comments Fading Ways' Neil Leyton, "but I think the golden era is still to come. We're 10 years behind the open source movement, but I believe, as others do, that the curve will be about the same in terms of growth and the opening up of markets. True independents, creative thinkers, will thrive."

Indeed, in contrast to the limited opportunities for exposure and sales in the traditional world of physical distribution, many opportunities are now opening up for independent artists and labels to market and sell their music on line. With new marketing avenues such as web logs and podcasting allied to companies that can help musicians get their music into a wide range of web-based music stores, the independent sector has many reasons to be excited about the Internet. SOS



Tapco Mix 220FX

Compact Mixer With Effects

A suprisingly versatile member of Mackie's affordable Tapco family.

Paul White

In today's computer-based recording studio, the large-format mixing console is becoming something of an endangered species, but there are still numerous situations in which a smaller mixer can earn its keep. It's often cheaper to buy a small mixer than to buy separate mic preamps, and a suitable small mixer also makes an excellent input-source selector and master monitor controller.

For those not yet in the know, Tapco is the name of Greg Mackie's original mixer company and has now been adopted as the name of a separate brand within the Loud Technologies group, to offer a cheaper alternative to their Mackie range. Tapco's Mix 220FX is designed to appeal to studio musicians and live performers looking for a low-cost, compact mixer with a flexible range of features.

Overview

The 220FX measures just 292 x 360 x 96 mm (WxHxD) and weighs in at 4kg. It offers four mic/line input channels with globally switchable phantom power, plus two line-only stereo input channels. In addition to these eight input channels, there are also two sets of stereo return jacks and a stereo CD/Tape input. Like some early Mackie mixers, the Mix 220FX has four output busses with a button on each channel to route the signal to either the main left and right outputs or the Alt 3/4 outputs. This button also doubles as a channel mute if you don't happen to need the Alt 3/4 outputs. One important factor here is that each of the four mic channels can be routed separately to each of the four mixer outputs simply by using the pan knobs and routing buttons, so if you need four mic feeds to drive a four-input, line-level audio interface, this will do the job nicely.

Another treat the mixer has in store is its built-in 20-bit digital multi-effects section. A 16-position rotary knob selects the effect, while a second knob chooses one of the 16 variations provided for each effect, giving a total of 256 presets. Many of these are sensibly chosen reverb and delay treatments. The rest provide modulation effects, including rotary-speaker emulation, chorus and flanging. There are also two banks of chorus-plus-reverb and chorus-plus-delay effects.

The Mix 200FX's master section features stereo CD/Tape inputs and outputs on RCA phonos with monitor-source switching, so you could record into a computer interface via the input section of the mixer and monitor back via the CD/Tape returns at the same time. As with almost all such mixers, you also get a decent headphone outlet so you can save yourself the cost of a separate headphone amplifier too.

SOUND ON SOUND

Tapco Mix 220FX £160

pros

- Sensibly priced.
- Good technical spec.
- Subjectively good effects.
- Comprehensive routing and monitoring facilities.

cons

- There's no separate channel mute facility if you use both the main and Alt outputs.

summary

The Mix 220FX has just the right feature set to attract both the home studio user wanting a mixer to double as a four-channel mic preamp and monitor controller, and the live performer needing a small mixer with integral effects.

To keep the cost down, the mixer is designed in the US but built in China and features a robust, all-metal case with bright yellow side panels that remind me of a Meccano construction kit! All the main outputs are on the rear panel. The main mix is on balanced XLRs, while the Alt 3/4 outs and control room outs are on quarter-inch jacks that can be used balanced or unbalanced, with a maximum output level of 22dBu. The sockets look durable, and power comes in via a proper mains lead rather than via an external mains adaptor. A switch-mode power supply automatically adapts to mains voltages from 100 to 250 Volts. The mains power and phantom power switches are also on the rear panel (with status LEDs on the top panel), while all other connectors are at the upper edge of the top panel. For use in high-risk areas, there's a slot on the rear panel that accepts a Kensington security chain.

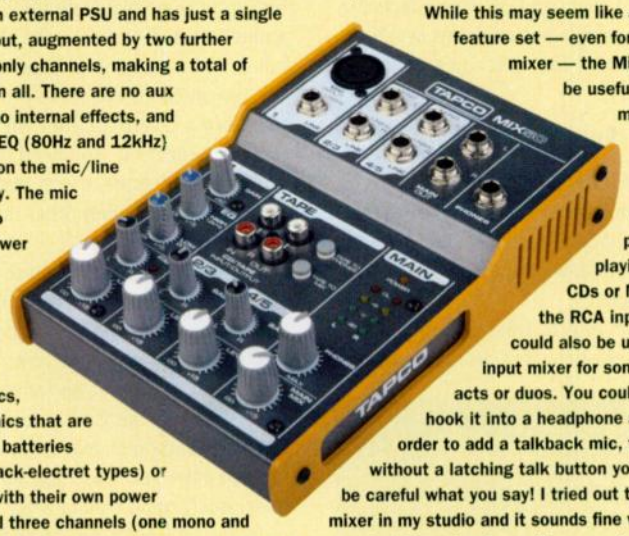
Channel Facilities

Each of the four mic/line channels has both balanced XLR and jack inputs with a rotary gain trim control and a switchable 75Hz low-cut filter. This may be a budget mixer, but the mic preamps have a very wide audio bandwidth, extending from 10Hz up to 200kHz (-3dB) with the gain trim at zero (this will inevitably be reduced as more gain is applied). The equivalent input noise is an impressive -129dBu (A-weighted) and both the crosstalk and distortion figures are sensibly low. These might not be esoteric preamps but they turn in a good, workmanlike performance.

The EQ on all channels is a fixed frequency, three-band system offering 15dB of boost or cut at 60Hz, 2.5kHz and 12kHz, and while there is no EQ bypass, the controls all have centre detents. Aux 1 may be switched pre- or post-fader for each channel while the post-fade Aux 2 feeds both the internal effects processor and the ▶

Tapco Mix 50

If your mixer requirements are more basic and you're on the lookout for something that's about the size and price of a small box of (admittedly very good quality) chocolates, the Tapco Mix 50 (around £45) might be what you need. This mixer runs from an external PSU and has just a single mic/line input, augmented by two further stereo line-only channels, making a total of five inputs in all. There are no aux sends and no internal effects, and a two-band EQ (80Hz and 12kHz) is provided on the mic/line channel only. The mic input has no phantom power and so is really suitable only for use with dynamic mics, capacitor mics that are powered by batteries (typically back-electret types) or tube mics with their own power supplies. All three channels (one mono and two stereo) have pan and level knobs and the master section comprises one main output-level control plus one headphone level control feeding a quarter-inch stereo output jack. There are also stereo CD/Tape inputs and outputs on RCA connectors with 'tape to phones' and 'tape to mix' buttons, as well as a stereo, four-segment output-level meter.



While this may seem like a very basic feature set — even for a basic mixer — the Mix 50 could be useful as a small monitor level controller for active speakers, with the possibility of playing back CDs or MP3s via the RCA inputs. It could also be used as an input mixer for some solo live acts or duos. You could even hook it into a headphone amp in order to add a talkback mic, though without a latching talk button you have to be careful what you say! I tried out this little mixer in my studio and it sounds fine with a typical dynamic mic, so if you need quality but little in the way of quantity, it's a practical and very affordable option.

TAPCO MIX 220FX

Aux Send 2 jack, in case you want to use an external effects device. Pan controls are located at the bottom of the channel strip, just above the 60mm fader, and there are Alt 3/4 and channel Solo buttons. The line-only channels have two jack inputs per channel, with individual +4dBu or-10dBV level selection buttons but no gain trims and no low-cut filters.

The Solo buttons solo channels over the control room and headphone outputs, but do not affect the main outputs. You can even choose AFL or PFL solo mode, which is unusual on such an economical mixer. AFL monitors the channel level after the channel fader, while PFL monitors the level prior to the fader and so is not affected by fader position. LEDs in each channel act as overload and routing indicators. The flashing 'Rude Solo' LED in the master section comes on when the Solo button is down, and solo'd levels are shown on the main eight-section LED meters.

In the master section, where you'll find the main and Alt 3/4 faders, are the two mono Aux Send jacks and two sets of stereo Aux Return jacks: send and return levels are controlled by knobs in the master section. Aux 1's return may also be routed to the main mix or the Alt 3/4 mix, while the Aux 2 return, normally fed to the on-board effects, can be mixed into the Aux 1 send for monitoring. The control room output and headphone output share the same level control and source-selection buttons, used to choose between the main mix, the two-track input and the Alt 3/4 output. Another button sends the tape input to the main mix, which is useful for using backing tracks live or for plugging in CD or MP3 players. Above the master controls is the effects section, with its two selector knobs and input-level control (with adjacent overload LED). The effects can be bypassed using a button in this section, or by connecting a footswitch to a jack at the top of the panel.

Summing Up

I've used this little mixer to drive a PA system at a pub gig and also in the studio. In both cases its audio quality was subjectively good, with very low noise and no nasty surprises.

The built-in effects are a definite plus point. I've been somewhat disappointed by the effects sections in some budget mixers, but here you have a choice of decent reverbs that are good enough to use for home recording as well as for live performance. There are some nice patches, including some plausible small room ambiences and plate reverbs. Also, the mixer's routing options mean that you can use the on-board effects to provide monitoring reverb for your vocalist without having to record it. You need to take care not to overdrive the effects section (watch the overload LED), but other than that it's as simple as it gets.

I particularly like the fact that Tapco give you proper control over control-room source switching and both send and return levels: not every manufacturer is so thoughtful. By way of performance, I think there's little to choose between this mixer and the standard Mackie mixers that preceded the VLZ Pro range, so it's capable of seriously good results when used properly. Of course, any fixed three-band EQ is a bit limited, but this one works well for adding some gentle high-end sheen or low-end warmth to tracks, while the mid band is useful for taking out harshness or honkiness.

If you need a few mic preamps, a monitor controller and headphone amp, then the Mix 220FX will fit the bill, with the added bonus that the on-board effects can be used while monitoring but are also good enough to work in a mix. The compact format of the mixer, combined with its main and alternate outputs, makes it suitable for live recording into a laptop equipped with a suitable audio interface, and for duos or small bands wanting a mixer to drive their PA system it should do a great job. There's no shortage of low-cost mixers vying for your money, but this model has a good pedigree and it's clear that a lot of thought went into its design. SOS

Alternatives

Similarly-priced alternatives include the Alesis Multimix 12FX and the Behringer UB1202FX, both of which feature digital effects. If your budget will stretch a little further, the Soundcraft Spirit FX8 is worth a closer look. It's also a four-buss mixer, but it offers twice as many mic preamps and it has built-in Lexicon digital effects.

information

£ Mix 220FX £163.33; Mix 50 £45.63. Prices include VAT.
T Mackie UK +44 (0)1268 571212.
F +44 (0)1268 570809.
E uk@mackie.com
W www.tapcoworld.com

Create your perfect home studio

Powered by Red Submarine computer music systems

gear4music.com

The example laptop systems below can be configured to your requirements and budget. You can use our online configuration tool, or call our technical sales team for friendly advice and to discuss your needs in more detail. All our systems are designed to be compatible with a wide range of music and audio software - just log onto Gear4music.com to configure your ideal setup. You'll be creating great computer music in no time at all - secure in the knowledge that your system is supported by Red Submarine, one of the most respected names in the industry.

15" Intel Centrino Audio Laptop

SPECIFICATION:
- 15" XGA Screen (1024x768)
- 128Mb Shared Graphics
- 1.6GHz Mobile Processor
- 512Mb DDRII RAM
- 40Gb Hard Drive
- DVD/CD-RW Drive
- Windows XP Home
- Ghost recovery

from: **£816.00** inc VAT

15.4" Centrino with E-Mu 1616

SPECIFICATION:
- 15.4" WXGA Screen (1280x800)
- 128Mb NVidia Graphics
- 1.7GHz Mobile Processor
- 512Mb DDRII RAM
- 40Gb Hard Drive
- DVD/CD-RW Drive
- Windows XP Home
- Ghost recovery
- E-Mu 1616 + Cubase SE3

from: **£1362.00** inc VAT

17" Core DUO Audio Laptop

SPECIFICATION:
- 17" WXGA Screen (1440x900)
- 256Mb nVidia 7900 Graphics
- 1.83GHz Core DUO CPU
- 1Gb DDRII RAM
- 60Gb SATA Hard Drive
- DVD/CD-RW Combo Drive
- Windows XP Pro
- Ghost Recovery

from: **£1615.05** inc VAT

15.4" Core Duo Fujitsu Laptop

SPECIFICATION:
- 15.4" Widescreen (1920x1200)
- 256Mb ATI FireGL Graphics
- 1.66GHz T2300 Processor
- 512Mb DDRII RAM
- 60Gb Hard Drive
- DVD/CD-RW Drive
- WiFi & Spillproof keyboard
- Modular design fits a second drive or second battery.
- Optional port replicator

from: **£1937.00** inc VAT

SYSTEMS CAN INCLUDE: Wireless and gigabit LAN, 56K modem, bluetooth module, spare batteries, original Windows disks, 3 years warranty and lifetime email and telephone support. You can add an audio interface, audio sequencer and audio editor, midi controller, microphones, furniture and more online.

:: 64 bit Audio Workstation ::

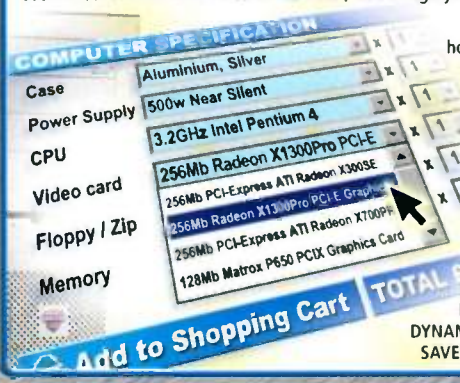
This powerful and quiet system is ready to take your music to the next level. Capable of running Dual Core CPU's and large amounts of RAM, up to 8Gb. The benefits of running a 64 bit system cannot be ignored, particularly with large projects typically found in post production or orchestral compositions. Now is the time to upgrade your audio system, configure your suite with us today.

SPECIFICATIONS:
- Intel Bad Axe 975 Motherboard
- 3.2GHz Intel Pentium D 940 4Mb L2 Cache
- 4Gb DDR2 533MHz RAM
- NEC DVD Burner
- 120Gb Program Drive
- 250Gb Media Drive
- 256Mb ATI Graphics
- Windows X64 Pro
- Sonar 5 Producer + E-Mu 0404

from: **£1899.00** inc. VAT

:: Design your PC online ::

Use our state-of-the-art 'CONFIGURATOR', and design your own audio solution online.



Desktop, laptop or complete home studio: it's easy to add or remove items from the drop down menus and select the hottest products available.

Add soundcards, keyboards, microphones... and even furniture! - we will build, install, test and deliver.

FEATURES:
REAL-TIME PRICE UPDATES • COMPATIBILITY CHECKER • INTELLIGENT MENU SYSTEM • DYNAMIC INFORMATION WINDOW • SAVE SYSTEMS AND EDIT QUOTES•

Novation ReMOTE ZeRO SL

Need more manipulation for your sound? Ideal for live or studio use, this gives you fast and flexible control with your software.

£209.00

Korg Kontrol 49 Keyboard

Get one of the most versatile controllers around this Autumn at this fantastic price. 49 Keys and 40 further controllers in one simple unit.

£224.00

Native Instruments KORE

New for 2006 KORE brings you more control to your VST's, ideal for producers, DJ's and live performances.

£339.00

RME Fireface 400

New for 2006 get two high quality Mic Preamps and the most accurate digital audio for all your recording needs

£649.00

Focusrite Saffire Pro 26 I/O

With 8 Focusrite preamps, MIDI, ADAT, word clock and combined with the unrivalled Saffire plugins make this the hottest soundcard around

£469.00

Digidesign M-Box 2

Excellent Mic Preamps and now with MIDI, this portable interface is ideal for your Pro Tools sessions anywhere you go

£299.99

Propellerheads Reason version 3.0

All-in-one sequencer, synths, sampler, drums and more

£244.00

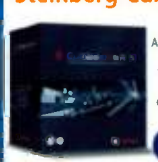
Cakewalk Sonar Producer 5

Sonar 5 Producer is the first 64bit capable sequencer. Get your audio system ready for the future now

£329.00

Steinberg Cubase SX 3 (PC/Mac)

Advanced new features and state-of-the-art technologies establish Cubase SX3 as a quantum leap forward in music production



£495.00

TC Powercore Compact

Now your favourite DSP plugins are portable. This lightweight firewire interface comes with 12 plugins, increasing your sound processing tools wherever you are.

£389.00

All prices are inclusive of VAT and are correct at time of going to press, E&OE.

Gear4music.com is *the* musicians online superstore

Selling over 4000 quality music making products, including mixers, microphones, PA systems, soundcards and software, plus a wide range of great value musical instruments.

www.gear4music.com Tel: 0871 309 0800 email: info@gear4music.com

Sound & Vision

Making A Living From Music For Picture: Part 9

Bill Lacey

Rather appropriately, we round off this series by looking at what you need to consider when composing music designed to play while the end credits roll...

Last month I introduced a documentary film for television, *Dinka Diaries*, and made three scenes available for you to score. In this, the final instalment of this series, we'll look at how I scored these scenes, and consider music for the closing credits of programmes and films. I'll also try to explain the importance of backing up your work, and the significance of stem mixes of your finished scores.

Mission Accomplished

One of the most challenging aspects of composing using a computer is to have your tracks sound as if they were played by real musicians (assuming you are playing all the parts yourself with virtual instruments, which I do most of the time). I avoid quantisation whenever possible, which goes a long way towards meeting that goal. In Part 7 of this series, I discussed using tempo changes and markers to assist in accommodating picture edits. However, there are times when it can be far more effective to rely on the old-fashioned approach of simply playing along with the video. And that's just what I did with the first of the three scenes I introduced last month (you can download the finished scene with music at www.soundonsound.com/sos/soundandvision/dinkaphotosafterfinal.mov).

I established last month that the director of *Dinka Diaries*, Filmon Mebrahtu, wished to use a nylon string guitar and a string orchestra. Having thought up some harmonic changes, I rehearsed my guitar part while viewing the picture. It was important that I didn't play too busy a part, as I didn't want to distract from the dialogue. By closely following the picture and listening to the dialogue, I was able to create a part that flows in a way that would be difficult to create with a click track or tempo map. The slight tempo variations add a distinctive human element. Once I had recorded this part while playing along with the picture, I loaded strings from the Vienna Symphonic Library (VSL) *Opus 1* collection into Native Instruments' *Kontakt* and played to the tempo established by the guitar.

The second scene required two acoustic guitars to 'converse', played in the style of the late North African blues guitarist Ali Farka Touré. This was the most challenging

Classical Inspiration

This series has given you some suggestions about contemporary film composers to listen to, as well as the great masters who developed the art form. But who do the great film composers listen to? Where do they get *their* inspiration? Typically, they listen to works of the great classical composers. And while it's not unusual for classical music to be used in films and on television, let's not forget that many classical composers wrote specifically for the medium, among them William Walton (*Henry V*), Ralph Vaughan Williams (*Scott Of The Antarctic*), Malcolm Arnold (*Bridge Over The River Kwai*), Dmitri Shostakovich (*The Fall Of Berlin*), John Corigliano (*Altered States*) and Aaron Copland (*Of Mice And Men*).

If you like Samuel Barber's *Adagio For Strings* (used, of course, in Oliver Stone's film *Platoon*), I think you'll love *Musica Celestis* by Aaron Jay Kernis. Kernis is a gifted composer whose symphonic works seem very 'cinematic' — to me, at least. And if you're a fan of Ridley Scott's *Alien*, as I am, you probably loved Jerry Goldsmith's masterful soundtrack. The music that accompanies the climactic scene where the alien is blown out into space at the end of the film, however, is not Goldsmith's music, but the end of the first movement of Howard Hanson's second symphony (also called *Romantic*). The music continues on through the closing credits, and it's hard to believe Hanson didn't write the music just for this film, as it works perfectly.

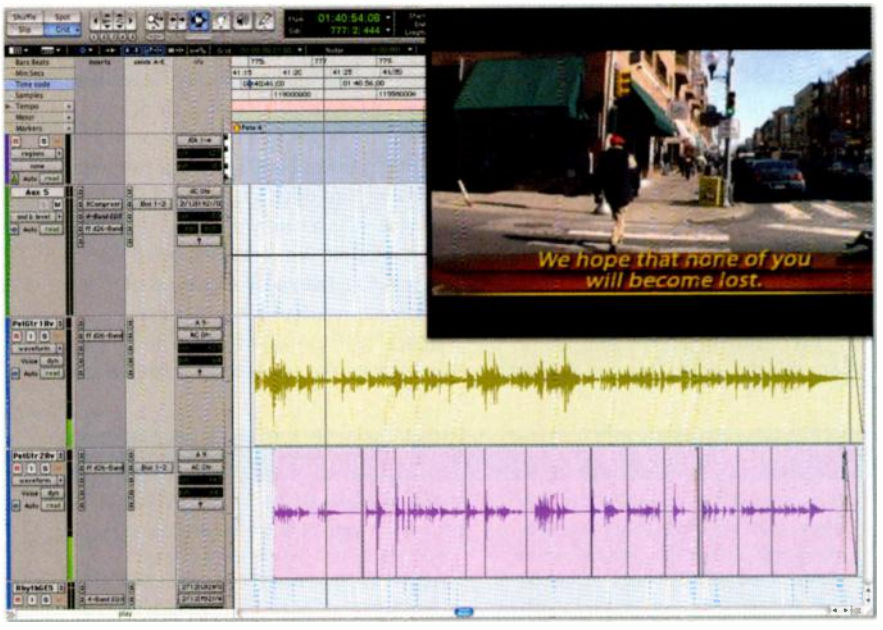
Besides the obvious composers to listen to (Mahler, Tchaikovsky, Strauss, Ravel, Debussy, Holst, and Wagner were all highly imaginative and adept at creating evocative music that lends itself well to accompanying film, which is perhaps why their works are used so often), consider the music of 20th-century composers such as John Adams, Henryk Gorecki, Béla Bartók, Philip Glass, Jennifer Higdon, Jean Sibelius, Igor Stravinsky, Charles Ives, Libby Larsen and Alban Berg (particularly the latter's violin concerto).

piece for me — I dread to think what it must have been like for those of you who are not guitar players — and I found it hard to get it 'just right'. As with the first piece, I played the guitar while viewing the picture and listening to the dialogue. This was actually the fourth of five scenes in the documentary where music accompanies one of the youths while they are listening to a Sudanese village elder on headphones. I recorded a number of different ideas on the guitar for Filmon to choose from, built from thematic riffs that I had already established by this point in the scoring process. Once I had the first guitar recorded, I proceeded to play the second part. Listening carefully to both the dialogue and guitar, I played a track that complemented the first part, featuring more improvisation. I recorded a number of takes of the improvisation, and as you can see from the screenshot on the right, the final track was edited together from several of these takes. You can download the finished scene from the film, with its completed music, at www.soundonsound.com/sos/soundandvision/villageelderafterfinal.mov.

The third and final scene featured a map, and like the other map scenes in the film, this one needed some rhythmic movement. I turned to East West's *Ra*, which offers a number of authentic African percussion instruments to choose from. The first instrument loaded was an African metal shaker, which provided the pulse. Next I added a kalimba, which served as a melodic element as well as reinforcing the pulse. I also added a xylophone from my VSL *Opus 1* library to double the kalimba part, fading it in and out of the track. This added a thickening to the kalimba part that I liked. Finally, with the percussion in place, I loaded up the string orchestra from the *Opus 1* library. It's interesting to note that as the Kalimba is playing an ostinato part, it helps to serve as a neutral foundation and a balance against the strings, which adds a more serious tone to the maps. Depending on the instructions from the director, I am always careful not to inject my personal feelings into the score. And that's not easy when the subject matter is controversial. In this case, I was able to strike the right balance between the seriousness of the situation being discussed and the neutral intent of the director. You can hear the final clip at www.soundonsound.com/sos/soundandvision/dinkamapafterfinal.mov.

Upon completion of the score, Filmon asked me to provide stem mixes, so that he could control the balance between the strings, the guitars, and the percussion. To learn more about this, see the box about stems on the next page of this article.

Every scoring project will have its own set of circumstances, and *Dinka Diaries* proved to be a significant challenge. But sometimes the most rewarding projects are the ones that challenge us the most, and in the process, teach us something we didn't previously know anything about. In the end, the director was extremely happy with the music, which is all that really matters, and I certainly learned something new by having to explore the music of Ali Farka Touré.



The finished musical arrangement for the 'Village Elder' section of *Dinka Diaries*, as seen in *Pro Tools*. Note the lines denoting the edits in the second guitar part, where it has been assembled from multiple takes.

Here Today...

Recently a fellow composer called me with some very bad news. His main audio drive had failed, and he had not backed it up in over a year or more, so virtually all the audio from his recent projects was now lost. This was a hardware failure, so available software fixes were ineffective, as he could not mount the drive on the desktop. It may seem hard to believe that he had not made any backups over such a long period of time, but the story is true, and he has learned a painful lesson. It remains to be seen whether or not all of his files can be recovered. The only course of action he can attempt now is to take the drive to a data-recovery service — and of course, there is no guarantee that he will successfully recover his data at the end of it. What's more, data recovery is an extremely expensive option. You literally don't want to go there.

If you want to make a living working in film and TV, you'll need to adapt some backup strategies to prevent such a disaster from happening to you. Archiving of data is an essential part of a professional workflow; clients will not tolerate delays, and wholesale loss of data can rapidly lead to the loss of a job.

Of course, backing up isn't very easy (which is perhaps why so many people don't do it). File sizes for musical projects can be rather large, and Quicktime video files only add to that. To give you an example, the size of my *Pro Tools* project folder for *Dinka Diaries* was in excess of 10GB, and the video files were 13GB. Managing files of this size can be

I didn't get where I am today without... lots of backup drives. But just to prove that no-one's perfect, as I write this, I'm trying to recover some lost data from a crashed drive. And of course, as luck would have it, the drive that's crashed is the only one that *wasn't* fully backed up...

a tall order, and you need to consider many factors, such as how much suitable backup hardware will cost, and how long it will take to back up your current projects with it, before you make your final decision.

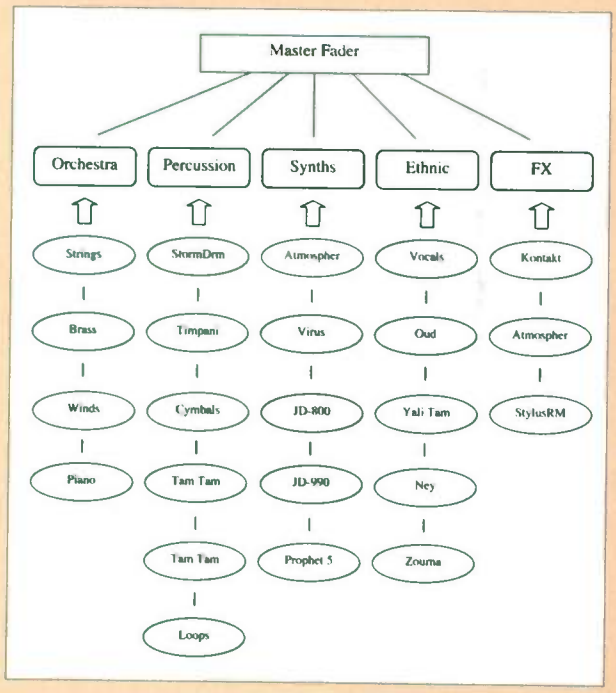
The following strategy works well for me (but your mileage, as ever, may vary). The quickest way to back up data at the end of a session is to copy files to a spare external Firewire drive, and fortunately, the cost of such drives is fairly reasonable these days. My first ever hard drive had a capacity of 30MB (yes, that's Megabytes, not Gigabytes...) and it cost me 800 dollars (about £450). These days, you can't even get RAM that small any more. In the USA, the current cost is around a dollar per Gigabyte of external Firewire hard drive space, but this is just as well — as a hard-working music-for-picture composer, you'll need access to *hundreds* of Gigabytes of storage, probably split across many drives, to house all the variations on your work that you may need to create, to say

The Importance Of Stems

In the film and television world, it is often required that in addition to providing your clients with a stereo mix, you also provide stems, or split mixes. These are premixed files, often in stereo, that usually contain all of the instruments of a certain type (for example strings, drums, or guitars) to facilitate greater flexibility when mixing. However, the exact configuration can change depending on the type of music, its complexity, and the requirements of the post-production audio mixer. When called upon to provide stems, I usually make separate stereo mixes of percussion instruments, symphonic instruments, synths and finally sound-design elements. For the film *Ghost Soldier* (covered in parts 7 and 8 of this series), I also added a separate stem of ethnic instruments (see the diagram below).

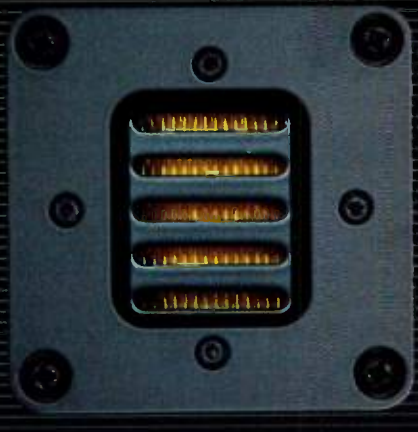
As you can probably imagine, generating all of these mixes can be time consuming. So why do it? The goal is to give the audio mix engineer the ultimate in flexibility when balancing music with dialogue, ambience and effects. Be sure to have what is known in movie-speak as a 'two beep' (a 1KHz tone lasting the duration of one frame of video), or at least a count-off, at the beginning of each stem mix, so that the mix engineer can precisely synchronise all of your stem files. At the very least, make sure your stems all start at exactly the same frame!

At the beginning of a scoring session, I'll often set up Aux channels to accommodate the stems. Sometimes I'll even create sub-stems that are routed to the main stem. For instance, I often create separate sub-stems for brass, strings and woodwinds, which are all routed to the main orchestra stem. Another advantage of configuring a session this way is that it makes it easy to control the overall level of your session. Reaching for the master fader and pulling that down is not always the best solution; but when your session is set up in the way I've described, it takes just a few moments to drop a *Trim* plug-in across all of your stems and pull everything back a decibel or two. One final note of caution: be sure to set up your reverb and effects so that they are routed to the appropriate stems. I typically use separate reverbs for each stem. You can get around this by bouncing each stem separately with effects, but this is very time consuming and you lose flexibility. In *Pro Tools*, I'm able to bounce all of my stems to individual audio tracks at the same time.



Audio stems supplied to the director of *Ghost Soldier*, the film mentioned in Parts 6 and 7 of this series, to allow for greater control when mixing the soundtrack.

ADAM
PROFESSIONAL AUDIO



The ADAM A.R.T. Tweeter
Accelerated Ribbon Technology

The A.R.T. principle achieves a 4:1 velocity transformation between driving diaphragm and driven air due to the unique folded diaphragm, i. e. the air moves in and out four times faster than the folds itself are moving. The results are incredible clarity and transient reproduction. No other ribbon tweeter is designed this way or sounds this good.

A7

ADAM Audio shatters price barrier with A7

ADAM Audio A7 two way active nearfield combines A.R.T. (Accelerated Ribbon Technology) tweeter with a state of the art 6.5" carbon fiber sandwich woofer that features high rigidity and internal damping with low weight, resulting in a fast reacting woofer. This teamed with the now legendary A.R.T folded ribbon tweeter results in an extremely accurate monitor with all the clarity, details and spectacular imaging traditionally associated with ADAM Audio.

- Two 50 watt amps (one per driver)
- 46Hz - 35kHz +/- 3dB frequency response
- 105dB SPL @ 1 metre
- Front panel: stepped level control & power switch
- Rear: tweeter level, 150Hz & 6kHz frequency controls
- XLR balanced and phono unbalanced connectors
- 2 year warranty

- Oh, and they're £667.00 a pair inc VAT !

THRILLING EARS AROUND THE WORLD

UNITY AUDIO LTD

Tel: 01440 785843 | Fax: 01440 785845
sales@unityaudio.co.uk | www.unityaudio.co.uk

Boss RT20

Rotary Speaker Effects Pedal

The latest box of tricks from Boss is a pedal-based Leslie speaker simulator with some cool flashing lights.

Paul White

Rotary speakers, or more commonly, systems of rotating baffles placed over fixed speakers, were originally developed for use with the electric organ, but before long, more experimental bands were also using them on guitar and vocals. Recreating this unique sound, which involves variations in level, timbre and pitch, is not easy, and for this latest attempt Roland's Boss division have used their COSM technology to model the characteristics of the rotary speaker in great detail. But what most people will notice first is the wonderful swirling LED display at the centre of the front panel!

Features

In true Boss tradition, the robust pedal is made from cast metal. It can be powered either from six AA batteries or from an optional external power adaptor. All the connections are on the rear edge of the unit. Both the input (switchable to suit guitar or keyboard) and output can be in mono or stereo. There's also a headphone output, and you can plug in an expression pedal to control the effect's speed directly.

The left pedal turns the effect on or off and the right pedal switches between slow and fast speeds. Pressing both pedals together activates the brake, imitating the effect of bringing the rotors to a halt over the period of two or three seconds. The right pedal LED blinks when the brake is active, and stomping on the speed pedal starts the rotary effect again.

There are four effect modes, three of which are rotary speaker emulations while the fourth is inspired by the Univibe pedal made famous by Jimi Hendrix. Mode 1 is designed to sound like a Leslie 122, including that model's overdrive characteristics, while Mode 2 replicates a close-miked version of the same unit to give an enhanced modulation effect. Mode 3 combines the rotary speaker of the first two models with a 1959 Marshall guitar amp, offering more distortion. Finally, there's the Univibe setting, again combined with the guitar amp distortion of Mode 3.

Other controls adjust how long it takes the effect to speed up and slow down when the speed is changed, the level of the effected and direct signals, the bass/horn balance, the slow and fast rotary speeds

SOUND ON SOUND

Boss RT20 £169

pros
- Excellent controllability.
- That cool display!
- Very musical sound.
- Four different useful modes.

cons
- Some noise when in operation. While this can help add to the authenticity, I'd have liked the option to do away with it if it was added deliberately.

summary

The RT20 puts good-quality rotary speaker emulation into a practical pedal format that should work well for guitarists and keyboard players alike, both live and in the studio. It combines flexibility of operation with ease of use and has what is probably the most hypnotic visual display in the history of effect pedal technology.

and the amount of distortion, from off to fully saturated. The central elliptical display comprises multiple red and blue LEDs that are sequenced to provide a sense of bass and horn rotation.

Turning Heads?

The first thing you notice about a real Leslie is that it reproduces no real high end at all, with the treble horn carrying mainly mid-range information with a lot of tonal coloration. The same is true of the RT20, which reshapes the high end very much like the real thing. In my view the dynamics of the speed change feel pretty good, as does the brake option, and being able to change the horn/bass balance gives you a reasonable amount of control over tonality. The effect works nicely both in stereo and mono. In fact, it seems to lose very little for being used in mono. I'm not entirely convinced that it exactly captures the more subtle nuances of the watery shimmer that a real Leslie produces but that's hard to judge — with the real thing much depends on the room you record it in and on how you mike it up. However, the RT20 is certainly warm and musical and sounds good on both guitar and keyboards. As a gentle and more musical alternative to a slow chorus or phaser, the slow setting is very hard to beat. The Univibe setting gives a lovely soft phasey chorus, with the balance knob acting as a depth control.

I did notice that the unit produced a noticeable amount of noise that also gets processed via the rotary effect, but this dies away smoothly as your input sound decays. The noise does add an organic (no pun intended) quality to the sound and it may have been added intentionally to simulate the wind and mechanical noise of the real thing, but if that is the case, I'd have liked the option of turning it off!

Having adjustable distortion is very useful, both for guitar and keyboards. With keyboards it gives a nice low-end growl but can also tend to make the sound a bit muddy. As promised, the guitar-amp-style distortion of Mode 3 and 4 is a bit more far-reaching but still very dense at the lower frequencies.

Impressions

While there are software emulations and effects built into keyboards and amps that sound to my ears just as authentic as the RT20, having everything in a pedal really is a bonus in terms of controllability, especially when playing live. It isn't particularly cheap, but the RT20 lets the user tailor the effect to a useful extent and switch speeds and brake using pedals. Its replication of the mid-range tonal coloration of a Leslie speaker is also probably more accurate than most, while the addition of a very sweet Univibe emulation and built-in distortion is icing on the cake. There's also the advantage that the RT20 can process mono or stereo signals, producing a mono or stereo output, and it's switchable to suit guitar or keyboard sources. Using the keyboard setting, it's possible to process line-level sources, so you could use it in the studio for treating already-recorded tracks. While the purists will continue to argue about the authenticity of all such units, I know which I'd rather to carry up the fire escape to a gig! SOS

information

£	£169 including VAT.
T	Roland UK +44 (0)1792 515020.
F	+44 (0)1792 799644.
W	www.roland.co.uk
W	www.bosscorp.co.jp


DUENDE

SSL Duende

DSP Plug-in Processor

This sleek silver box promises to bring all the mixing power of a 32-channel SSL console into your computer recording setup.

Paul White

SOUND ON SOUND

SSL Duende £999

Photos: Mark Ewing

pros

- Sensibly priced.
- Both the *Channel Strip* and *Bus Compressor* plug-ins are excellent.
- Thirty-two plug-ins always available, except at high sample rates when 16 are available (stereo counts as two plug-ins).
- Easy to use.

cons

- Mac-only at present.
- Currently, only one Duende can be used at a time.
- Some limitations of the original have been retained when they could have been avoided.

summary

When it comes down to it, 32 channels of real SSL EQ and dynamics is seriously attractive. It's just a pity that PC users will have to wait a little longer for it.

Few musicians need an introduction to SSL mixing consoles, as they've been an integral part of the high-end studio scene for as long as most of us can remember. As the project studio world has expanded, SSL have moved into that market too, with products derived from their flagship consoles. We've already seen their XLogic analogue outboard gear, based on elements of their console channel strips, but the new Duende sees SSL entering the DSP-powered DAW plug-in arena alongside TC Electronic's Powercore, Universal Audio's UAD1, Waves' APA units and the recently launched Focusrite Liquid Mix.

Duende went into development in 2005, with the aim of recreating the EQ and dynamics processing sections of an SSL console as plug-ins that could be hosted on a rackmounting processor unit, hooked up to your computer via Firewire. Audio Units and VST plug-in formats are supported, and the plug-ins can also be used in *Pro Tools* thanks to an included version of FXpansion's *VST To RTAS Adapter* wrapper software. The Mac OS version of Duende is already shipping, and a Windows version is expected in the autumn. Duende makes 32 channels of EQ and dynamics available simultaneously at sample rates up to 48kHz, with half that at 88.2 and 96 kHz, and you also get an SSL *Bus Compressor* plug-in recreating the hardware unit that is a must-have tool for many professional mix engineers. Four SHARC-based DSPs provide the power for Duende, utilising 40-bit floating-point maths.

But wait a minute, surely any digital plug-ins purporting to be SSL channel strips must simply be emulations, because the big SSL consoles are analogue, right? The answer isn't as straightforward as it seems, because SSL also make the large-format C-series digital consoles, which run digital emulations of their analogue channel strips. However, these consoles are recognised as having a sound in their own right, and to complicate things further, Duende doesn't run exactly the same algorithms as the C-series console either, because SSL's engineers revised and tweaked the C-series algorithms to get them even closer to the sound of SSL's analogue consoles in the course of developing Duende — which presumably leaves them wondering whether they need to update the C-series console software to keep up!

And why the weird name? Apparently 'duende' is a Spanish word meaning 'spirit' and the SSL guys liked it.

Way To Go, Duende

The Duende hardware is an elegant 1U box with little more than a power switch and a pair of rear-panel Firewire (400) connectors. In most cases it can be buss-powered, though a PSU that adapts to the local voltage is included for situations where buss power is not available. However, the real face of Duende is in its plug-in windows. Rather than providing a suite of different plug-ins, Duende ships with just two — *Channel Strip* and *Bus Compressor* — though more may be added in the future.

Within the plug-in window, parameters are adjusted by dragging and clicking the controls in the usual way and a small diagram at the bottom of the page shows the way the channel-strip elements are routed. A black box at the top of the channel strip displays the current parameter value, which will also show up above individual controls during adjustment in software version 1.1 onwards (due for release as I'm finishing this review). As you'd expect, plug-in parameters can be automated to the extent that plug-in automation is supported by the host application, and all the plug-in settings are saved in your host DAW's song file in the normal way.

A Preferences panel shows information about the Duende hardware and also displays DSP loading information so you can see how much free DSP you have available at any time. There are 32 slots available (16 at high sample rates) and each slot may be filled with either a *Channel Strip* or a *Bus Compressor*. Note, however, that a stereo version of a plug-in takes up two slots, and each half of a stereo plug-in must run on the same one of the four DSP chips, rather than being spread over two chips. If you end up with just two slots left on different DSP chips and you need to load in another stereo instance, closing and then reloading the song should remedy the ▶

SSL DUENDE

problem, as the DSP resources are reallocated when a song opens.

EQ And Dynamics

The most commonly used of Duende's two plug-ins is likely to be *Channel Strip*, which is based around the characteristics of the EQ and dynamics sections of SSL's XL9000K console. *Channel Strip* is set out as a horizontal version of what you'd find on one of their hardware consoles, and its features include input filtering, an expander/gate, a compressor, low and high-pass filters and a four-band EQ.

The filters comprise a 12dB/octave low-pass filter and an 18dB/octave high-pass filter with cutoff frequencies continuously variable from 20Hz to 500Hz and 3kHz to 22kHz respectively. Both can be bypassed and the filters can be switched out of the audio path and into the dynamics side-chain if required. Additionally, the filters can be switched pre- or post- the main EQ. Filters of this kind are very useful for constraining the frequency range of tracks when you're trying to create separation in a mix. A routing block diagram is always shown in the plug-in window, so you can always check that the signal is being routed as you expect.

A four-band equaliser handles the main EQ functions, and like the XL9000K EQ, has two distinct switchable characteristics based on the SSL G and E-series designs. The four EQ bands comprise high and low-frequency shelving equalisers, which can also be switched to peaking curves using a Bell button (which I personally find a very useful option), and two variable-Q, variable-frequency mid-range parametric sections. The lower of the shelving bands has a range of 40 to 600 Hz, with ±16.5dB of gain available, while the higher works from 1.5 to 22 kHz and offers a generous 20dB of cut or boost. The two parametric bands offer ranges from 200Hz to 2kHz and 600Hz to 7kHz, with Q adjustable from 0.5 to 2.5. Again, ±20dB of gain is available.

In the G-series mode, the shelving curves EQ exhibit some overshoot, like their analogue counterparts, while switching to E-series mode gives a gentler slope with no overshoot. In G-series mode, the bandwidth of the parametric bands varies with the amount of cut or boost applied, whereas the E-series version operates in constant-Q mode; this may be perceived as being less subtle, and you have a definite choice of EQ flavours here. It's also possible to bypass the EQ section of the plug-in independently of the rest, or place the EQ in the dynamics side-chain. If you've already assigned the filters to the side-chain, the EQ comes before the filters in the signal path.

The dynamics section of the channel strip is relatively simple but nevertheless very powerful. Compression can sound different depending whether it is applied pre- or post-EQ so there a Pre EQ switch to

Latency & System Load

One major concern about DSP-assisted Firewire devices is the inevitable increase in latency caused by having to send the audio out of the computer, process it and then send it back. Duende is no exception to the general rule that your host software's audio driver latency will be roughly doubled. This isn't an issue when mixing but may mean it is best to record your session, including any overdubs, before opening up any Duende (or other DSP-powered) plug-ins. As Duende's plug-ins are intended for mixing, unlike soft synths and some other plug-ins, this isn't really a problem, but it is something you need to be aware of, and if SSL could add a control to bypass all of Duende's plug-ins at once for those occasions where you need to come back and do an overdub, that would be a great help.

Most host applications automatically compensate for plug-in latency providing the plug-in correctly publishes its delay characteristics to the host software, so all your tracks retain the correct time relationships with each other. Providing PDC is switched on, the process is transparent to the user. Note, however, that not all host applications offer delay compensation for sends, group busses or outputs, and in the case of programs like *Logic*, you have the option to switch this on or off — so you should ensure it is switched on if you are using plug-ins in buss insert points. (This is of course true of any plug-in, but as DSP-powered plug-ins tend to double the latency set by your original buffer size selection, the result can be very noticeable if you forget!) *Pro Tools LE* 7.1 and earlier does not feature PDC so any compensation has to be done manually.

Currently there is no expander option to allow 32-channel operation at high sample rates, and multiple Duende units are not supported, but SSL are actively researching ways around this limitation. I also did some tests to see how much CPU load Duende placed on the host processor; some administrative load is inevitable, but on a Mac G5 the extra load is pretty much insignificant — probably around the same as running one or two native compressor plug-ins.

SHOW US YOUR BEST STUFF.
WE'LL SHOW IT TO THE LONDON SYMPHONY ORCHESTRA.

Enter the NOTION REALIZE MUSIC CHALLENGE for a chance to win A RECORDING SESSION WITH THE LSO AT ABBEY ROAD STUDIOS!

It's time to put those years of study, your love of music, and that computer to good use.

You could win a trip to London to watch your original score being recorded by the London Symphony Orchestra, your own copy of the award-winning NOTION software, plus $2500 in cash. All you'll need to enter is the free NOTION demo software, now available at **www.notionmusic.com/win**

It's time to show each other what we've got.

To enter and for contest details, visit:
WWW.NOTIONMUSIC.COM/WIN

NOTION™
MUSIC COMPOSITION AND PERFORMANCE SOFTWARE

provide this choice as well as a bypass button. When the dynamics is switched pre-EQ, it still comes after the input high- and low-pass filters.

Featuring a continuously variable ratio from 1:1 to infinity (hard limiting), the compressor/limiter has a threshold control adjustable from -20dB to +10dB, a choice of auto or fixed 1ms attack times, and a release time variable from 0.1 to 4 seconds. There's also a choice of detection algorithms. Choosing Peak means the compressor responds to high signal levels regardless of how short they are in duration, while RMS makes the compressor respond more to the average input signal level as it would be perceived by a human listener. As a rule, peak compression is more assertive and is well suited to percussive material. In this design, the RMS mode uses a soft-knee compression curve whereas the Peak setting reverts to a conventional hard knee, so the RMS mode is clearly the more gentle of the two. A meter with yellow and red 'LEDs' shows the amount of gain reduction being applied.

Expander/gates are always useful for mopping up noise in pauses, and the one here features an attenuation range variable from 0 to 40 dB, a threshold variable from -30dB to +10dB, switchable normal or fast attack times and a hold control variable up to 4 seconds. At the fast attack time, it takes 0.1ms to reach 40dB attenuation, while the normal attack time is 1.5ms per 40dB. Hold is an important control as it prevents the gate 'chattering' when the audio envelope fluctuates, as it can otherwise do if a fast release time is used. The release time can be set anywhere from from 0.1 to 4 seconds.

Because the filters and equaliser can be inserted into the dynamics side-chain, frequency-conscious gating is possible. Green 'LEDs' show the amount of gain reduction applied when the gate or expander is closed. In expander mode, the expansion ratio is fixed at 1:2. At the output of the dynamics section is a ±20dB output gain control (to compensate for any gain loss in the compressor), and a six-segment level meter. A side-chain listen button enables the user to hear the side-chain signal complete with any filtering or EQ that has been applied.

Magic Buss

The *Bus Compressor* plug-in is definitely Duende's secret weapon and is, again, derived from the buss compressor used in the XL9000 K-series console. It is available in stereo or mono versions and may be deployed anywhere, though it is commonly used for processing subgroups or complete

stereo mixes. Unlike the real thing, however, you can open multiple instances up to the maximum of 32 DSP slots. There's a meter to show gain reduction (3 or 4 dB is usually enough!) while the threshold can be varied from -20dB to +20dB. As with the original, the attack time is switchable in six steps from 0.1 to 30 ms, while the release is also switchable, this time from 0.1 to 1.2 seconds in five steps. Make-up gain compensates for compressor gain reduction and is adjustable from -5dB to +15dB. A Compressor In button acts as a bypass. There may be relatively few controls, but the compressor itself is based on a very complex circuit that has a particular character and adds density and punch without sounding unmusical. This buss compressor is often favoured by hip-hop producers, but it is applicable to many types of pop and rock music, and depending on how it is set up, it can add subtle body to a mix or really make it pump.

Alternatives

You could buy a TC Powercore plus the Sony *Oxford EQ* and *Oxford Dynamics* plug-ins, which offer the same calibre of performance, but they aren't the same thing as SSL algorithms. Similarly, the Focusrite Liquid Mix will give you a greater choice of different EQs and compressors, but if you want genuine SSL, then Duende is the only game in town.

Duende In The Studio

Duende installed without fuss in my system, and a firmware updater on the CD-ROM was used to bring the hardware up to the current spec. It ended up sharing a Firewire port with my TC Powercore, which was also being used in the test song, and I didn't encounter any problems there, successfully running all 32 channels at the same time as a fair number of Powercore plug-ins while also streaming audio from an external Firewire drive, though this was connected to a different port on my G5. If you mix 'in the box' rather than streaming multiple audio channels out to an analogue mixer, therefore, you may well get away with using your computer's built-in Firewire ports. However, Firewire bandwidth isn't infinite, so it might be a good plan to add an additional multi-port Firewire card to your computer if you have a free PCI slot, especially if you also need to communicate with external hard drives or other DSP processing boxes.

When I spoke to the SSL engineers, they also pointed out that unlike some DSP boxes, version 1.0 of Duende isn't able to handle off-line bounces, so mixes need to be rendered in real time. I checked this out and they're absolutely right — if you try to work off-line you get glitches. Apparently software version v1.1 will allow off-line bouncing, although this will still be no faster than real-time operation. While offering no speed advantage, it will at least avoid inadvertently creating files with glitches in them!

I tried out the *Bus Compressor* plug-in first as part of a Mix Rescue project I was doing and it worked a treat, fattening the mix, adding lots of loudness and somehow blending the various elements without losing clarity or focus. If you push the ratio and amount of gain reduction up you can definitely hear it working, but in a way that is still musically useful for many styles, so you get plenty of creative leeway with this particular plug-in. I also used it on a second remix project involving a very nice jazz outfit where more subtle settings were in order, and again, it helped knit the mix together without getting in the way of the music.

When I came to try the channel strip I was impressed by how predictably everything worked, making it fast and easy to get the job done. Both EQ options sound 'right' and you only need to apply small amounts of cut or boost to make the required changes to the sound, just like

with a good analogue equaliser. The high and low-cut filters are also extremely useful, though the designers may have missed a trick by only allowing you to put these in the side-chain for the whole dynamics section, rather than allowing you to filter just the compressor or just the gate side-chain. This may be a feature of the original but there's nothing to say that software shouldn't be more flexible than the original in areas where it doesn't detract from the authenticity of the sound. Other than that, the EQ sections behaved beautifully, and you don't seem to have to fight to get the right results. The only slight disappointment for me was that the low-mid parametric band only goes down to 200Hz, and I often find there are low-mid problems that need addressing just below this frequency, so a bit more range would have been appreciated. In an ideal world, I'd have liked this band to go down another octave and the upper mid to go up by a further octave.

Though it has relatively few controls, the channel-strip compressor works really well in both its 'stealth' RMS mode and more overt peak mode. Whether you need subtle gain control or the effect of something being audibly knocked into shape, you've got it. Indeed, there are few if any applications where I'd feel the need to resort to a different compressor other than for ducking (as there's no way to source the side-chain from a different audio track). Whether you are levelling vocals, adding punch to drums or adding foundation to a bass guitar, this compressor makes a good job of it without any fuss whatsoever and without making the sound dull or 'spongey'.

Bargain Or What?

At just short of £1000, Duende might look like a lot to pay for a couple of plug-ins, even ones that can be run with virtually no CPU hit. Looked at more logically, though, it gives you the essential functionality and tonal character of 32 channels of SSL console to use inside your own DAW, for less than the cost of the VAT on a hardware SSL power supply — or so I'd imagine! The audio quality of these plug-ins is first-rate, and because they are based on the algorithms already used in SSL's own digital consoles, they're more than just emulations that don't quite hit the mark. I've pointed out a few limitations inherent in the originals that could have been worked around in the software versions, but there's plenty of time for SSL to issue updates if they feel the need. As it stands, these plug-ins do pretty much what you can do on a C-series SSL console, and in my book, the ease with which they get the job done is a major bonus. You can spend all day twiddling the knobs on an indifferent EQ without getting the results you're after, but here it all seems effortless.

If you like the SSL sound but are never likely to be in a position to own one of their consoles, you can now have all the sonic benefits from within your own Mac-based DAW at a very affordable price. I've only been playing with it for a few days and already I'm hooked. SOS

information

- £ £999 including VAT.
- T Sound Technology +44 (0)1462 480000.
- F +44 (0)1462 480800.
- E info@soundtech.co.uk
- W www.soundtech.co.uk
- W www.solidstatelogic.co.uk

Digidesign Icon

Hardware Control Surfaces For Pro Tools

Icon is Digidesign's collective name for the D-Control (above) and D-Command surfaces.

Digidesign's D-Command and D-Control represent perhaps the most ambitious attempt yet to integrate recording and mixing systems. Do these Icons symbolise the future of digital audio workstations, or are they merely expensive control surfaces for Pro Tools?

Mark Wherry

For those interested in etymology, the word 'icon' came via Latin from the Greek word 'eikon', meaning a likeness, or image, and originally from 'eikenai', an Indo-European word meaning 'to be like'. Without wishing to seem too pretentious, Icon is therefore a rather appropriate name for a system that can be used like a large-format mixing console, without actually being a large-format mixing console.

Icon (which was actually derived from the term 'Integrated CONsole') is the general product name given to the combination of a number of Digidesign's products that can be used together as one integrated system for recording, editing and mixing. At the heart of Icon are Pro Tools HD, with its associated array of DSP cards, audio and MIDI interface options, and the Pre eight-channel preamp; but the most significant part of an Icon system is either a D-Control or a D-Command control surface. The D-Control has the appearance of a large-format mixing console, while the D-Command is intended for the project studio and is the most obvious replacement for the company's previous flagship control surface, the Pro Control.

For this review, we'll be focusing mostly on the D-Command and D-Control surfaces, as Pro Tools HD has been pretty well covered in previous issues of *SOS*. If you want to read a closer examination of the audio interfaces and general architecture, check out Hugh Robjohns's original Pro Tools HD review in May 2002's issue (www.soundonsound.com/sos/may02/articles/protoolshd.asp).

Anatomy Of An Icon

Like many high-end digital consoles, the D-Control surface is modular, and a basic system consists of one Main Unit, a Fader Module, and the XMON monitoring system (see the 'XMON' box for more information about this component). The Main Unit contains all the global controls, such as master controls for the control strip, sections for controlling dynamics and EQ plug-ins, a remote control for XMON, buttons accessing general *Pro Tools* shortcuts, and a full QWERTY keyboard and trackball (which can be replaced with a mouse pad if you wish) for working with *Pro Tools*.

The Main Unit's meter bridge has eight LED level strips for metering the master output in up to 7.1 surround configurations, and there are multiple LED segment displays to show the Project time and locator positions. In front of the meter bridge is a large flat area to accommodate a VESA-compliant monitor arm that can suspend an LCD screen in front of the console, which makes the *Pro Tools* display really accessible. This can easily support a

23-inch monitor; Apple's 30-inch Cinema Display might be a problem, but you probably won't want to use such a big monitor in this position for acoustical reasons.

The Fader Module contains the actual channel strips, and each Fader Module offers 16 channel strips with stereo metering for every channel on the meter bridge. Up to five Fader Modules can be used with one Main Unit, giving you up to 80 faders, and I'm sure this would look quite impressive!

The D-Command is designed to provide similar functionality to the D-Control, but in a smaller unit at lower cost. It retains the basic elements, such as dynamics and EQ sections, monitoring, channel strips, XMON and so on, but provides fewer physical controls. Unlike the D-Control, the D-Command Main Unit includes eight faders. An optional D-Command-specific Fader Module provides a further 16 faders, to give 24 in total, but you can only use one Fader Module with each D-Command Main Unit.

Overview

An Icon surface shows as being off-line until you open a Session in *Pro Tools*, whereupon it springs (quite literally, thanks to the motorised faders) to life. The first area of the surface you might look at is the Transport section, which basically provides a way of controlling the transport in *Pro Tools*, including a scrub/shuttle wheel. The

The D-Command uses slightly cheaper faders compared to the D-Control, but still offers the same functionality in terms of touch-sensitivity and motorisation. The top five LEDs by the fader indicate automation status for that channel, the two below show whether the channel is grouped or in Custom Fader mode, and the Auto Match arrow LEDs indicate whether a fader is above or below the level of existing automation.

D-Control also provides a duplicate set of these basic transport commands for controlling the transport of an external machine, although this wasn't implemented at the time of writing. There's a set of Transport Mode buttons that let you enable and disable whether *Pro Tools* is on-line, loop playback, set punch modes, and so on, and you'll also find a group of Locate button for commands such as Return To Zero.

In addition to transport commands, the Master Unit also offers quite a number of basic *Pro Tools* shortcuts, such as a Zoom/Navigate section that enables you to zoom horizontally or vertically, access zoom presets, and navigate to different Tracks and Regions in a Project. The Window Management buttons (12 on D-Control, nine on D-Command) open and close various windows in *Pro Tools*, of which the common ones are the MIDI Event, Workspace, Transport, Memory Locations, Mix and Edit windows, while the Session Management section contains a handful of global Session controls, such as Save, Undo, Redo and Publish, the last being useful if you want to send your Session as a DigiDelivery.

One of the key areas on the Main Unit is the Soft Keys section (see photo below). This is pretty much identical on the D-Control and D-Command, featuring six main displays, each comprising two LCD display strips, and six corresponding switches. The Soft Keys section is used to set various preferences for the surface, in addition to providing access to commands in *Pro Tools*. For example, you can create and work with Tracks, Groups and Playlists or trigger various editing commands via the Soft Keys, or page through and select different Memory Locations, with the names of the Memory Locations being shown on the Soft Key displays.

Channelling The Strip

There are three different types of channel strips used on the Icon surfaces, starting with the 'Channel Strips' that represent the on-screen channels of the *Pro Tools* mixer. These are the Channel Strips you get more of when purchasing Fader Modules.

The second type of channel strip is the Channel Strip Master section, which contains global controls that affect the Channel Strips and other related parameters. For example, this is where you'll find the Solo and Mute Clear buttons. There's also a handy Escape key that mirrors the Escape key on the computer's keyboard, which is useful for cancelling out of dialogue boxes that might appear, and an Automation Suspend button to temporarily suspend the reading and writing of automation data. The D-Control features two Channel Strip Master sections, one on ▶

The Soft Keys section on the D-Control provides access to many different parameters for setting both various surface preferences and triggering commands in *Pro Tools*. Here you can see the preferences for setting the number and alignment of the left and right banks of Custom Faders.

SOUND ON SOUND

Digidesign Icon

pros

- A large-format or project-studio console that doesn't just integrate with *Pro Tools*: it is *Pro Tools*.
- The control surface will seem instantly familiar to *Pro Tools* users, as many aspects of its layout and implementation have been derived from the existing user interface.
- Working with plug-ins via Custom Faders and the dedicated EQ and Dynamics sections is really well thought-out.

cons

- A few operations you might usually carry out on a console while a track is playing can't be done with the Icon unless *Pro Tools* is stopped.
- There isn't a really good way to use multiple Icon systems in one large configuration.
- As with many other digital consoles, running large projects at 88.2kHz could be tricky with the current Pro Tools HD Accel architecture.

summary

Digidesign's Icon is a comprehensive system that, in its first incarnation, sets a new standard for integration between consoles and digital audio workstations. If you're already a *Pro Tools* user looking for a high-end project-studio controller or large-format surface, the D-Command and D-Control are just what you've been waiting for.

each side of the Main Unit, which makes the controls accessible no matter where you place the Fader Modules; the D-Command has just one in the middle of the surface, to the right of the Channel Strips.

Finally, the third type of channel strip is the Focus Channel Strip, which is basically a channel that allows more detailed tweaking than is possible on a standard Channel Strip. You'll see how this ties into other features on the Icon surfaces later in the review. Both the D-Command and D-Control use the Focus Channel concept, but only the D-Control has a dedicated physical Channel Strip just for the Focus Channel.

At the bottom of each Channel Strip is a touch-sensitive, motorised fader that has a range of -infinity to +12dB. The D-Control uses high-quality Penny & Giles faders, whereas the D-Command offers slightly cheaper faders. I didn't mind the faders on the D-Command, despite the fact that they didn't feel quite as smooth as the D-Control's faders, but it's always worth checking out the feel of any surface before you make a purchase.

Next to the faders is a set of LED indicators displaying the status of automation (Write, Touch, Latch, Trim, Read and Auto Match) and grouping for the channel, and showing whether that strip is in Custom Fader mode (which we'll look at later). Auto Match is Digidesign's term for the up/down LEDs that indicate whether the fader needs to move up or down to match the automation recorded for a channel at a given point in the Session.

Above the fader are the Channel Strip function controls, which is where you'll find the Input Monitor, Record Enable, Mute, Solo, Select and Automation Mode buttons for each channel. These work just as the on-screen versions in *Pro Tools*' mixer, with one exception. The Automation Mode of a channel is set by toggling the Automation Mode button rather than using a pop-up menu, though there's also a dedicated Trim button for selecting the Trim Automation Mode.

When a parameter controlled by an encoder is armed for automation, the Automation LED beneath the encoder is illuminated. With the appropriate mode enabled, parameters can be armed for automation by simply touching the relevant encoders.

In order to make it easier to use the modifier keys normally found on the computer's keyboard (Shift, Ctrl, Option/Start and Command/Alt), the modifier keys are duplicated at the bottom right of the Main Unit and each additional Fader Module. This is obviously convenient, and these work in exactly the same way with controls on Icon surfaces as they do in the software.

One small detail I found disappointing, though, is that there's no visual indication on the surface of whether a channel is in Solo Safe mode. A blinking solo light would probably have been annoying, but if the Solo

The XMON Monitoring Controller

The XMON is a 2U monitoring controller that looks exactly like a 192I/O box without anything on the front panel except for the power switch, a mute button and indicator (to mute all audio output), and a MIDI activity light to show when the XMON receives control data from the surface. Both D-Control and D-Command contain controls that can select inputs and outputs, make level adjustments, and so on, for an attached XMON. It's important to note that this monitoring system operates independently of *Pro Tools*, meaning that the monitoring part of the console is always active even if a *Pro Tools* Session (and thus mixer) isn't open. This is useful if you have a CD player connected through the XMON, for example, but it does mean that XMON settings aren't stored in a *Pro Tools* Session and the XMON itself can't interact with *Pro Tools*.

The XMON audio signal path is purely analogue, so you'll need converters and interfaces, such as Digidesign's own 192I/Os, to get audio out of *Pro Tools* into the XMON. On the back of XMON are four D-Sub connectors for inputs: a Main input for your main stereo or surround output from *Pro Tools*, a Surround input for a second stereo or surround input such as a CD/DVD player, a Cue input that provides three stereo cue feeds into XMON (useful for headphone monitor mixes), and, finally, a Stereo input that provides four additional stereo inputs. On the monitoring section of the Main Unit, you can select any of these inputs (with the exception that D-Command can only address three stereo inputs from the Stereo input) as the input source, or you can sum the selected inputs together if the Sum button is enabled.

Moving onto the outputs, there are three D-Subs: a Main output, for attaching your main stereo or surround monitor speakers, a Cue output, to output the three Cue stereo cue feeds along with a stereo pair of studio loudspeakers (in a live room), and an Alt Speaker output, to connect an alternative set of stereo or surround monitor speakers. The D-Control's monitoring section features three separate encoders for the output levels of the Main, Alt and Mini outputs, though only one can be active at any time. D-Command has just one encoder which is switchable between these functions.

As you would expect, the output section also includes a calibration mode, along with master Mute and Dim buttons (the D-Control offers a separate encoder for setting the Dim level), and a Mono button. As with the dynamics and EQ sections of the surface, the monitoring section offers Channel switches to mute or solo a given output channel; again, the D-Command supports up to six channels (for 5.1) and D-Control handles nine channels (for 8.1).

For cue mixes, you can adjust or mute the incoming cue inputs independently via three separate encoders and switches on the D-Control, though once again this functionality has been condensed on the D-Command, which only addresses two cue inputs. Cue inputs basically pass straight through this gain stage in the monitor, along with any talkback input, and are routed to the Cue outputs.

Finally, there's also a Utility D-Sub providing external talkback and listen-back mic inputs, AFL inputs (if you have an AFL solo buss set up in *Pro Tools*), a stereo 'mini' speaker output for connecting a pair of nearfield reference monitors, such as Auratones if you're doing TV work — some people still use them — and a talkback/slate output. Plus, in addition to the control lead that connects to the Main unit from the XMON carrying MIDI/RS422 information for communication with the Icon's monitoring section, there's also a stereo headphone output signal to drive the headphone output on the Main Unit, along with a talkback input signal from the talkback mic built into the Main Unit. The D-Control offers separate encoders and switches for the headphone, studio loudspeakers (useful if you want to slate the talkback into the live room, for example), and AFL/PFL solo options, while the D-Command condenses all of this functionality with the Cue encoder.

The monitoring section on the D-Control (pictured) and the D-Command is the only part of the console that operates independently of Pro Tools. It remotely controls the included XMON monitoring system.

button could have been illuminated with half the brightness, say, that would have been handy.

A Channel Strip's Select button has two different modes of operation, known as Select and Focus, and you can select the current mode with the Select/Focus Mode switch in the Channel Strip Master function controls section. The behaviour of this mode switch is one of the elements of the design you'll find repeated for different mode controls all over the Icon surfaces, where there's a button and various 'mode indicator' LEDs that show the current mode of a particular function. In the case of the Select/Focus Mode switch, there are two mode indicators, one either side of the button, and pressing the button alternates the mode between Select and Focus.

Select mode uses the Channel Strip Select buttons to choose channels in *Pro Tools*, and there's a useful preference on the Icon surfaces that lets you set whether this Select behaviour should Latch, where selecting a new channel doesn't automatically deselect the previous one, or be ExclOr (Exclusive Or), where only one channel can be selected at a time. Focus mode allows you to use the Select button to set a given channel to be the Focus Channel on the control surface and doesn't affect any channels currently selected in *Pro Tools*.

The Channel Strip function controls section also includes the Channel's 'Scribble Strip', a small, green, six-character LCD that displays the name of that channel. There are several display modes available to the Scribble Strip here, including the default Track Name Mode and other options allowing you to see which group a channel belongs to, the number of a given channel, the available headroom, or the peak level. The track name is shown as inverted green if the track is disabled, and the D-Control takes the Scribble Strip concept one step further with tri-colour LCDs. If a channel clips, for example, the Scribble Strip turns red.

At the top of every Channel Strip on the Fader Unit is the stereo-capable level meter on the meter bridge. The design could be improved here by including Scribble Strip displays at the bottom of each channel's meter, because when you look across the console it's sometimes hard to see at a glance which channel is associated with which meter. This is only a minor quibble, though.

Knowing The Knobs

Above the Channel Strip function controls are six (on the D-Control) or two (on the D-Command) rotary encoders. Like the faders, the encoders are touch-sensitive, which is a feature that not even all high-end, large-format digital consoles offer, and their response can be set via the Soft Key preferences. You can choose various velocity acceleration settings (how fast you turn the knob affects the resolution of the parameters you step through), Fixed mode (the normal behaviour of the encoders) and Fine mode, which allows high-resolution adjustments.

Below each encoder on the Channel Strip are an LED to show when a parameter is armed for Automation, and a six-character LCD. Each encoder also has a 15-LED indicator ring showing the current value of the parameter being controlled by the encoder. On the D-Control, both the LCD and encoder LEDs are tri-colour. Every encoder section on this part of the console has an accompanying Select button and a multi-function Bypass/Mute/Pre(-fader) button with two indicator LEDs. The D-Control also has additional Dynamics and EQ LEDs, which are illuminated if an encoder is controlling a parameter from a dynamics or EQ plug-in; these LEDs on the D-Command indicate Clip and Flip. (On the D-Control, Clip and Flip are indicated for an encoder by the encoder's LCD changing colour.)

The encoders in the Channel Strip have no fixed function, and you can set what they should control via buttons in the Channel Strip Mode controls at the bottom of each strip of encoders. The D-Control features eight Mode buttons (Dynamics, EQ, Input, Mic Pre, Inserts, Pan, Bypass-Mute and Sends), while the D-Command offers four (Inserts, Pan, Bypass-Mute and Sends) and both surfaces feature page up and down buttons which are illuminated when an additional page of controls is available in a given mode.

Pressing a Channel Strip Mode button sets the mode for the encoders on only that channel, but if you want to access the sends on all channels, for example, there's a duplicate set of Mode buttons on the Channel Strip Master section. Pressing one of these sets all Channel Strips on the surface to that Mode, which is useful when you want to see all sends at once, or all inserts, and so on.

Pan mode is the default mode, and for a stereo or mono track routed to a mono or stereo destination, the pan controls appear exactly as they do in the *Pro Tools* software, with one or two pan controls appearing from the bottom up on the encoders. If a mono or stereo track is

routed to a surround output, the panning controls are more extensive: D-Control users will find the most common Pan controls on the first page, as will D-Command users; it's just that D-Command users will have to do a good deal more paging to access all the parameters.

High-end consoles commonly have preamps that can be controlled from the console and stored as part of a recallable mixer template, and *Pro Tools* has offered this kind of functionality for some time now, when it's used in conjunction with Digidesign's own Pre, which is an eight-channel preamp for connecting mic, line and DI-level inputs. The Pre is a purely analogue box (you'll still need a 192I/O with at least eight analogue inputs) and is remotely controlled from *Pro Tools* via MIDI. Parameters for the preamp can be edited on Icon encoders where a channel takes its input from a Pre, and the D-Control has dedicated buttons for accessing this function.

From a sonic point of view, the Pre is pretty nice: it's fairly transparent and doesn't add a great deal of warmth or character to the sound. Its only real down side is that there's an audible clicking sound (sometimes referred to as zipper noise) when you make adjustments to the parameters, which means that this isn't the preamp for you if you like to ride levels when recording.

Having a workstation, mixer and microphone preamps completely integrated is incredibly convenient, and while not everyone will necessarily want to use the Pre, Digidesign do at least publish its MIDI control protocol in the Pre's manual (which is freely available on-line) so that other companies can adopt the same protocol. Reso Audiotronics, for example, have already done just that with the Reso Pre 873, an eight-channel unit 'built using a Neve 1073 analogue circuit design', which can be integrated into a Pro Tools environment.

You can edit the settings for an Insert plug-in on a channel in Inserts mode, and the console spills the controls across many pages available to the encoders for that channel.

Inserts

Inserts mode is used to add or control plug-in inserts on a channel, and each encoder in the strip represents one insert slot on the *Pro Tools* mixer; there's a handy Soft Key preference where you can reverse the order so that the inserts run from bottom to top, instead of the default top down. This is useful because reaching the top knob on a D-Control can be a bit of a stretch. With the D-Control's six encoders per channel, it's pretty easy for the five insert slots to be accommodated, and there's a Soft Key preference that lets you keep the basic pan control on the lowest encoder at all times. However, you'll need to use the Page Up and Page Down buttons to access all the slots on the D-Command's two encoders.

To add an insert to a Channel, press the appropriate encoder's Select key and, just as in *Pro Tools*, you'll be able to navigate a hierarchical menu of available insert plug-ins using the encoder and the Select button. A nice touch when scrolling through the plug-in list is that the full plug-in name is displayed across the LCD displays of adjacent channels; once you take your finger off the encoder, the name shrinks back to a shortened version on just one LCD again. When a plug-in has been added to an insert slot, its name appears in abbreviated form on the appropriate encoder's LCD display. If the name appears in inverted green, it means that insert is inactive, which happens if you don't have enough DSP power, or you've opened a Session that includes plug-ins you don't have.

To edit the parameters for a plug-in, press the appropriate encoder's Select button and all the encoders on that Channel Strip will show the plug-in's parameters. Since most plug-ins have more than six (or two) parameters, you need to use the Page Up and Page Down buttons to access all of them.

To bypass an insert, you press the Bypass-Mute-Pre button next to the appropriate encoder; if you want to bypass all the inserts on a Channel Strip in one go, press the Bypass/Mute button on the strip's Channel Strip Mode controls instead.

The D-Control has dedicated Dynamics and EQ buttons in the Channel Strip Mode controls, which set a Strip's encoders to control the first dynamics or EQ plug-in on that channel, and pressing them multiple times cycles through any additional dynamics or EQs. Although this is useful, it's worth bearing in mind that both the D-Control and D-Command feature dedicated controls for working with dynamics and EQ plug-ins, which we'll investigate later.

Send Me An Angel

Working with sends is quite similar to working with inserts. In Sends mode, each encoder represents a different send, and, again, this will be more convenient on the D-Control than on the D-Command, where you have to page through more sets of encoders to access all the sends. However, with *Pro Tools 7* adding support for 10 sends per channel, even D-Control users will have to use the Page buttons to access all 10 sends.

Adding a send is like adding an insert: in Sends mode, press the Select button in the encoder group that represents the send slot you want to use, and navigate through the menus with the encoder and Select button to specify where you want to send the signal. Once a send has been added, you can adjust the level with the encoder. One really neat touch is that the encoder ring of LEDs can act as a level meter for a send.

When an encoder is acting as a send, the channel's Bypass/Mute/Pre button is used to either mute a send or set it to Pre-fader mode, with two indicator LEDs above indicating its status, while pressing the D-Control's global Bypass/Mute button in the Channel Strip Mode controls section will mute all sends on a given channel. Additional settings for a given send, such as pan, are accessed by pressing the appropriate encoder's Select button to display the send settings across the encoders.

One feature that's especially convenient

Icon Pricing

Existing Pro Tools users can purchase a D-Control or D-Command surface to complement their Pro Tools systems. The basic price list is as follows (including VAT):

- D-Control 16-fader Main Unit £42,294
- D-Control 16-fader Expander Module £21,144
- D-Command eight-fader Main Unit £11,274
- D-Command 16-fader Module £10,569

There are also a number of upgrade possibilities from Pro Control and Control 24 configurations, as well as a selection of bundles for those looking to purchase a complete Icon system for music or post-production mixing in either stereo or surround. The bundles include a significant discount as an incentive for purchasing the whole system at once, and as well as the hardware listed, come with DigiDelivery Serv LT and a plug-in bundle. Here are some example bundles:

- 24-fader D-Command Music Bundle £46,994: D-Command eight-fader Main Unit, D-Command 16-fader module, Pro Tools HD3 Accel Core, 2x 192I/O, 192 AD, MIDI I/O, Sync I/O, Pre.
- 16-fader D-Control Music Bundle £52,869: D-Control 16-fader Main Unit, Pro Tools HD3 Accel System with PCI cards, 3x 192I/O, MIDI I/O, Sync I/O, Pre.
- 32-fader D-Control Music Bundle £70,494: D-Control 16-fader Main Unit, D-Control 16-fader Expander Module, Pro Tools HD3 Accel System with PCI cards, Sync I/O, 3x 192I/O, MIDI I/O, Pre.

INTAaudio

Pro Audio Workstations

System Price Only. Now just add Soundcard & Monitor

£799.00 ONLY INC VAT

AMD 64 Athlon X2

INTAaudio AMD 64 DC Workstation

- AMD 64 3800 X2
- Zalman 7000 AL Ultra Quiet CPU Cooler
- Zalman Fanmate Fan Controller
- Chieftec DX-01 Silver ATX Chassis
- ATI RADEON 9250 128MB Dual Head Graphics
- 1024 DDR 400 MHZ PC3200
- ASUS A8V Deluxe Motherboard
- 250GB Western Digital Program Drive
- 250GB Western Digital Audio Drive
- NEC 16x DVD-r
- 1.44 Floppy
- 530w Super Quiet Power Supply
- Cordless Keyboard and Mouse Set
- AcoustiPack - Acoustic Foam Kit

AMD 64 3800 X2	250Gb Audio Drive 250Gb Program Drive	1 Gb pc3200 DDR 400MHz MEMORY
DVD-R WRITER	Acoustic Foam Kit	ATI Radeon 128Mb Dual Head
Cordless Keyboard & Mouse	XP Home	530W Quiet Power Supply

Upgrade to 2Gb Ram for just £89.99

Monitors 15" TFT Black/Silver £119.00 17" TFT Black/Silver £149.00 19" TFT Black/Silver £199.00

Upgrade to 2Gb Ram for just £89.99

2.0GHz 533 FSB Processor	CORE DUO T2400, 1.83GHZ
DVD-R Writer	DVD-R Writer
80GB 7200RPM Hard Disk Drive	80GB 7200RPM Hard Disk Drive
15.4" WXGA Widescreen	17" WXGA Widescreen
NVidia GO6600 128MB Graphics	ATI Radeon X1600 128MB Graphics
1 Gb DDR II MEMORY	1 Gb DDR II MEMORY

£1079.00 ONLY INC VAT

£CALL.00 ONLY INC VAT

All Laptops include FREE setup, Windows XP Home Edition, plus 12 months collect & return warranty. Phone for Full spec's of Laptops. PICTURES FOR ILLUSTRATION PURPOSES ONLY

INTAaudio recommends E-MU

M-AUDIO

Audiophile 24/96 PCI	£54.99
Firewire Audiophile	£139.00
Firewire 410	£239.00
Firewire 1814	£335.00
Firewire Solo	£115.00
Fastrack USB	£65.00
Fastrack Pro	£159.00
Project Mix	£859.00

RME

96/8 PAD	£219.00
9632	£309.00
9652	£345.00
Digiface	£call
Multiface	£call

E-MU

0404	£59.00
Proteus X	£95.00
1616	£275.00
1616M	£325.00
1820	£275.00
1820m	£315.00
Emulator X	£189.00
Emulator XS	£379.00

Motu

Traveler	£699.00
2408 MK3	£749.00
24 I/O Core System	£1069
828 MK2	£540.00
896HD	£769.00
Ultra Lite	£495.00

Edirol

FA-66 Firewire	£219.00
FA-101 Firewire	£289.00
UA-25	£145.00
UA-101	£299.00

Mackie

Spike	£149.00
Big Knob	£249.00
Oynx 400F	£599.00
Oynx 800R	£999.00
Oynx 1620	£649.00
Oynx 1640	£999.00

Pentium D Workstation

£799.00 ONLY INC VAT

Upgrade to 2Gb Ram for just £89.99

- Antec P150 ATX Silent Case
- Intel Pentium D 805 2.66Ghz Dual Core CPU - 2x 1Mb Cache
- Scythe Katana Heat Pipe Quiet CPU Cooler
- 250Gb SATA Program Drive 8Mb Cache - Western Digital
- 250Gb SATA Audio Drive 8Mb Cache - Western Digital
- 1GB DDR 400 Mhz - Memory
- ATI RADEON 9250 - 128Mb Dual Head
- ASUS P5P800 - SE - Intel 865 Chipset
- 430W NEO Power Quiet PSU
- NEC DVD-R Writer - Dual Layer
- Windows XP Home
- PS2 Keyboard
- PS2 Optical Mouse
- 120mm Ultra Quiet Case Fan
- 2 Port NEC Firewire Card
- 6x USB 2 Ports

2.66Ghz Dual Core CPU

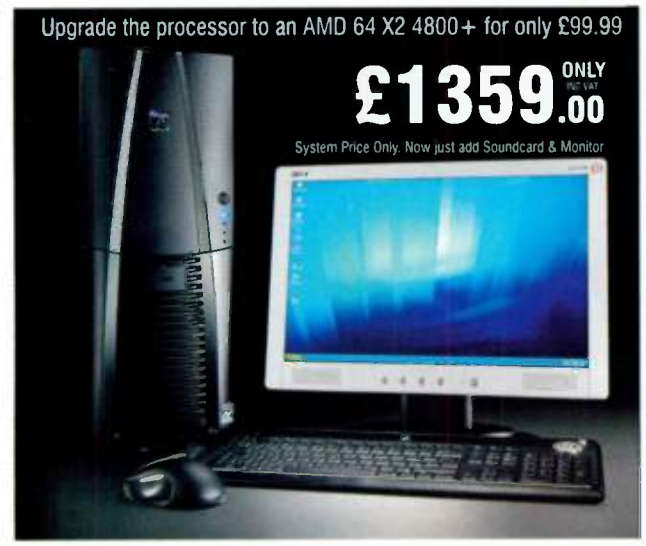
System Price Only. Now just add Soundcard & Monitor

Pentium D 805 2.66Ghz DC	250Gb Audio Drive 250Gb Program Drive	1 Gb pc3200 DDR 400MHz MEMORY
DVD-R WRITER	Antec Acoustic Foam Kit	ATI Radeon 128Mb Dual Head
ANTEC P150 Silent Case	XP Home	430W Quiet Power Supply

Upgrade the processor to an AMD 64 X2 4800+ for only £99.99

£1359.00 ONLY INC VAT

System Price Only. Now just add Soundcard & Monitor



The INTAaudio AMD 64 X2 Dual Core Workstation

AMD 64 X2 4400+	250Gb Program Drive	2 x 74Gb Audio Drives 10,000 RPM, on RAID 0
DVD-R WRITER	Acoustic Foam Kit	ATI Radeon 256Mb Dual Head
2 Gb pc3200 DDR 400MHz MEMORY	XP Home	500W Quiet Power Supply

Monitors 15" TFT Black/Silver £119.00 17" TFT Black/Silver £149.00 19" TFT Black/Silver £199.00

- AMD 64 X2 4400+ Processor
- Arctic Cooling Freezer 64 Pro
- ANTEC Tri Cool 120mm Chassis Fan
- ANTEC TX1050B ATX Chassis
- ATI RADEON 9250 256MB Dual Head Graphics
- 2048Mb DDR 400 MHZ PC3200
- ASUS A8V Deluxe Motherboard
- 250GB Western Digital Program Drive
- 2 x 74Gb 10,000 Rpm SATA Western Digital Audio Drive
- NEC 16x Dual Layer DVD-R
- 1.44 Floppy
- 500w Super Quiet Antec Power Supply
- Cordless Keyboard and Mouse Set
- AcoustiPack - Acoustic Foam Kit

INTAaudio Pro Audio Workstations

www.inta-audio.com Tel: 0870 199 9397

Unit 2a Bodmin Road, Walsgrave, Coventry CV2 2DB Open Mon-Fri 9am-5.30pm

Switch VISA MasterCard

In Sends mode it's possible for an encoder to show metering for the level of the send and the actual level the send is set to. On D-Control's tri-colour LED encoder ring, you can see the metering in green and the value of the parameter in red. The illuminated Pre mode indicator LED beneath the encoder's Scribble Strip shows that the send is pre-fader.

for working with sends is Flip mode. The Channel Strip Master section has a Flip button for each row of encoders. Simply press Flip and the corresponding row of encoders will be swapped with the faders, the Flip button will light up, and on the D-Control, the encoders' LCD Scribble Strips will become orange to indicate that they have been flipped. On the D-Command, each encoder's Flip light illuminates instead. This facility has numerous uses: in Sends mode, for example, you could Flip the first row of encoders, which might be controlling the first Send slot for a headphone mix, to the faders, causing the channel volume to appear on the first row of encoders instead.

The D-Control also features a variation on Flip mode, known as Flop mode, which transfers any other row of encoders to the bottom row. It's a nice touch to put encoders' controls in a more convenient position, but I preferred to use the 'bottom to top' arrangement for encoders most of the time.

Custom Faders

By default, the Icon surface mirrors the order of faders in *Pro Tools*' Mix window, which is generally fine if you're working with smaller Sessions that have no more channels than you have faders on the surface. The usual banking system is implemented for accessing additional channels, but to make mixing more manageable, there are also several different Custom Fader modes available from the Custom Fader controls on the Main Unit of the Icon surface. One of the most useful is a feature you'll find on most higher-end digital consoles — it's called a Custom Group in Icon terminology — where you can assign a given channel in *Pro Tools* to a specific fader on the surface. For example, if there's a drum kit in the Session, you could create a Custom Group that contains only the individual channels for the drum kit. Now, when you want to make an adjustment to a drum kit channel, instead of banking up and down the console to find the drum kit channels, you just recall the drum kit Custom Group.

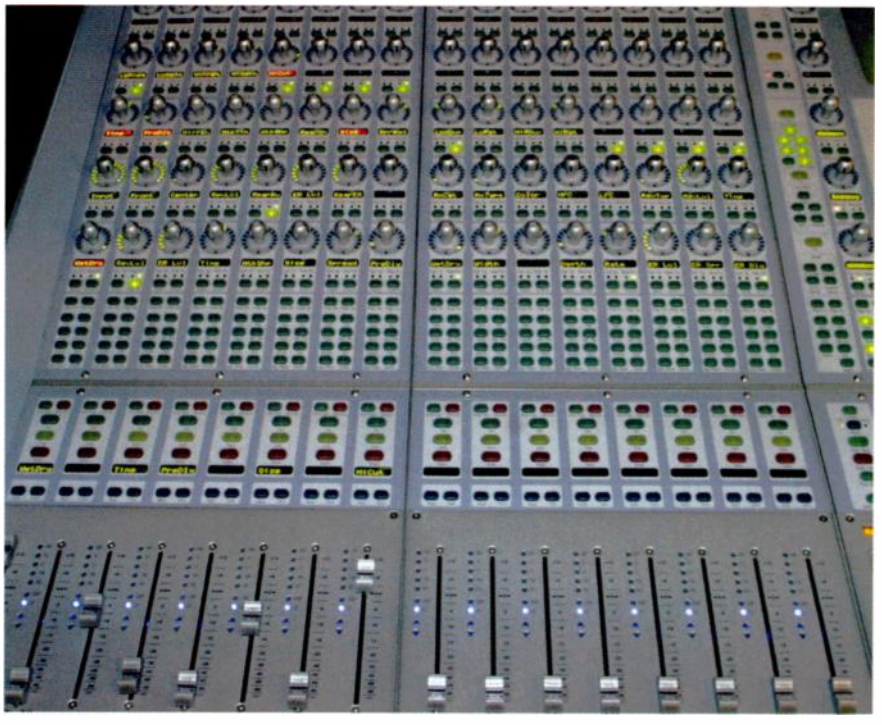
Creating a Custom Group is easy, and when a channel strip is in Custom Fader mode, a blue 'CF' indicator lights up next to the fader. The reason each fader has its own CF indicator, as opposed to there being one global CF indicator for the whole surface, is that it's possible to partition the console so that only a few channel strips become Custom Faders when CF mode is activated, and the rest of the channel strips behave normally. In this case, banking up and down channels causes only those channels not in CF mode to scroll.

Unfortunately, it's not possible to choose any number of faders to be available in CF mode, but there are plenty of options. On the D-Command you can assign four, eight, 16 or 24 faders (the whole surface) to be Custom Faders, while on D-Control you can set the number of Custom Faders in banks of eight. On either surface, the Custom Faders can be justified to the left or right of the surface. While it would be nice to choose an arbitrary number of Custom Faders, this isn't a big limitation. A further addition to CF on the D-Control is that it's possible to have two banks of CFs on the surface simultaneously, which can come in particularly handy if you have the Main Unit sandwiched between several Fader Modules.

The D-Control can store up to 48 Custom Groups, accessed via four pages of 12 Bank Select buttons, while the D-Command offers eight. Custom Groups can be named and edited once created, and channels in the Custom Group are ordered in the same sequence they're added to the Group. This means that if you want to re-order the sequence you have to remove channels and add them again. A nice touch is that it's possible to add the same channel twice, should you want to see different settings for the same channel, such as inserts and sends, simultaneously on adjacent strips.

One slight drawback of the D-Control's Bank Select area is that, unlike the Soft Keys, the buttons for triggering the different Custom Groups don't have Scribble Strip displays. This inevitably means that you end up with little bits of paper on the console to help you remember what a given button represents, which is a bit annoying, especially since Custom Groups are stored within the *Pro Tools* Session and can therefore be different in every Session. On the plus side, though, it is possible to access the Custom Groups via the Soft Keys, so you can see the Group names.

Because the Custom Group behaviour is handled by the *Pro Tools* software and stored in a *Pro Tools* Session file, a welcome feature in future versions of *Pro Tools* would be a Custom Group editor window for Icon users. Although it's easy to set up Custom Groups using the surface, being able to manage and rearrange them by dragging and dropping on-screen objects would be useful. Hopefully, this would also make it possible to import and export Custom Groups between different Sessions that have similar track layouts; as it



Here you can see the parameters of the reverb plug-in *Revibe* displayed via the rotary encoders of the D-Control in Custom Fader mode. The yellow-on-red display of certain Scribble Strips indicates that that parameter is also assigned to the faders, and note the blue Custom Fader LED next to the faders, showing that the channel is in Custom Fader mode. This makes plug-in editing easy, especially with plug-ins that have a large number of parameters.

The Portico™ Series

A Range Of All-analogue Building Blocks

designed by

Mr. Rupert Neve

- *Available in both Vertical & Horizontal Configurations*
- *Custom Rupert Neve Designed Transformers on Inputs and Output provide Sweet Musical Response of Mr. Neve's Classic Modules*
- *Highest Grade Components, Carefully Designed and Assembled in Texas by Rupert Neve Designs*

Rupert Neve Designs

The Signature of Excellence

For more information visit our web site

www.RupertNeve.com

▶ is, you have to set up the Groups every time in every new Session.

More Custom Fader Modes

Although Custom Groups are independent of the long-established Mix and Edit Groups in *Pro Tools*, it's possible to use the Custom Fader mode on the surface to access these types of Group as well. Other Custom Faders modes include Tracks mode, which enables you to assign only tracks of a certain type within the Session to the Custom Faders, where Bank Switches 1-5 bring up all Master, Audio, Auxiliary, MIDI and Hidden tracks from the Session to your CF bank. This is particularly useful for keeping a small CF bank of just Master Tracks, since neither Icon surface offers dedicated Master faders — and with this feature, there's almost no need.

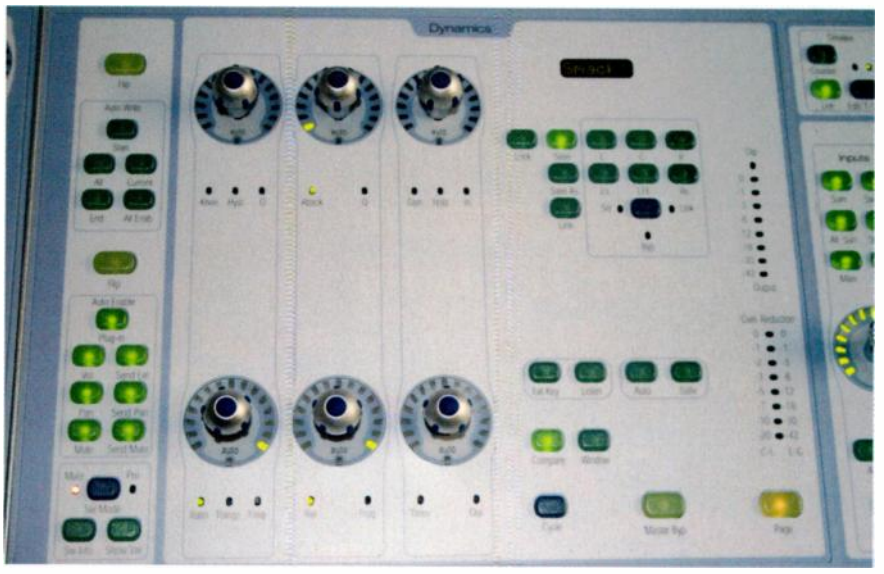
Another really nice Custom Fader mode enables you to fill all the encoders on the CF channels with parameters from a single plug-in. This works better on the D-Control where you have more encoders, of course. Once the plug-in parameters are laid out on the encoders, you can make use of their touch-sensitivity to bring certain parameters down to the faders in the CF channels, which is useful for gaining easy access to basic reverb parameters such as room size, pre-delay and so on. In Automation Enable mode, you can also simply tap the controls to toggle automation arming for these parameters. Very neat!

The best thing about plug-in mapping in Custom Fader mode, though, is that, unlike Custom Groups, the maps can be imported and exported between different Sessions, and *Pro Tools* remembers the plug-in maps for the last Session that was opened when you start a new one.

Dedicated To Dynamics

Having a dedicated set of controls on a mixing console for adjusting the built-in dynamics and EQ parameters on a given channel is pretty common, and the Main Unit in both Icon surfaces features such dedicated controls. As *Pro Tools* users will know, the application's mix engine doesn't actually offer built-in dynamics and EQ for audio channels; if you want to use either dynamics or EQ on a channel, you simply insert a suitable plug-in. So the really neat thing about the Icon's dynamics and EQ controls is that compatible plug-ins automatically map their parameters to the available controls on the surface. For instance, you don't have to think about how to adjust the threshold of the compressor, depending on what compressor plug-in you're using: the threshold parameter of a compatible plug-in always maps to the Threshold control on the surface. I say compatible, because obviously a plug-in needs to present its parameters to *Pro Tools* in a specific way for the full Icon functionality to work. Fortunately, most plug-ins are now Icon-aware.

In these three pictures you can see how the parameters of a Pro Tools dynamics plug-in (in this case, *Smack!*) are mapped to the dedicated hardware controls for adjusting dynamics on both the D-Control and D-Command surfaces. Notice how the rotary encoders on D-Command have more mode indicator LEDs underneath that light up to show you what parameter a given encoder is controlling.



When you select a channel as the Focus Channel on the surface, the first EQ and dynamics plug-ins used on that channel are automatically mapped to the Main Unit's EQ and dynamics sections, and you can toggle to the next suitable plug-in on the channel by pressing the relevant section's Cycle button. If you want to keep a given instance of a plug-in mapped to the EQ or dynamics sections when you select a different channel as the Focus Channel, such as for an EQ or limiter plug-in on the master channel, both sections feature a Lock button that keeps the current plug-in 'locked' to those controls when you change the Focus Channel.

Both the EQ and dynamics sections offer Channel Select switches for use with multi-mono plug-ins, which comes in handy when working in surround. The D-Control features nine Channel Select switches (L, Lc, C, Rc, R, Ls, Cs, Rs and LFE), allowing you to control up to eight (for 7.1) channels, whereas the D-Command offers six Channel Select switches (L, R, C, Ls, Rs and LFE) for six-channel (for 5.1) mixes. The Channel Select switches offer various different modes, such as Link mode, where you can select which channels are linked, so that altering a parameter on one channel alters the same parameter on all other linked channels. This

D-Control Surround Option

When the D-Control first shipped, the lower right-hand panel under the monitoring section on the Main Unit was empty. However, this space can now be filled with the optional Surround Panner, featuring two separate banks of panning controls and a touchscreen. Each bank offers a non-motorised, touch-sensitive joystick for panning in a 360-degree sound space, two Punch switches for punching in and out of automation writing with the joystick, and an encoder (as found in other areas of the surface) for controlling parameters such as position and divergence, as set by the Panner Control switches underneath.

The touchscreen is split up into two main areas that represent the current state of the surround panners controlled by the joystick. In a way, this compensates for the lack of motorisation in the joysticks, since the touchscreen shows the movements, and if the joystick's position gets out of sync with the channel's actual pan position, the screen shows two separate cursors. Digidesign have implemented what is termed a 'proximity takeover' effect, so you can move the joystick to roughly the position of the current pan position before the joystick takes control again, eliminating 'jumps' in pan position. You can, however, do the reverse and have the current pan position snap to the location of the joystick; and, since it's a touchscreen, you can move the pan cursor on the screen with your finger instead of the joystick.

As with the Dynamics and EQ sections of the D-Control, the Surround Panner shows panning for the Focus Channel and features a Lock button to lock the focus of the currently selected Channel (or Channels), if you want to change the Focus Channel for other parts of the surface. A handy Link mode enables you to set whether the panners control the two mono sides of a stereo track routed to a surround output, or two separate mono tracks.

The added bonus of the Surround Panner is that the joysticks can also be assigned to control other parameters instead of just panning. For example, you could assign the 'X' axis of one joystick to an EQ plug-in's frequency parameter and the 'Y' axis to the gain control of a single band.

can be useful if you want to treat the left and right channels as a pair, or maybe omit the LFE channel so that it can be treated independently.

Pressing a Channel Select switch in Select mode sets which channel's parameters should be edited on the surface, and a nice touch is that if the surface's 'Channel Window Display' preference is set, the plug-in's editor window in *Pro Tools* also displays the selected channel's parameters. You can also toggle the appearance of the EQ or dynamics plug-in editor window by pressing the Window button in the EQ or dynamics section. Finally, Bypass mode enables you to mute specific channels, and a Master Bypass switch is also provided in both EQ and dynamics sections, to bypass the entire plug-in.

In terms of the available controls, there are 15 parameters for dynamics, controlled by 12 encoders on the D-Control and six on the D-Command, with a Page button to access the ones that won't fit on these. D-Command users will be pressing the Page button a little more often to access all the functions, but Digidesign have been pretty smart in putting the six most common controls for dynamics processors on the first 'page': Knee, Ratio, Attack, Release, Gain and Threshold. The D-Control's dynamics section offers eight-LED meters for input and output levels, Compressor/Limiter Gain Reduction and Expander/Gate Gain Reduction, plus an additional Clip indicator LED. This is great for visual metering feedback from the plug-in without having to open the plug-in's editor window on screen. On the D-Command, the Compressor/Limiter and Expander/Gate Gain Reduction displays share one eight-LED section, and instead of both input and output-level metering, there's an eight-LED input level display with a Clip indicator for the output signal.

EQ Control

The D-Control's EQ section features 19 controls for five bands of parametric EQ plus high- and low-pass filters, and an additional two controls for setting the input and output level of the EQ plug-in. This perfectly fits plug-ins like Digidesign's own *EQ III* or Sony's *Oxford EQ*, for example. The filter strips each include two controls for setting Q and frequency, plus a Notch/Shape selector switch, while the low- and high-band strips add a third control for gain. The low-mid, mid- and high-mid-band strips are almost the same but don't have the Notch/Shape selector switch, and all strips offer an 'In' button, to toggle whether that band is active or not in the EQ plug-in.

The D-Command offers the same functionality as the D-Control, but condensed down to 12 controls organised into four strips, each of which is shared — for instance, the low EQ band and high-pass filter share a strip. The EQ section also offers metering: two eight-segment LED meters for input and output levels, plus a clip indicator, on the D-Control, and a single eight-segment LED

meter for the output level on the D-Command.

Overall, the implementation of the EQ and dynamics sections is simple and brilliant, and it illustrates one of Digidesign's advantages, in that there are few companies who could coordinate such tight integration of third-party plug-ins with a control surface in this way. There are plenty of mixing consoles with dynamics and EQ sections, and there are plenty of workstation-based mixers with dynamics and EQ plug-ins, but no-one has bridged these two worlds quite as successfully as Digidesign.

In Use

The curious thing about products such as the D-Control and D-Command is that they don't actually add any new features to your system or improve the sound quality. However, what they do add is usability, and an Icon surface improves the Pro Tools user interface and experience tremendously, especially if you prefer working with physical mixing surfaces instead of the mouse. In terms of usability and integration, a small feature that impressed me when I first saw an Icon surface was the way in which touching controls that are automated can automatically switch the automation view for the relevant track in *Pro Tools*; what's great about Icon is the way in which you start to view *Pro Tools* as an extension of the surface, rather than the other way around, as with most other systems.

Here you can see an interface for Sony's *Oxford EQ* plug-in in *Pro Tools*, alongside photos showing how the same plug-in is mapped to the D-Control and D-Command's respective EQ sections. As you can see, *Oxford EQ* is particularly well suited to the D-Control, having all five bands of parametric EQ plus filters. Don't try this EQ curve at home, though...

However, the integrated workflow isn't quite seamless just yet. For example, there are many operations in *Pro Tools* that are blocked when the transport is running, such as adding a send to a channel or changing I/O routing. There are, of course, a mixture of technical and legacy reasons why this is the case; but if you're used to working with a stand-alone mixer alongside *Pro Tools*, you're probably not used to having to stop playback to add a send to a channel. I'm sure Digidesign will work towards improving this area of *Pro Tools* for D-Control and D-Command users in the future, as that is Icon's *raison d'être*.

When you compare the D-Control to large-format competitors, it would also be nice to have more flexibility on the input and outing routing. For example, *Pro Tools* has the ability to send a single channel to multiple outputs, but there's no way to access this from the surface — it would be neat if you could use the encoders to mult outputs from a channel. On the input front, meanwhile, something that *Pro Tools* can't do right now is to assign two inputs to a channel and A/B between them.

However, the Icon surfaces do have some neat tricks, such as D-Control's Inline Console mode, where if you have an auxiliary track routed as the input to an audio track, you can control the level of the auxiliary track input from the bottom encoder of the audio track by pressing the audio track's Input button in the Channel Strip Mode controls. This is especially useful where you have an audio input from a controllable preamp like the Pre as the input for the auxiliary track, as Input mode for the audio track receiving this signal displays both the level for the auxiliary track on the bottom encoder, and the preamp settings on the subsequent encoders.

On the aesthetic aspect of the D-Control, one slightly pedantic comment I have is that, compared to other consoles, it's a shame the arm rest at the bottom part of the surface isn't bigger. It's quite common to lean over consoles with your arms (especially at those times when you're burying your head in your hands!) and D-Control isn't particularly welcoming in this area — not to mention for those people who like to kick back and put their feet up on the console! (It does happen...)

In terms of the actual surface design, the layout of controls on the D-Control is quite spacious, partly because the rotary encoders each take up quite a bit of room. I quite liked

the spacious nature of the surface, although I know a few people who think the controls might be a little too spread out, and while the D-Control isn't physically much different in size to similar surfaces, I think this gives it the feeling of being slightly bigger. The D-Command has almost the opposite quality; although it doesn't feel overcrowded, it does feel different to the larger Icon surface.

While the two surfaces follow a similar design aesthetic, there are a few quirks when it comes to consistency. For example, some of the controls are in geometrically different parts of the surface, and where the Utility button is yellow on the D-Control, it's green on the D-Command. This is perhaps a little pedantic, but when I first started using the D-Command after the D-Control, I was looking for the Utility button and I'm sure I would have found it quicker if the colouring was consistent.

On the subject of button colouring, the one feeling I did have about the colour scheme of the Icon surfaces (one that I know others share) is that there are just too many green buttons! As Kermit the Frog so eloquently said, 'it's not easy being green', and, at times, it really does become hard to distinguish different areas of the console with a casual glance. You do get used to it, but I can't help thinking some contrast in the design would have helped here.

Iconclusion

What I find most interesting about the Icon system is that Digidesign have, in effect, built a mixing console with an integrated digital audio workstation in the reverse order to that which most manufacturers have chosen. The company started off with a computer-based recording system that had some basic mixing capability, later adding effects and more comprehensive mixing functionality with TDM plug-ins and more DSP power in Pro Tools III, before introducing control surfaces like Pro Control and Control 24 to tie the whole system together in later versions of *Pro Tools*. In the case of Icon, the mixer is the digital audio workstation.

Traditional mixer manufacturers, however, have taken pretty much the opposite approach, starting off with mixing engines and control surfaces, and only later adding the ability to control digital audio workstations. Notable examples include Yamaha in the project-studio world, SSL at the high end, and now Euphonix, also in the high-end market, with their System 5 MC, which is probably the Icon's closest competitor.

The problem these other manufacturers face when competing with the Icon, though, is that *Pro Tools* is the most widely used digital audio workstation, and Digidesign have locked everybody else out from creating feature-rich competitors. Non-Icon surfaces from other manufacturers communicate with *Pro Tools* via Mackie's HUI protocol, which, while adequate for delivering basic operation, provides nothing like the functionality and integration that Digidesign's own control surfaces offer. Digidesign, of course, would be foolish to invite competition when they have a perfectly good (and growing) console business; but all this does mean that if you want a control surface primarily for controlling *Pro Tools*, you really would be best off investigating either the D-Control or the D-Command.

The big question for manufacturers and users alike is whether studios are ready for the level of integration offered by products like the Icon, and, ultimately, the dependency on one system. You can still integrate outboard gear with the Icon, like any other mixing system, of course, but there's a big difference between having a Pro Tools system attached to a dedicated mixing console, and a system where Pro Tools *is* the mixing console. Digidesign would argue this difference is the company's advantage, and conceptually I'd have to agree that integrated systems are the obvious progression for studio hardware.

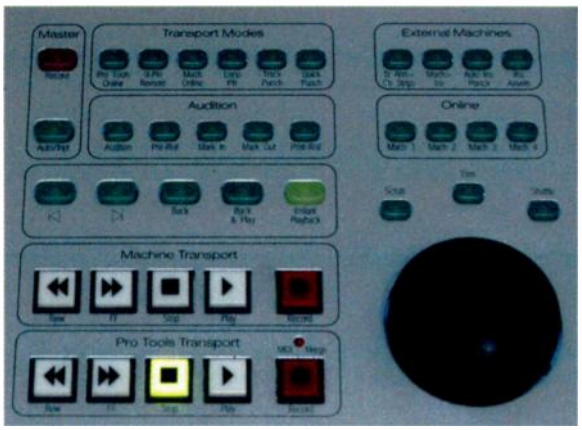
For as long as *Pro Tools* remains the most widely used digital audio production tool, Digidesign are uniquely positioned to offer perhaps the best level of integration of any system. Arguably, no other company has such tight control of hardware, applications and plug-ins, and the Icon really shines where these strengths are fully utilised. There are still a few rough edges, but overall, the Icon is a stable platform that is constantly being improved. The D-Control and D-Command are great controllers for *Pro Tools*, and more importantly, for a large number of customers, especially those in music, the Icon stands up as a system in its own right, comparable to and instead of a large-format console (in the case of the D-Control). Personally, I can't wait to see what the future holds for integrated consoles. SOS

Getting Started With The Icon

Assuming you already have a Pro Tools HD system up and running, completing an Icon system by adding a surface and XMON is fairly straightforward. Each element of the control surface (the Main Unit and Fader Modules) connects to the host computer (the one running *Pro Tools*) via 100-base Ethernet, so to get started you can simply connect your computer, Main Unit and any additional Fader Modules to a network switch. If you need to connect your computer to a LAN or need Internet connectivity via the Ethernet port, it doesn't seem to be a problem to connect to a larger network with the same switch, although I noticed I often needed to unplug from a network when performing the firmware updates that are frequently supplied with new versions of the *Pro Tools* software.

The minimum requirements recommended for a computer to be used with an Icon surface are either a dual-processor Power Mac G5 or dual-processor Windows XP machine with 2GB RAM. There are no recommendations for clock speed, but from my experience I would say at least a dual-2GHz Power Mac or a dual-2GHz Xeon/Opteron system. The reason for these requirements is that the majority of processing required by an Icon surface is taken care of by the host computer, so if you run a large number of HTDM or RTAS plug-ins with an Icon surface, you're going to need a fair amount of power. On really demanding projects, for example, occasionally I noticed the level meters becoming a little sluggish.

The audio output from your Pro Tools rig needs to be connected to the main audio input on the XMON; if you're using a 192I/O with an analogue output card, this is a simple matter of using one D-Sub-to-D-Sub lead from the 192I/O to the XMON. Next, connect a pair of speakers to the outputs of the XMON for monitoring using an appropriate D-Sub breakout cable (not included), and attach the control lead from the XMON to the D-Control or D-Command Main Unit.



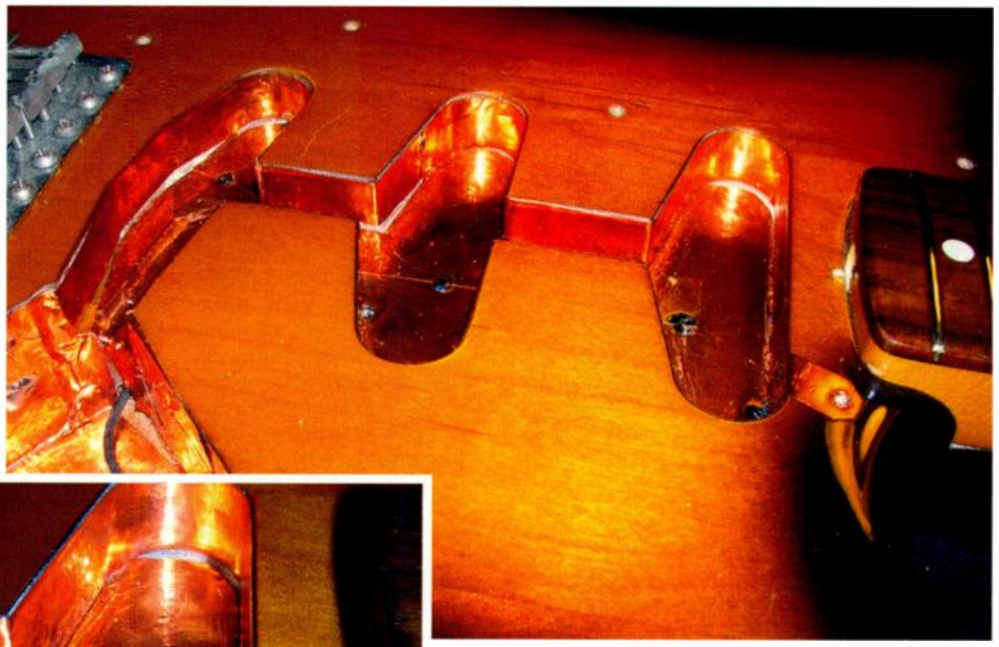
The D-Control's Transport section provides familiar transport controls for using both *Pro Tools* and an external machine, although the functionality for the latter isn't implemented in the current version of *Pro Tools* at the time of writing. The Master Record and Auto Input button in the top left of the picture enable all suitable channels in *Pro Tools* for either record or Input.

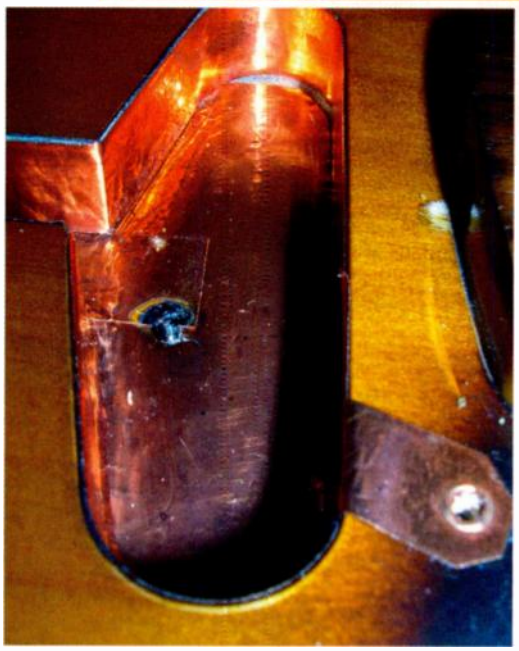
information

- £ See 'Icon Pricing' box.
- T Digidesign UK +44 (0)1753 655999.
- F +44 (0)1753 658501.
- E infouk@digidesign.com
- W www.digidesign.com

TECHNIQUE Minimising noise in electric guitar rigs

Electrical noise — hiss, buzz and hum — is something that plagues every electric guitarist to some degree, but noise comes in a variety of forms and it is important to establish exactly which kind(s) you are experiencing in order to devise an appropriate solution. Most noise in an electric guitar rig emanates from one or more of five different sources: amplifier self-generated hum and/or hiss; hum or buzz picked up by the guitar itself; self-generated noise from any pedals/processors in the circuit; gain structure-related noise, such as cascaded distortion stages; and ground-loop-related hum. If you think you are suffering from noise that isn't generated in one of these ways, I'd like to hear about it!





Above: A properly screened Strat-type guitar. To make a significant difference, the whole surface area of the pickup and wiring cavities must be covered and connected to the earth side of the circuit. There must be electrical continuity throughout, and in this example, every join in the copper foil is bridged with solder.

Left: A small tag of copper foil is used to continue the screening onto the underside of the shielded scratchplate. Passing one of the scratchplate screws through it ensures consistent contact.

The most efficient way to track down noise in a guitar system is to think of the amplifier or studio monitor system as the end of your signal chain and work systematically back from there. If you don't do this, you have no idea whether the noise you are hearing from the amplifier is being generated within the amp itself, or being picked up by the guitar and fed to the amp. You could end up taking steps to solve a problem you don't have, as well as completely failing to solve the one you actually do have. If your amp or monitoring system hums or buzzes excessively with no input connected to it, then you've got an equipment malfunction. That is beyond the scope of this article, so for these purposes I'll assume that that part of your rig is clean. From here on, I'm also going to treat amps and recording processors (Line 6 Pods and the like) as the same, because it is the 'upstream' noise of the guitar itself and related systems that we are interested in.

When tracking down noise it always pays to initially reduce your system to the minimum number of components, so begin by connecting your guitar directly to a single amp or recording processor via a screened cable, set the volume of the amp or monitor system to a normal operating level, turn the guitar's volume control all the way down and just listen. If there is any more noise than there was before the guitar was connected then the cable is at fault. With nothing connected, the amp's input jack will be automatically short-circuited to ground; with the guitar connected, but turned down, the input is again shorted, but at the other end of the cable, so the cable is the only variable.

Assuming all is well with the cable, now turn up the guitar's volume to maximum, hold the strings in a normal playing fashion and listen again. If you hear no more noise than before, congratulations; you must have a fantastically well-screened guitar and the perfect guitar-recording environment. The rest of us will be hearing at least a bit of buzzing and maybe a bit of 50/60Hz hum as well. Move the guitar around over an area of a few feet either way to see if the

Guitar Grounding Practicalities

If touching your guitar's metal jack socket when your guitar is connected to the amp kills the noise, but touching only the strings doesn't, then you may have a faulty grounding connection inside the guitar. Invariably there's a wire connected to the back of the pots or the cold side of the output jack that connects to the bridge directly or via the tremolo spring anchors, so check that this is intact. If you're not sure, make a temporary connection with another piece of wire to confirm your suspicions. Pretty much anything will do. If the hum goes when you fit your piece of wire, then the existing ground wire is broken or detached somewhere.

If you hear electrical noise (including crackling) when you move the tremolo, that's probably because the tremolo springs or some other non-moving part of the tremolo is grounded, but the strings are connected to it only via the tremolo pivot points, and if these don't have a very low electrical resistance at all times, you'll get noise that will vary as the resistance varies. Spraying a contact enhancer such as DeOxit on the pivot points can help, but in some tremolo designs it may be best to use a thin, very flexible wire to ground the moving part of the tremolo providing you can find or create a suitable attachment point, such as a tag washer fixed under one of the existing screws.

hum goes away. The level of hum is usually directly related to the guitar's proximity to any large mains transformers in the room. If you are using conventional (non-hum-cancelling) single-coil pickups and you are within the radiated field of a mains transformer, you will get hum. Exactly how much depends on the gain in your system and your proximity to the source. If you can't work out the origin of the hum field, try switching off everything except your amp (or monitor system, if you are DI'd) and then switch things back on one at a time to see when the hum reappears. When it does, see if you can re-site the offending item further away. The only solution is physical separation, as the amount of additional screening required to keep induced hum out of the pickups would actually prevent the guitar working at all. Of course, if you are using humbucking pickups, you are in the clear on this one, but the chances are you'll still have some 'buzz'.

Buzz has a lot more high-frequency content than hum. If you are unsure which you have, try turning your guitar's tone control all the way down; if the noise mostly goes away, you are dealing with buzz rather than hum. Buzz will also often be greatly reduced when you touch the strings or any other metal part of the guitar, sometimes accompanied by an audible click, whereas hum will remain unchanged. The common explanation for why noise goes away when you touch the strings or metalwork is that you are adding to the overall amount of screening. I'm not so sure about that, because certain types of noise actually get louder when you hold a guitar close to your body without touching the strings. This suggests to me that the player's body is, effectively, conducting the interference into close proximity with the guitar. The noise goes away when you touch the strings because that interference is safely conducted away to ground.

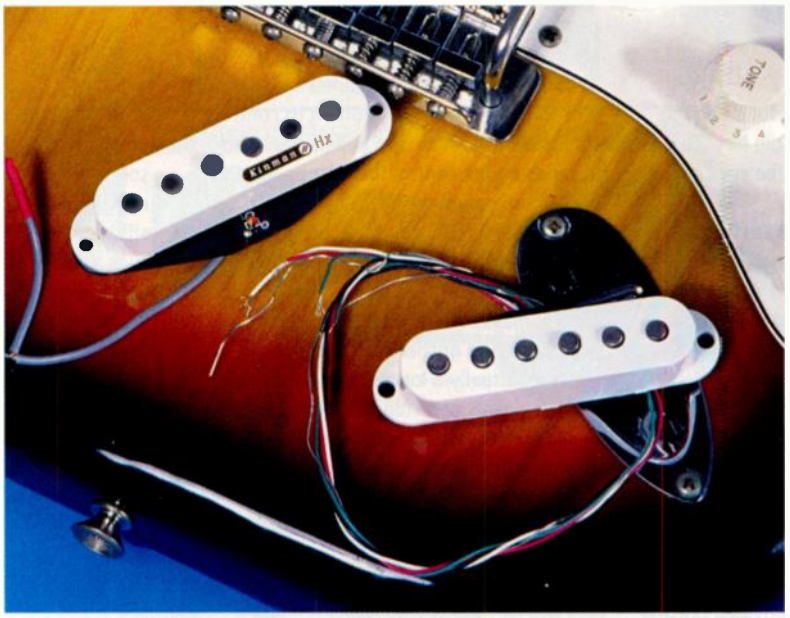
Unlike hum, which is generally induced directly into the pickup coils themselves, buzz gets in everywhere, so any bit of unshielded wiring can be the source. Even guitars with humbucking pickups will often still buzz. This is, understandably, very frustrating if you've just shelled out for a set 'noiseless' pickups for your Strat and find out the instrument is just as noisy as before; it no longer hums, but the amount of buzz is unchanged because the noise is getting in via the control cavity and the unscreened wiring rather than the pickups. The only answer is to screen every part of the internal electronics with copper foil or conductive paint, which is then connected to the earth side of the circuit. Do not attempt to screen the pickups themselves, or even the pickup covers, however, as this will alter the sound.

Screening will make a major improvement, but if you are using single-coils with a high-gain setup or lots of compression, you will still have some noise pickup. Buzz is often sensitive to the angle at which you hold the guitar, however, so you can always try to find the 'null point' at which the noise is least intrusive and simply do your best to keep the guitar at that angle whilst recording. It sounds crude, I know, but pro Strat and Tele players have worked that way in the studio for years because, until recently, there were no hum-cancelling single-coil pickups that sounded enough like the real thing to make the trade-off worthwhile.

The most common sources of buzz are TVs and CRT computer monitors, computers themselves and lighting dimmer switches. Just occasionally you'll also find a poorly designed external power supply for some piece of equipment in your rig that puts buzzy noise back onto the mains and thereby affects everything in the room. So, switch the TV off, use a flat-screen (non-CRT) monitor if possible, site your computer over four feet away, dump any noisy PSUs and use only conventional incandescent lighting.

If you've done all of those things and you've still got a nasty, edgy-sounding buzz, then the chances are that there is a lighting dimmer involved somewhere. The trouble is, it doesn't have to be *your* dimmer — lighting dimmers can affect you from an adjacent room, or a room above or below you. And it doesn't even have to be the dimmer itself — the cable running between the dimmed lamp and the dimmer control can also emit interference and this is often routed across the middle of the room within the ceiling void. Dimmers make most noise when they are actually dimming, so the noise will improve slightly when you turn the dimmer all the way up, but only switching it off altogether will make the interference go away. Dimmers that work on an entirely different principle and do not create electrical interference are just starting to appear on the market — watch this space for news.

Over the next few issues we'll tackle sources of noise beyond the guitar itself: pedals, cascading gain stages and earth loops. *Dave Lockwood*



'Noiseless' single-coil pickups, such as these models from Kinman and Dimarzio, will cancel hum effectively, but noise can still get into the system via the wiring and controls.

Strat Pickup Combinations

You can combat both hum and buzz in Strat-type guitars by using a reverse-wound, reverse polarity middle pickup (as the name suggests, the magnets are the other way up and the coil is wound the opposite way) and most modern (non-vintage reissue) Strats are now wired this way as standard. When used in combination with either of the other pickups, the RW/RP pickup creates a parallel-connected humbucker. Noise, which is induced into the coils only, is cancelled out as the two coils are, effectively, out of phase, whilst the strings are sensed by the magnets, initially out of phase (due to the reverse polarity of one of the pickups) and then restored to in-phase by the reverse winding. The net result is a clean signal with no noise. It works a treat, but only in switch positions two and four, which, ironically, are the ones that you would rarely choose for high-gain work.

One of Paul's more obvious tweaks to Dave's production sound was his replacement of the original bass patch with a layered sound combining a main Access Virus bass and an sub-bass sound from *Logic*'s *ES M* synth.

Mix Rescue

This month, Dave Rogers' dance track gets a dose of production 'fairy dust'.

Paul White

Regular readers may recognise Dave Rogers, as we paid him a Studio SOS visit back in *SOS* December 2005. A couple of weeks ago he gave me a call and asked if he could bring a mix up for me to listen to. Dave's background is primarily in DJ'ing rather than live musical performance, though he's made progress with the keyboard since he started composing. The mix he played me was an 11-minute dance epic with a nicely chilled feel constructed with a DJ's ear for club play, but Dave felt the mix could use some tidying up and he was also keen to see what kind of production tricks might be used to add interest.

On first playing, the intro, which was based around a rhythm loop, went on rather too long for my taste with nothing really happening, and then when the beat did kick in with the sampled vocal phrases over the top, it soon reached another plateau that left me wanting to hear something new. Dave had treated the vocal samples with echo, which suited the style of song very well and he had also underpinned the arrangement with a well-chosen club-style kick drum played back from *Logic*'s *EXS24* sampler. All his mix really needed was some fine-tuning and a bit of ear candy.

Adding Ear Candy

Rather than try to rework the entire track, I edited it down to around 2.5 minutes and then decided to try to bring something new to each of the sections — a slightly different tactic to the usual Mix Rescue approach of treating what is already there. For the intro section I kept Dave's original drum loop, but halved the length of the intro to stop it becoming monotonous — if a longer version were needed to satisfy the dynamic of a club audience, Dave could always remix my remix. In addition to the drum loop, the elements Dave had available for the intro were limited

Rescued This Month...

In 1996 Dave Rogers got his first taste of the rave scene, and has been passionate about DJ'ing ever since. Resident DJ at the Vertigo club in Boston from 2000 to 2003, he has since made appearances in clubs and on radio stations across the USA, playing alongside DJs such as Josh The Funkyone, Tym Ryan Macguyver, Ashley Cassell, and Manolo. In 2004 he made a move to Bristol in the UK, where he now lives, and he's been with Globaldancefloor agency since then. In addition to DJ'ing, Dave has been producing his own music for the last six years, and his influences include DJ Macguyver, Venom, Josh The Funkyone, Badboy Bill, Junior Jack, and Armand Van Helden.

W www.djdaverogers.com

Need Help With *Your* Mix?

If you're having trouble with a mix, then you can submit your track for the Mix Rescue treatment. Either email an MP3 file of your mix to the address below, or post a CD to Mix Rescue, *Sound On Sound*, Media House, Trafalgar Way, Bar Hill, Cambridge, CB3 8SQ, UK. Please include a daytime contact telephone number, some information about how you recorded and mixed your version of the track, and your views about what aspects of your mix are causing you most concern.

E mixrescue@soundonsound.com

to a repeated tinkly analogue piano phrase, which worked fine, but that was about all until the main beat kicked in and the sampled lyrics started up. To give the intro some dynamics, I recorded a lot of distorted guitar effects such as harmonics with tremolo bends, short riffs, clicks, and scrapes, and then cut these up to create interesting musical punctuations. I also added a very sweepy string-synth melody line over the intro to set the mood for the piece.

My final addition, some whimsical, breathy vocals, were taken from a sample CD and were arranged one phrase per note for easy triggering. They followed the style of, as Terry Pratchett might have it, ladies of negotiable affection going about their business! On their own these seemed just a bit too overt, but by adding tempo-sync'ed echo via a send and then chopping the dry sound up into eight slices per bar using *Logic*'s tremolo/panner set to square-wave modulation, the entire part took on a much more textural role. Only the first phrase was left unchopped, as it seemed a nice way to lead into the song. I also added tempo delay to a tortured guitar harmonic that comes in beneath the intro. What started out as a rather long and stark intro now only had two bars of completely exposed rhythm loop before the other parts started to sneak in, which for me helps create an atmosphere of tension and anticipation prior to the main rhythm part coming in.

Dave had brought in the main part with a cymbal crash, but I replaced this with a reversed cymbal. Although far from original, this does create the right kind of rush as the main section breaks in. I also took the liberty of replacing the bass sound Dave had originally chosen (using an Access Virus bass underpinned by an analogue sub-bass sound from the *ES M* synth), and I tidied up the timing of a few of the notes where the quantising seemed to have gone awry. Some serendipitous guitar clunks were chopped out, copied (four to the bar) and then reversed (with added echo) to provide a short rhythmic distraction beneath the intro, and this was repeated during the second measure of the main song. Dave had also played a soft organ-like pad beneath the body of the tune, and this worked perfectly well as it was.

Creating Reverse Reverb

As soon as the main section of the song starts, there's a repeated sampled vocal phrase 'Letting go of all your pain', with added delay, which sounded fine. However, I wanted to make it sound more dramatic to get the song off to a powerful start. Dave had asked me how you create the reverse reverb effect where the reverb seems to start before the word that triggers it, so I explained the old analogue method and then set about doing a digital recreation.

In the days of analogue multitrack tape, you'd turn the tape over so it played backwards, feed the track in question through a reverb device (often a plate), then record the 100 percent wet

Remix Reactions

"I'd like to say thanks to Paul and the *SOS* team again for a great job. When I went to see Paul I felt the mix was not right — it needed something. Paul had some fresh and great ideas to implement in the mix, as well as new sounds and techniques. I was amazed how just the right sounds put in the proper spots made such a difference. Paul also showed me some really useful setup tips for *Logic*, for example organising my mixer so I could see all the tracks in one go. Now the mix sounds full of body and life. Paul also explained to me that with the proper compression I could get a lot out of my drums. Thanks again!"

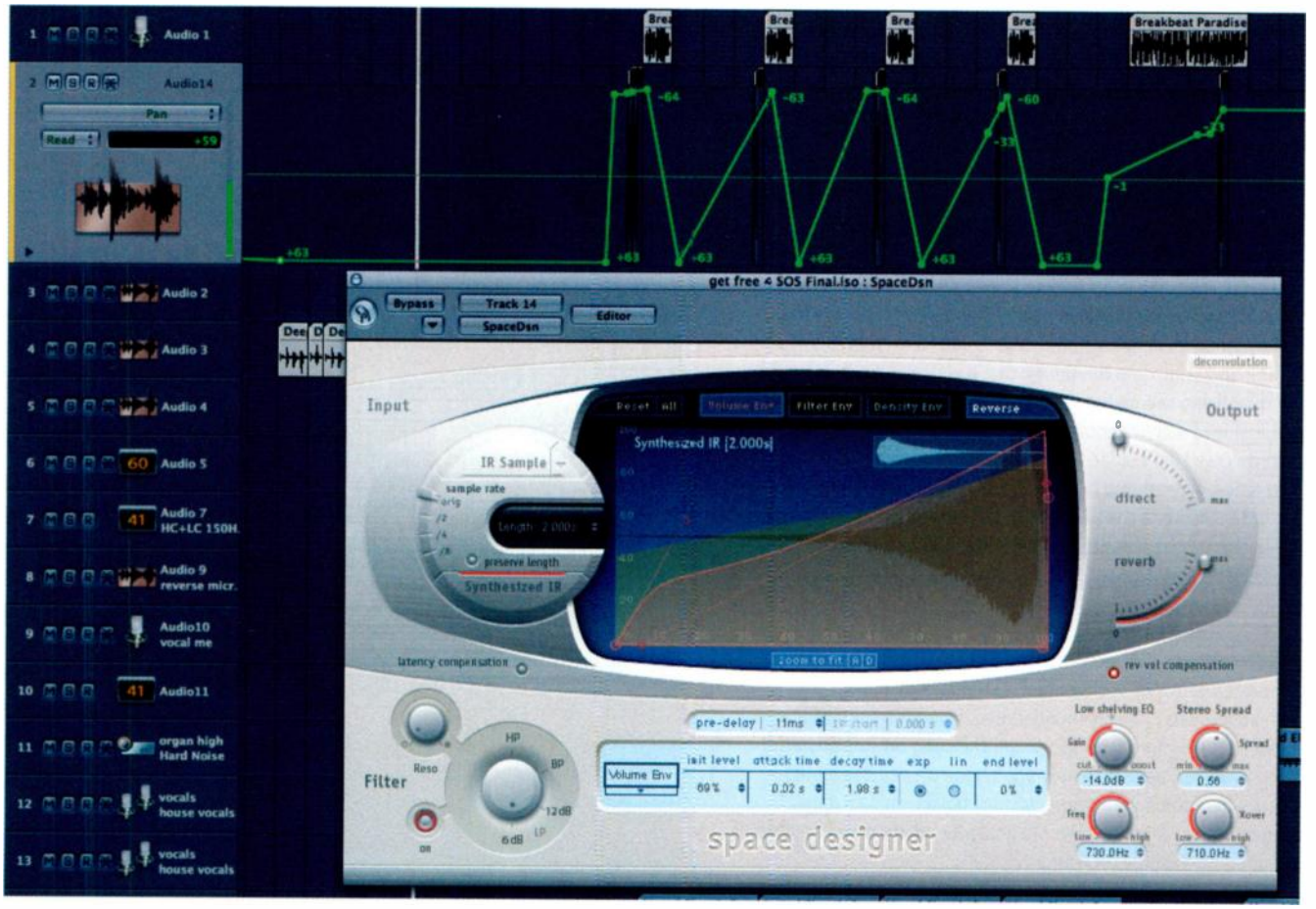
▶ reverb onto a spare tape track. Having done this, you'd put the tape back onto the machine the right way around, play it and there would be your reverse reverb, swelling in before each vocal section. Using *Logic*'s *Space Designer* digital reverb, I set about recreating the same effect, and if you're interested in trying it yourself, these are the steps I followed.

The first job is to copy the original vocal audio onto a new track, then insert a suitable reverb processor into that track. I say suitable, because the ideal choice is a convolution reverb with the option to reverse the direction of the Impulse Response, though a synthetic reverb with a fake reverse reverb envelope can also give satisfactory results. In *Space Designer* I simply loaded a plate reverb of around four seconds long and then hit the Reverse button. Your reverb should be set to 100 percent wet, and if you solo it and then hit play, you should hear the familiar reverse reverb effect. All that's left to do now is slide your new audio track to the left so that it is two or three seconds ahead of the original dry audio part, then play both parts together. Once you can hear both parts, it is easy to fine-tune the position of the reverb track so that it swells in naturally just before the dry vocal track comes in. This only took a few minutes to do and it worked perfectly.

I could have used the reverse reverb effect just as it was, but to make the entry even more dramatic, I passed the reverb track through a slow, deep flanger and then drew in some panning automation data so that the reverse reverb would whoosh from one speaker to the other. The other thing I did was shorten the audio parts triggering the reverse reverb so that the effect wouldn't go on too long and swamp the main vocals — I wanted it to whoosh in and then fade away each time. A couple of guitar improvisations were mixed underneath this section at a fairly low level just to add some variation.

Vocal Processing

After going around the vocal phrase four times, I cut to Dave's first breakdown section, which was based on a longer sampled phrase taken from the same library and clearly treated with *Auto-Tune* (or similar device) turned up to 'Stun' so as to reproduce the old warbly cliché. Dave had already passed this through the eight-to-the-bar chopper as I'd showed him how to do during our Studio SOS visit, and in all it worked well for the part. To give this a little rhythmic underpinning, I took a very breathy sample and triggered this from 16th notes, which produced a kind of human hi-hat. In the second half of the section I also copied in the rhythmic guitar clonks again, which helped lead into the next full-on section, and I also dropped in another



A reverse reverb effect was created by reversing a four-second plate impulse response in *Logic*'s *Space Designer* convolution plug-in. A flanger was then applied to the reverb to make its entry more dramatic, and the pan control of the reverb return was also automated.

of those reverse-reverb vocal snippets to give a suitable build up, culminating in the first downbeat of the next part. Rather than throw everything back in at once, I left it eight bars before the drums came back in, giving the bass the chance to carry the rhythm. As the drums come in, there's also a perfectly suited, sparse analogue 'plinky' riff that Dave came up with, which really helps the track build.

Again there are vocal samples over this section, which eventually leads up to a new drop-down section heralded by the same vocal and reverse reverb part we heard right at the start of the song. Dave had also placed a crash of thunder here which I left so that it would roll on when the vocals faded. This time a bass part and a rhythm part carry on through the break with the reintroduction of some wailing guitars in a supporting role. After four bars this brings us back into the drastically shortened end section which has more variations on the sampled vocal phrases to carry it along, plus the introduction of my repeated siren-like guitar tremolo bend for the final eight bars. The song then comes to a fairly abrupt end with a low, echoed guitar note carrying on for a while to produce a low 'D' drone that simply echoes into the distance and dies.

To beef up the overall mix I used the PSP Audioware *Vintage Warmer*, but other than the tempo tremolo chopper and the reverse reverb, there were no really unusual effects.

Hear The Differences For Yourself!

Listen to the changes I made by checking out the following audio examples available for download at www.soundonsound.com/sos/aug06:

- **/audio/ReverseCymbal.mp3**
- **/audio/HumanHiHat.mp3**
- **/audio/EffectedElecGtr.mp3**
- **/audio/LayeredBass.mp3**

Some of the new 'ear candy' parts I added to the track for interest.

- **/audio/OriginalVoxSample.mp3**
- **/audio/ProcessedVoxSample.mp3**
- **/audio/OriginalBreathyVox.mp3**
- **/audio/ProcessedBreathyVox.mp3**

My creative processing of two different vocal parts which appear in the track.

- **/audio/OriginalMix.mp3**
- **/audio/Remix.mp3**

Dave's original mix, and my shortened remix.

Noveltech's *Character* was used to add a bit of bite to an FXpansion *BFD* drum line, and only two send effects were used — a basic reverb on send one and a tempo-locked delay/echo on send two. Some of the distorted guitar parts were compressed to keep the level even, and I rolled off some low end from a funky wah guitar part Dave had brought in somewhere in the middle of the song. Dave had used a couple of *Logic*'s low-horsepower *Silver Reverb* plug-ins inserted on certain of his parts and I left them as he'd set them.

Final Thoughts

My final mix is obviously far too short for this type of track, but I hope it serves to demonstrate the little production touches I've talked about and also underlines the benefits of adding 'real' performance sounds where possible (in this case the electric guitar) to counter the sterility that so often comes from working entirely with samples, loops, and soft synths. The beauty of dance tracks like this one is that there are countless right ways to produce and mix them, and they provide a great playground for experimenting with new production ideas. SOS

CALL 01837 55200 OR VISIT

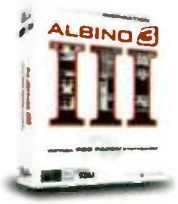
YOUR NO 1 SOURCE FOR AWARD-WINNING

Best of...

ALBINO 3 has loads of amazing new creative features like the two new filters "Scream" and "Comb"; four FX units with new FX like LoFi and WahWah; a Compressor and an improved Reverb.

But the biggest new feature is that you can layer up to four instances of Albino in one single preset!

There are more than 2100 'Rob Papen' presets in various styles plus two banks by the very talented REYN (Famous Dutch producer), Noisia (Dutch DnB) and NarvanaX (Dutch Hardcore).



Albino3 is available now for just £132.95.

Registered users of Albino2 can upgrade at www.robpapen.com

Also avilable by Rob Papen is the cross-fusion virtual synth 'BLUE'.

Blue is now in version 1.5 and is also £132.95.

ROB PAPEN
INSPIRATION SOUNDWARE

EZ DRUMMER (TOONTRACK)
EZdrummer is a multi-microphone drum sampler designed for musicians and producers in need of a compact, affordable, easy to handle plug-in without compromising sound quality or control. The interface combines auditioning of sounds and drumkit construction. The drag and drop midi-library (inc 2000+ midifiles) enables users to create a drumtrack in just a few clicks.
VST+AU INSTRUMENT - £99

STRADIVARI SOLO VIOLIN (GARRITAN)
The Stradivari Violin is regarded by experts as the best violin ever made and a genuine Stravidivari violin is worth millions. An instrument of this distinction required a unique way to faithfully capture and recreate its sound quality and playability. "Sonic Morphing" technologies provided the answer. The result is extraordinary levels of realism, expressiveness and performance.
KONTAKT 2 DVD-ROM - £129

ARTIST COMPLETE (BEST SERVICE)
Artist Complete combines previous sample libraries Artist Drums, Artist Grooves and Artist Rhythm plus Artist Drums 5.1 Surround Edition + additional GM Kits and Real Giga Drums multichannel drumkits. A huge collection of acoustic drum grooves and kits from seven of the world's best drummers. Also includes a large number of MIDI files.
VST+AU+DX+RTAS INSTRUMENT - £166

ORIENTAL RnB (UEBERSCHALL)
Oriental RnB is the ultimate point where the east and the west intercept on the highest musical and sonic level. This very point is touched by spicy oriental flavour combined with modern dance, pop and RnB tunez. Oriental RnB comes with Ueberschall's brand new "Elastik Engine" technology with all the functionality which is needed for professional use.
VST+AU+RTAS INSTRUMENT - £69

PERSONAL ORCHESTRA (GARRITAN)
Personal Orchestra contains samples of all the major instruments in a symphony orchestra: strings, brass, woodwinds, and percussion, as well as a Steinway concert grand piano, a Stradivarius violin, concert harps, a Rudolf von Beckerath concert pipe organ, a harpsichord, and many other exquisite instruments.
VST+AU+DX+RTAS INSTRUMENT - £169

JAZZ & BIG BAND (GARRITAN)
From the producers of the award-winning Garritan Personal Orchestra comes this comprehensive Jazz & Big Band collection featuring Saxophone, Keyboards and a superb Rhythm Section - over 50 individual instruments in all. Now you can create realistic-sounding Jazz & Big Band arrangements quickly & easily.
VST+AU+DX+RTAS INSTRUMENT - £169

CHRIS HEIN HORNS (BEST SERVICE)
An outstanding sampled brass instrument library by Chris Hein. A huge library of over 3.5 Gb / 20,000 samples consisting of Alt-Sax, Tenor-Sax, Trumpet, Trombone & Trumpet-Section. Up to 44 different playing techniques each presented in one program with up to 44 keys switches. Up to 20 velocities per note with up to 189 variations.
VST+AU+DX+RTAS INSTRUMENT - £232

STRING ESSENTIALS (BEST SERVICE)
Peter Siedlaczek's comprehensive virtual string orchestra! You get over 17Gb of string ensembles from 14 violins, 10 violas, 8 cellos & 6 double basses. With key switching and instant access to the all important articulations, String Essentials will enable you to focus on the most essential part of your creativity - your music!
VST+AU+DX+RTAS INSTRUMENT - £232

OUTER LIMITS (ZERO-G)
Ian Boddy's follow-up to his critically acclaimed Morphology Kompakt instrument but this time he has delved deeper and darker than ever before. 4Gb of shimmering pads, to mysterious drones, menacing mechanisms to strange vocal synthesis! It is a journey through dreams and nightmares, full of sounds beyond your imagination.
VST+AU+DX+RTAS INSTRUMENT - £115

VOCAL FORGE (ZERO-G)
A uniquely inspiring new library featuring 800 patches and over 2,500 samples of brand new material sourced entirely from pro vocal performances. Male and female MCs, sweet R+B, gospel singers, dance and pop vocals all recorded with accompanying harmonies. Plus there is a vocal Toolbox loaded with single lines, FX, etc.
VST+AU+DX+RTAS INSTRUMENT - £115

REAL GUITAR 2L (MUSIC LAB)
RealGuitar 2 is here! Now you can create realistic solo and accompaniment guitar parts quicker and easier than ever before! RealGuitar 2L version is equipped with the powerful MIDI Pattern Library for easily creating guitar accompaniment tracks. RealGuitar allows you to perform guitar parts with a whole new level of expression.
VST+AU+DX+RTAS INSTRUMENT - £133

IVORY (SYNTHOGY)
Three Glorious Grands in one multi-platform Virtual Instrument. The largest, most skillfully designed Grand Piano virtual instrument ever! Over 4000 samples are combined with a sample-playback engine built from the ground up to bring out the resonance, response and character of the world's finest Concert Grands
VST+AU+RTAS INSTRUMENT - £189

HIP HOP UNDERGROUND (UEBERSCHALL)
Tight groovez, Dope beatz & Mad Flava! With over 140 high quality construction kits and more than 2.2 Gb of Content. Look forward to enjoying a massive Library jam packed with Drums & Percussions, Basses, Chords, Vocals, FX & tons of single drumsounds. The sound library comes along with the brandnew "Elastik" loop player.
VST+AU+RTAS INSTRUMENT - £69

60s A GO GO (UEBERSCHALL)
Authentic retro loops & samples inspired by lounge, easy listening, french and italian film music from the faboulus sixties. Played on real vintage instruments and amps! Over 1.5 GB sample material to give your productions real 60s retro vibrations. The sound library comes with the brandnew "Elastik" loop player.
VST+AU+RTAS INSTRUMENT - £69

Time+Space, PO Box 4, Okehampton, Devon EX20 2YL. Prices include 17.5% VAT. UK mainland carriage £2.95 or £5.95 next working day delivery. International £10.

WWW.TIMESPACE.COM

TIME+SPACE

VIRTUAL INSTRUMENTS AND SAMPLE LIBRARIES

ETHNO TECHNO XPANDER (ILIO)
Ethno Techno features rare instruments and unheard of rhythms and textures, with a combination of Bashiri Johnson's (Whitney Houston, Miles Davis, Madonna, Saturday Night Live) inspired live performances and additional remixes by Eric Persing. From broad and spacious ambient explorations to radical heart-pounding beats, all with a unique organic flavor.
S.A.G.E. XPANDER - £55

STARK RAVING BEATS XPANDER (ILIO)
Aptly described as both fast and furious, this library features aggressive live drums, wild percussion and tricked-out remixes. The original groove tempos start at 105 BPM and accelerate to a heart-pounding 170, but using SAGE™ (Spectrasonics Advanced Groove Engine) technology, you can expand that range as far as you like, always with intriguing results.
S.A.G.E. XPANDER - £55

SKIPPY'S BIG BAD BEATS XPANDER (ILIO)
Skippy uses his wild imagination to create Hip Hop, Trip Hop, and other dance groove styles that come in a variety of flavors, including smaller mixes, no-kick, no-snare, and even breakdown versions! The mix variations allow you to stick to the basics if you're just looking for a simple, slammin' beat, or you can add as many elements as you like to create grooves that demand attention.
S.A.G.E. XPANDER - £55

SKIPPY'S NOIZBOX XPANDER (ILIO)
Hot techno-club beats, a hard driving Euro attitude, innovative and cinematic sounds and textures set this collection apart from any other. Noizbox is an explosion of sonic fireworks with a huge variety of hits, beats, and sound-design elements that will add a whole new dimension to your grooves, especially when combining them with elements from the Stylus RMX™ core library.
S.A.G.E. XPANDER - £55

FIRESTORM BY DJ STAKKA (ZERO-G)
The ultimate Drum & Bass sample 'beast' produced by long time veteran of the scene DJ / Producer Stakka. This massive 24-bit 1Gb library has been 2 years in the making & puts cutting edge production at your fingertips. Over 950 samples in numerous formats, it is packed to the brim with fresh and exciting sounds.
ACID/WAV+EXS+HALION+KONTAKT +REASONREFILL+REX - £59.95

TOTAL REX (ZERO-G)
The world's largest collection of professional REX files in a single product! TOTAL REX features OVER 15,000 loops / 10 GIGABYTES of groove inspiration! These samples can be used with any application that recognizes the .REX file format including Stylus RMX, Reason, Cubase, Logic, Kontakt and many more!
REX DVD-ROM - £99.95

URBAN ELEMENTZ (ZERO-G)
Over 1350 samples of the coolest, freshest hip hop and R&B in the business - direct from NYC. Urban Elementz is a brand new collection of East Coast Hip Hop and R&B loops and samples crammed full of earth-breaking beats, cool keys, pimpin' basses, slinky guitars, synths, strings, licks, fills, percussion, fx etc!
ACID/WAV+EXS+HALION+KONTAKT +REASONREFILL+REX - £59.95

BROKEN BREAKS (LOOPMASTERS)
The ultimate collection for breakbeat enthusiasts, bringing an incredible collection of future classic beats and breaks to studios worldwide. Features over 770 brand new Breakbeat Samples, over 450 single hit drum samples and 44 pre-programmed Super Funked Drum kit patches ready to load directly into your sampler!
ACID/WAV+EXS+HALION+KONTAKT+NNXT+REX - £39.95

DRUMMEAR (TEKNIKS)
DrummEAR is Tekniks first DVD release dedicated to drum loops and hits for Logic's powerful EXS24 Sampler. Recorded in various studios throughout Norway using the very best mics, outboard equipment and desks for quality sonic delivery. As well as the 5 multisampled kits there is also a huge amount of grooves to get you started.
EXS24 DVD-ROM - £59.95

DEEP TRANCE & TECHNO (LOOPMASTERS)
Dedicated to Trance and Techno, featuring a mind blowing collection of over 1,500 brand new samples ready to fuse into your tracks with minimal fuss and maximum impact. Heavy Club beats, Soaring Pads, Filtered Arpeggios, Pulsating Basslines and Full Fat Samples! Inspired by Anjunbeats, Platipus Records, Nebulae and Nukleuz.
ACID/WAV+EXS+HALION+KONTAKT+NNXT+REX - £39.95

RUSH (BIG FISH AUDIO)
If you want the latest sounds for your Progressive House and Trance productions, then what you need is Rush! From ethereal synths to driving bass, all the material you need to get your tracks into the biggest clubs is right here. Over 1.3 gigabytes of pure adrenaline, plus a bonus folder of 420 extra drum hits and loops.
APPLE LOOPS+REX+WAV - £55

CHILL (BIG FISH AUDIO)
Downtempo Loops and Beats! Enter into a different state of mind with Chill. This loungin' set of construction kits draws from jazz, hip hop, dub, funk, soul, ambient, and pop to create a whole new downtempo experience. Drums, guitars, keys, electric and acoustic bass, flutes and more gel somewhere between 60 and 110 bpm.
APPLE LOOPS+REX+WAV - £55

THE COLLECTION (NINE VOLT AUDIO)
This triple DVD collection contains the complete Downtempo Guitars Vol 2, Electronic Guitars and Chopped Guitars libraries. An indispensable collection of unique and flexible loops suitable for virtually any musical occasion. All loops have been recorded in 24-bit resolution and are presented in Stylus RMX, REX2, WAV and Reason ReFill formats
ACID/WAV+REX+REFILL+RMX - £95

LOUNGIN HOUSE (EQUIPPED MUSIC)
Want to delve deeper into the naked, intimate sound of House? Welcome inside the hypnotic and seductive grooves of Loungin' House. Luxurious dance floor beats, mixed with sexy latin style percussion, lush chords and more. With over 4Gb of material, this is the ultimate House groove toolbox for Stylus RMX and Reason users!
REX2+WAV - £79.95

Liquid instruments

Horn Section

£117.00

Unbelievable but true: no other plug-in lets you create such a flexible, authentic-sounding horn section as easily! Key, tempo, scale - everything at the touch of a button or even automatically synced. Even the single notes of a single voice can be individually worked like MIDI. Uncountable variations within seconds!

This is possible through a combination of live-recorded licks and the unique Melodyne technology from Celemony. The horn sections consist of baritone, tenor, and alto sax, trombone, trumpet and flute and were recorded with studio professionals playing high-quality instruments using no hall effect.

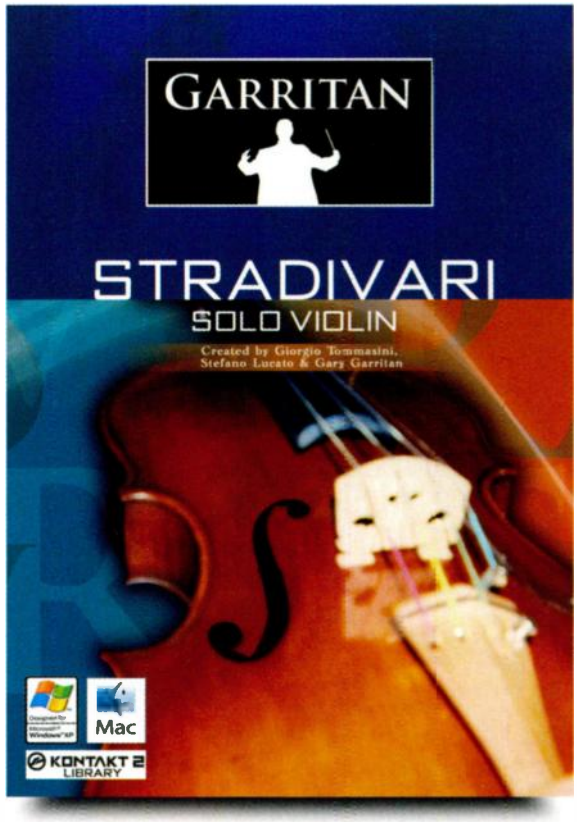
The styles run from 60s/70s Soul and Funk, Disco, 80s Pop, 90s Acid Jazz, Nu Jazz, Rock, Funk, House, RnB, HipHop up to the most current hit productions.

liquid instruments 1.1 available now

((•)) ueberschall

Garritan Stradivari Solo Violin

Virtual Instrument For Mac & PC



Gary Garritan's powerfully expressive solo violin instrument features some impressive technical innovations. Will it create a new generation of virtual Paganinis?

Dave Stewart

The DVD given away with *SOS* August 2005 featured an intriguing video clip of a man producing an amazingly lifelike and expressive solo violin performance from a bog-standard MIDI keyboard. Even allowing for the player's tasty keyboard technique, the soaring, emotive legato phrasing, dynamic response, swelling vibrato, and expressive portamento slides of the instrument were reproduced so accurately that it was hard to believe that some kind of sorcery was not involved. The man in the clip was Stefano Lucato, filmed playing Gary Garritan's *Stradivari Solo Violin* instrument live at the Frankfurt Musikmesse 2005 trade fair. This remarkable instrument is now available to all you budding Paganinis, retailing (I'm glad to say) at a considerably lower price than a real Stradivari violin. By the time you read this, the initial *Kontakt 2* version should have been superseded by a functionally identical *Kontakt* Player, which will also open up in *Kontakt 2*.

One Man's Industry

When it comes to sampling orchestral instruments, Gary Garritan has a long and distinguished track record. His 1999 *Gigaharp* library was one of the first to take advantage of *Gigastudio*'s unlimited sampling time. In August 2001 Garritan's *Orchestral Strings* caused a stir by offering 8GB (at the time an unprecedentedly large figure) of sampled string sections. This was followed by the critically acclaimed *Garritan Personal Orchestra*, which endeared itself to impecunious musicians and composers everywhere by squeezing the full instrumentation of an entire symphony orchestra into a 2GB player costing less than £200 in the UK. Although something of a stylistic departure, *Jazz & Big Band* was no less musically effective, but now the *Stradivari Solo Violin* (created by Giorgio Tommasini, Stefano Lucato, and Garritan himself) sees our man resuming his lifelong love affair with orchestral stringed instruments.

Someone once said that if all the people in the music business who were in it for the money were to pack up and leave, the industry would become a far healthier and downright more enjoyable place. Although he displays canny market awareness, Gary Garritan is clearly not out merely to make a buck, and his web site proves it: bristling with artistic and educational features (including over a thousand user demos, an interactive edition of Rimsky-Korsakov's *Principles Of Orchestration*, and a radio station featuring contributions from over 180 composers producing music using Garritan sounds), the site has a strongly supportive community feel. The most telling link is with the Drake Music Project in Edinburgh, which enables people with disabilities to play and compose music — using *Stradivari*, a program called *E-Scape*, and pressure sensors to control expression and vibrato,

Rhona Smith, an 18-year-old woman with cerebral palsy, has created a moving version of Massenet's 'Meditation' (www.garritan.com/mp3/Rhona-Massenet.mp3).

Like many manufacturers, Garritan is never slow to talk up his products' new features with a juicy bit of incomprehensible jargon. This time, the claim is that *Stradivari* uses 'real-time sound shaping' rather than 'freeze-dried samples', or if you prefer, 'morphing samples combined with instrument body resonances and intelligent performance scripts make the gestalt of these innovations something unique'. That sounds terrific, but what does it mean? In a nutshell, the idea is that rather than supplying a large number of disparate multi-sampled performance styles that have to be pasted together in a sequencer, the instrument is designed to let the player create a wide range of articulations in real time using combinations of controllers — to put it simply, you actually play the instrument as opposed to programming it. To do this in a musically effective and controlled way requires a slight learning curve, and also some adjustment of your expectations about how sampled instruments behave.

Homer At Moe's

I got a first-hand taste of this when I first played *Stradivari*: the initial pleasant surprise was that, rather than supplying the usual endless list of playing styles, the library (which contains about 520MB of mono samples) consists of just one program — from a reviewer's point of view, that was quite a relief! The next revelation was less welcome; despite approaching the instrument with the highest of artistic intentions, my first few notes sounded like Homer Simpson attempting to play the violin after an evening at Moe's Tavern. All I could muster was an unimpressive collection of short bow attacks which, unlike our cartoon hero, could only be described as unanimated. Reading the manual put me straight: as with all of Garritan's recent libraries, a MIDI controller has to be employed to activate the instrument's full sound. In the case of *Stradivari*, you need to use a MIDI expression/volume pedal to bring in the sustains, having first adjusted your keyboard settings so that the pedal outputs MIDI Continuous Controller number 11 (CC11), the Expression controller, rather than number seven, the more normal Volume controller. Once I'd hooked the pedal up, the violin began to come to life.

SOUND ON SOUND

Garritan *Stradivari Solo Violin*
£129

pros
- It sounds uncannily like a real violin, with highly realistic legato, vibrato, portamento, glissando, and trills.
- Seamless morphing between four dynamic layers enables smooth-as-silk dynamic transitions.
- The controller options allow very expressive and emotive violin performances.
- Very good value for money.

cons
- The tremolo simulation is not convincing.
- There are no long glissando, col legno, or artificial harmonics samples.

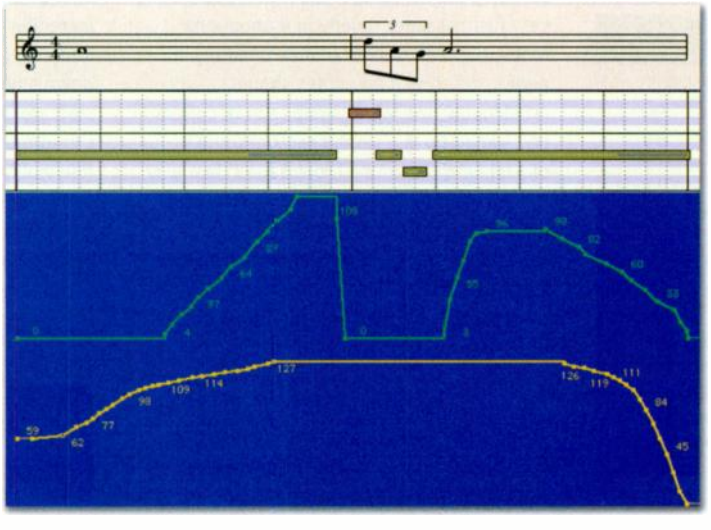
summary
If you're working with a modest budget and need a solo violin to add to an orchestral arrangement or to play on its own, Stradivari is the perfect solution. Although not a good bet for one-fingered programmers, it's a keyboard player's dream — it feels and sounds like a real instrument, and its expressive powers are second to none.

The single *Stradivari* program consists of a set of 10 keyswitches which select various playing modes and articulations. Keyswitch C1 activates 'mono legato' mode, which uses short bow attacks layered over the sustained notes. Note velocity determines the bow attack volume and sharpness of attack, while the level of the sustains is controlled by the footpedal. (This layering technique was first done by Denny Jaeger back in the '90s, and remains valid today: the initial bow attack covers up the relatively quiet start of the sustain sample, which in turn comes to the fore once the bow attack has subsided.) Keyswitch F1 substitutes a longer, louder downbow attack for the lighter spiccato attack used in keyswitch C1 — this heavier bowing works well at the beginning of a phrase, or for adding emphasis to a note. If you want staccato notes, you just play them that way — when you release a note, the spiccato bow attack (which uses a 'round robin' rotation of four takes) gives way to a release trigger sample of the bow bouncing off the string, creating a realistic simulation of short-note bowing.

Via some clever under-the-bonnet programming, made possible by *Kontakt 2*'s MIDI scripting facilities, the mono legato mode temporarily switches to polyphonic mode for up to four simultaneous notes if the time interval between the notes is less than 20 milliseconds — this means that you can insert a two, three or four-note chord in a legato line without changing playing mode. However, if you need to play multiple non-synchronous pitches, you can switch the *Stradivari* over to polyphonic mode by selecting keyswitch D1.

Weapons Of Mass Expression

Stradivari packs three innovative secret weapons, which together make it sound much more realistic than your average sampled violin. The first of these affects dynamic transitions: if you hold a long note and push up the expression pedal, the instrument moves through its four dynamic layers, cross-fading between the different sustain samples from *pp* to *ff* with no trace of phasing, chorusing, layering, break in the tone, or other unwanted artifacts — the transition is as smooth as silk, and sounds exactly like a violin player exerting more bow pressure while sustaining a note. This is achieved by a new patent-pending technique called 'Harmonic Alignment', developed by Giorgio Tommasini. In the inventor's words: "All samples corresponding to different dynamics of the same note are processed in order to temporally match the phase of the corresponding harmonics. This results in a set of sounds timbrally indistinguishable from the original samples, which can be played simultaneously with no phase artifacts." I've no idea how it is done, and haven't come across this phase-matching idea before, but can confirm that the sonic results bear out the makers' claims. ▶



Here you can see a violin phrase I played into Apple *Logic*, varying the amount of vibrato (green line) and the volume swells (yellow line) simultaneously using mod wheel and a footpedal. Note that I've used no vibrato on the three short notes to simulate how a violinist might negotiate this phrase.

Alternatives

Stradivari Solo Violin is unique in offering a truly solo solo violin — most libraries provide the instrument in a package comprising solo violin, viola, cello, and double bass. The *Advanced Orchestra* set created by Peter Siedlaczek in 1997 had a solo strings volume which is no longer separately available, but Dan Dean's *Solo Strings* is going strong in *Gigastudio* format. The East West/Quantum Leap *Symphonic Orchestra* strings volume has a solo violin in amongst its string sections — the instrument is improved in *Symphonic Orchestra*'s *Pro XP* expansion set, but that works only in conjunction with the original library and has no separate strings volume. The main contemporary alternatives to Garritan's product are Big Fish Audio's so-called *London Solo Strings* library and Vienna Symphonic Library's *Solo Strings*. The latter is available as a Horizon-series sound library and in expanded form as a Vienna Instrument.

The second innovative feature concerns vibrato. Mod wheel-driven vibrato tends to produce an unpleasant pitch wobble which degenerates into a ghastly siren-like wailing when the wheel is pushed all the way up. This synthetic racket bears little resemblance to real-life vibrato, and of late some orchestral-library manufacturers have admitted as much by dispensing with it altogether. However, via some more of Tommasini's technical trickery, *Stradivari*'s vibrato sounds extremely lifelike. Apparently the vibrato's pitch modulation effect is created by a LFO as usual, but an additional process ('convolution with a modal impulse response') recreates the characteristic amplitude changes (also known as 'tremolo') which occur in real vibrato. Again, I'm not clear how this works, but I am happy to say that it definitely does! The default vibrato speed can be adjusted via MIDI CC67 — you can also temporarily increase the speed of the vibrato by using aftertouch, but I found this rendered the effect somewhat less lifelike.

Save All Your Glisses For Me

Stradivari's third ace in the pack is its portamento (glide) effect. Unimpeded by frets, violinists often slide between notes, sometimes using small, subliminal pitch glides, but occasionally introducing a big glissando for obvious dramatic effect. Drawing on the same convolution technology devised for its vibrato (which, I'm told, introduces formant shifting on pitch changes) and some more clever MIDI scripting, the violin's portamento slides sound utterly convincing for both subtle and unsubtle types of delivery. However, making them obey your musical will is another matter, as I'll try to explain!

Portamento pitch glides are automatically created between overlapping notes when *Stradivari* is in mono legato mode. The longest, most pronounced glissandos are created when a sustained note overlaps with a second quiet note — if the second note is played louder, a quicker portamento occurs, to the point where a ff overlapping note produces no portamento at all. This is somewhat counter-intuitive — the tendency is to try to create a big glissando by playing a loud note, rather than a quiet one — but after a little practise I began to get the hang of it. The musical effect is great, but my only criticism of this style is that the slowest glissando (from bottom note to top) lasts for no more than a second — it would have been nice to have a few samples of slower, more exaggerated glissando effects, if only for those *Tom & Jerry* musical-comedy moments.

One downside of the built-in portamento scripting described above is that if you play very fast passages quietly in monophonic mode, there's a danger of unwanted big glides occurring if any notes should accidentally overlap. Overlaps can of course be cleaned up in a sequencer, but for live playing in mono mode I found the safest way to avoid accidental portamentos in very fast passages was to whack my keyboard's velocity scaling up to maximum and play loudly! Alternatively, you can simply play the fast passage in polyphonic mode, which has no built-in portamento effect.

Using these highly expressive techniques, one can play legato melodies with or without portamento, using the expression pedal to vary the dynamics and the mod wheel to control vibrato. It is of course possible to sequence the notes first and add the expression data later, but that would be far less fun and take 10 times as long!

Additional Styles

Stradivari supplies the straight, four-dynamic sustained notes and two types of bow attack described earlier in straight and con sordino (muted) versions, along with two-dynamic pizzicato samples and a set of natural (in other words non-chromatic) harmonics. The effect of the mute is quite subtle; it reduces brightness and creates a slightly more mellow sound overall, most noticeably in the violin's lower register. Being recorded in mono, the pizzicatos lack the sumptuous stereo room acoustic one hears in some string libraries, but they're perfectly adequate for plucked note passages. It's a shame that no artificial harmonics or col legno sustains were included — the natural open-string harmonics are mapped only to their real-life pitches, leaving wide chromatic gaps across the keyboard.

The instrument has a few other tricks up its sleeve: its 'trills' mode lets you play trills as a violinist would, in other words holding down a note while rhythmically playing a repeated trill note. The main note is automatically muted every time the trill note sounds, and (unlike on a real violin) the trill can be played between any two notes you like. Once again, the musical effect is incredibly lifelike. This keyboard mode has interesting extra applications — by holding down a note and playing a series of fast staccato eighth notes above and below it, you can introduce an Irish or Eastern European folk violin flavour to melodies, and it's also useful for certain legato effects.

Unfortunately, the tremolo mode is less effective: it outputs a kind of light double-bowing (a bit like a single repeat on

Gimme Five

Never one to let the grass grow under his feet, Gary Garritan is about to release a solo *Gofriller Cello* to accompany *Stradivari Violin*. A sneak preview revealed that the cello has a superbly mobile, lively sound and matches *Stradivari*'s levels of expressive realism. Work is also under way on a solo viola and double bass, as well as a second violin (this one made by Guarneri). Garritan has spoken about applying the same approach to other orchestral instruments — if the idea goes ahead and pricing levels remain the same, it would enable users to gradually build up collections of good-quality, expressive solo instruments one instrument at a time without breaking the bank.

Test Spec

- 2GHz Athlon PC with 2GB RAM running Windows XP Pro.
- Garritan *Stradivari Solo Violin* v1.06 for NI *Kontakt 2*.

a delay line) to simulate one fast back-and-forth motion of the bow, after which it's up to you to play repeated notes to prolong the tremolo effect. I found it difficult to synchronise the timing of my repeated notes to the fixed speed of the built-in bow repeat — you can overcome this difficulty to some extent by editing a sequence of repeated notes to the right length and timing, but even then the tremolo effect doesn't sound very convincing.

The long note samples play for eight to 10 seconds and are unlooped, but it's possible to artificially extend their duration by using a virtual bow-change — you can do this in mono legato mode by playing a long note, pressing the sustain pedal, then playing the note again. This produces a smooth legato re-bowing, preceded by a very slight downwards pitch modulation designed to mimic the behaviour of the real instrument. I wasn't aware that re-bowing a string had any effect on its pitch, but must say that the pitch modulation is so slight that I wouldn't have noticed it had the manual not drawn it to my attention. Although smooth and fairly unobtrusive, the re-bowing will always be noticeable in a solo passage — however, if *Stradivari* is played in an ensemble and you pick the right moment to replay the note, it might just pass unnoticed.

Pitch bend (controlled by the pitch wheel, naturally) has been set to the obligatory two semitones — I don't know whether it's possible to adjust this setting in the *Kontakt* Instrument version of *Stradivari*, as it wasn't available for review at the time of writing. On the subject of bends, it struck me that it would be great if one could bend just the lower of two held notes — this would facilitate all manner of country and folk licks and even Jan Hammer-style guitar pitch-bend simulations, which would help extend the violin's stylistic range beyond the strictly classical. Having said that, the range of demos on Garritan's web site prove that *Stradivari* is already pretty stylistically versatile!

Cadenza

As you would rightly expect from a priceless, 300-year-old instrument, the Strad violin (played by Pauline Kim) has a rich, vibrant mature tone with plenty of depth and body. The bowing is smooth, assured, and free of inconsistencies right across the G2 to E6 range, resulting in a musically satisfying set of samples. When allied to *Stradivari*'s technical innovations and advanced programming techniques, the raw sounds are transformed into a sophisticated and intelligent instrument which somehow manages to be more than the sum of its parts.

Forget all the technical stuff — the most important fact about the *Stradivari* violin is that it's enormous fun to play, a salutary reminder of why we all got started in the music business in the first place! As ever, a little bit of keyboard technique helps, but though it can make you sound like a virtuoso, you don't have to study for years to play it. With a little practise, your fiddling will soon rival that of the Enron directors, and unlike Nicolo Paganini (the great 19th-century violinist whose phenomenal technique gave rise to rumours of diabolical assistance), you won't have to sell your soul to the devil to cover the cost of this expressive instrument. SOS

Information

- £ *Kontakt 2* library £129; *Kontakt* Instrument, £tbc. Prices include VAT.
- T Time + Space +44 (0)1837 55200.
- F +44 (0)1837 55400.
- E sales@timespace.com
- W www.timespace.com
- W www.garritan.com

DPA 4090 & 4091

Instrument Microphones

Hugh Robjohns

DPA have used their hi-tech miniature microphone capsules to produce a pair of mics that you can use on just about any sound source.

The new DPA 4090-series instrument microphones are very compact omnidirectional mics designed for both studio and live sound applications. They're based on the thoroughly proven and very highly-regarded 4060 series of miniature microphones — the familiar lavalier or personal mics widely used in broadcasting and theatres. Each of the matt-black 4090 microphones is supplied in a smart plastic case with a stand-mount (including a 3/8-inch thread adaptor) and a foam windshield (see page 146). The older 4060 mics have often been used for discreet instrument-miking applications, so adapting the mic to suit this purpose makes a lot of commercial sense.

4090 Specs

There are currently two models in the new series: the 4090 and the 4091. The essential difference between them is that the 4091 can handle much higher SPLs than the 4090, but in other respects they are virtually identical.

The microphones measure roughly 123mm in length, and are the same diameter as the XLR connector at one end — about 19mm — narrowing to about 7mm at the capsule. The 4090 model features a gold-coloured protective grille, while the 4091's grille is black — but in both cases the grille and capsule are spring mounted, to afford a degree of protection to the capsule should the microphone be accidentally dropped or hit.

The internal capsule of the 4090 mic is the same as that used in DPA's 4060 miniature microphone — a sophisticated back-electret design with an unusually high self-biasing static charge of about 400V. This rationale behind such a high static charge is that it allows a greater separation between backplate and diaphragm, while maintaining sensitivity. The larger separation reduces distortion with high sound pressure levels (SPLs). In DPA's design, a Teflon coating on the backplate stores the static charge, and careful heat-curing processes performed during manufacturing effectively 'age' the capsule, to ensure a long and consistent life. The days of electret capsules dying after a decade are long gone.

The specs of the 4090 are identical to the 4060; the only real difference is that the instrument microphone is designed to work directly from phantom power. The frequency response is specified as being flat between 20Hz and 20kHz (+/-2dB) and the

omnidirectional polar pattern is very tidy, with only the inherent physical shadowing effect of the mic body reducing the rear sensitivity by about 3dB at 10kHz. The output level is a healthy 20mV/Pa and harmonic distortion is below one percent up to 123dB SPL. Peak level clipping occurs at 134dB SPL and self-noise is very respectable for a capsule of this size, at 23dBA (reference 20µPa).

The 4091 microphone employs the DPA 4061 capsule, which allows it to accommodate much higher SPLs, and the specifications vary slightly to reflect this. The output level is 10dB lower at 6mV/Pa, allowing 10dB more headroom to push the clipping level up to 144dB SPL. The self-noise is also 3dB higher, at 26dBA.

Omni Advantages

Omnidirectional mics have several advantages over cardioids and other directional mics. The bottom end always sounds more natural and more extended, they don't suffer from the proximity effect and they're far less worried by wind blasts and mechanical vibrations transmitted through the stand. The disadvantage, of course, is that they don't discriminate between sounds coming from different directions, which can translate to unwanted spill.

However, one way around the spill problem is to position the microphone closer to the wanted source: the inverse square law of sound propagation means that you can make a big improvement to the ratio between wanted instrument and unwanted spill simply by moving the mic a little closer to the source. This approach can be problematic with directional mics, because the inherent proximity effect gives a disproportionate lift to the low frequencies, but as omnidirectional mics don't suffer from this you can place them very close indeed, to maximise separation without altering the frequency response. Of course, very close miking has its own drawbacks sometimes — particularly because the mic may no longer be able to 'see' the whole instrument, and thus capture an unbalanced representation of it — but no one ever said the art of miking musical instruments was easy! Another advantage of omnis for stage applications is that the polar response remains very uniform in terms of both level and frequency (unlike most directional mics), and this helps to minimise the risk of monitor feedback. ▶

SOUND ON SOUND

DPA 4090 & 4091 £346

pros

- Very smooth and extended frequency response.
- No proximity effect.
- High immunity to mechanical vibration and wind noise.
- Compactness allows easy placement.
- The 4091 offers an alternative for higher SPLs.

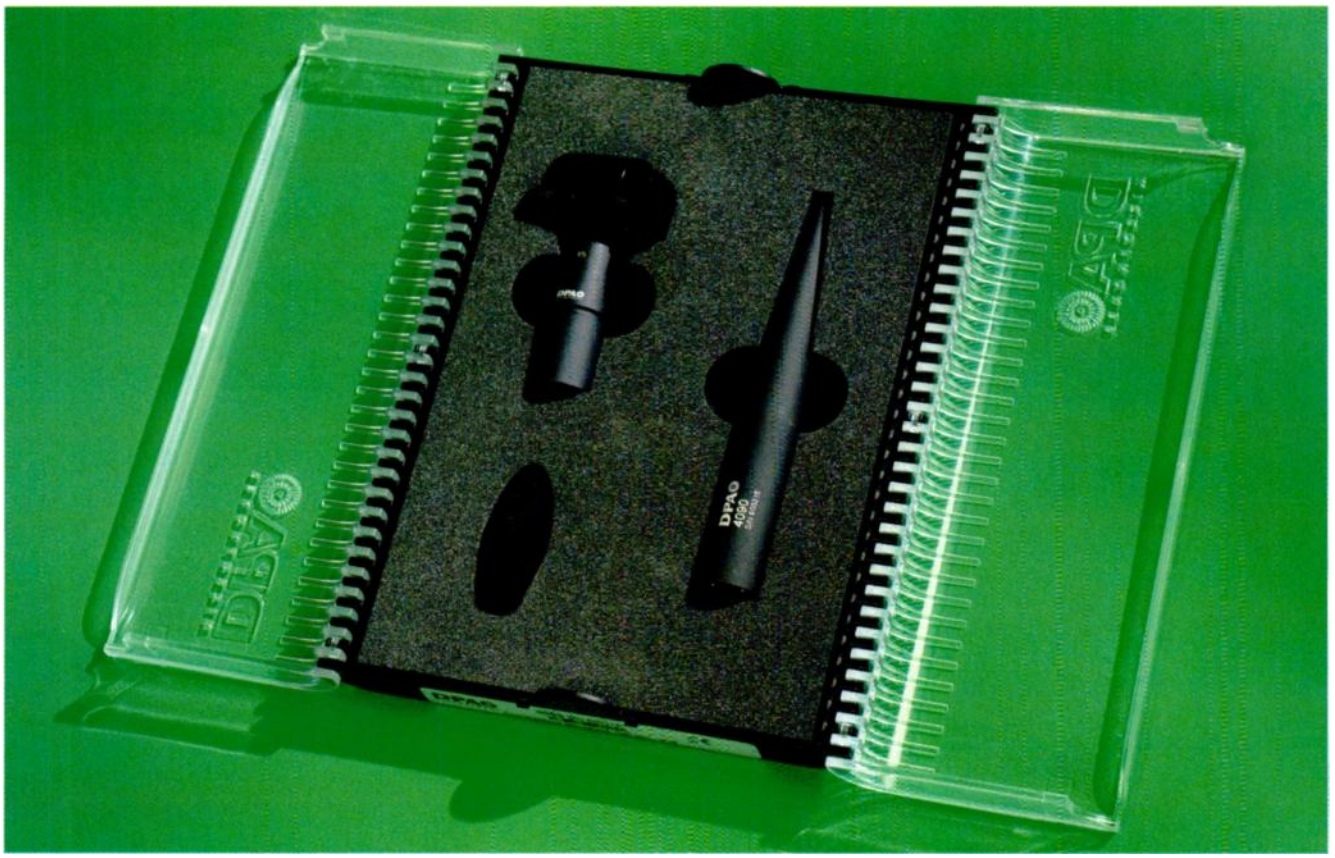
cons

- Slightly noisy if used at large miking distances.

summary

The 4090-series mics are capable of capturing a very natural and detailed representation of a wide range of musical instruments, while being easy to place close to the source to maximise separation. These compact omnidirectional electret microphones set the standard for this advanced technology.

DPA 4090 & 4091



In Use

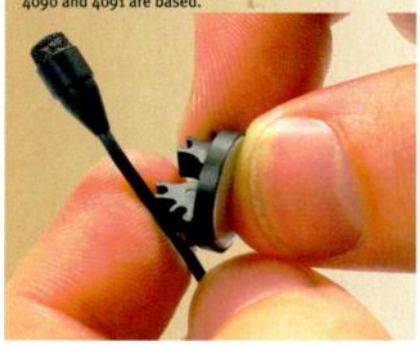
I have used DPA's tiny 4060 mics on numerous occasions, for discreet miking on guitars and in pianos, for example, and always with excellent results. The new 4090 and 4091 are very easy to position because they are so compact, and they look quite elegant and discreet on stage. I found the standard 4090 model easily capable of handling most sources without fear of overloads; it worked very well on acoustic guitars, solo violin and cello, and, in a stereo pair, as overheads on a jazz drum set. For louder sources, such as brass instruments, electric guitar amps and drummers playing in the more aggressive styles, the 4091 is more appropriate. Not only can it handle significantly louder SPLs, removing any fear of clipping, but the lower output level makes interfacing such loud sources with preamps easier too, minimising any risk of running out of headroom.

Both mics exhibit the same qualities, though — a very natural, open sound with an extended bandwidth, no hint of any off-axis colorations, excellent dynamics and fast, precise transients. The self-noise of a mic is related to the size of its diaphragm and, as the diaphragm in the 4090 and 4091 is tiny, the kinds of medium-distance placements that might be favoured with large- or even small-diaphragm condensers don't really suit it. The increased gain necessitated by miking from a greater distance tends to expose the higher self-noise. However, move the mic in close, as intended, and the increased level of the source quickly overcomes any worries about the mic's self-noise.

I found that the 4090 could be positioned within two inches of an acoustic guitar's strings, to achieve a superb degree of separation between instrument and band spill, and although some experimentation is required to find the optimum position for a properly balanced sound, the result was punchy, dynamic, detailed and very usable. A pair of 4090s as overheads above a drum set, with the 4091 in the kick drum, produced a very natural and crisp jazz-style drum sound too.

Alternatives

There are few direct alternatives to the DPA 4090 and 4091 mics, at least in terms of physical size. Earthworks produce several ranges of very small diaphragm omnidirectional mics that offer similar performance — the SR and TC series, for example — but they are all longer and slightly larger than these DPA offerings. The only other comparable option would be to use a standard miniature 'tie-clip' omni mic, which would be less elegant but should provide similar performance and ease of placement. A wider range of alternatives then emerge from the likes of Sennheiser, Sony, Beyerdynamic, Countryman Associates and, of course, DPA's own 4060-series mics (like the 4061, pictured below), on which the 4090 and 4091 are based.



Conclusions

As a high-quality and very versatile instrument mic, the 4090 is hard to beat, with the 4091 coming into play for the loudest of sound sources. Omnidirectional mics may not be very fashionable (many potential users fear a lack of separation), but with thoughtful application there are many practical and sonic advantages to the use of omnis, and the 4090 and 4091 make it easy to become hooked on this approach.
I definitely recommend trying out these neat little mics for studio and stage applications where quality is paramount. SOS

information

- £346.63 each, including VAT.
- Sound Network +44 (0)20 7665 6463.
- +44 (0)20 7665 6465.
- info@soundnetwork.co.uk
- www.soundnetwork.co.uk
- www.dpamicrophones.com

vienna instruments

VIENNA SYMPHONIC LIBRARY

The Power of 10.

- 10 Vienna Instruments, now shipping!
- 10 of the most powerful sample-based Virtual Instruments on the planet.
- 10 times faster than creating music with conventional samplers.
- 10 easy-to-use and intuitive software instruments with literally zero learning curve.
- 10 user-configurable interfaces for drag-and-drop Orchestra Creation.
- 10 real-time controls to trigger hundreds of articulations on a single MIDI channel.
- 10 "mind-reading" Performance Detection algorithms that switch automatically between articulations.
- 10 online Video Tutorials that show you how easy it is to achieve the dream.
- 10 Collections that make your computer the ultimate Orchestral Workstation.
- 10 years in the making.

VST
AU
XP
OS X

10 reasons to get all 10 Vienna Instruments now, available separately or together in the Symphonic Cube.

TEC Award Nominee 2006

All 10 NOW SHIPPING!

AU (OS X and Mac Intel)
VST (OS X and Win XP)

VIENNA SYMPHONIC LIBRARY
www.vsl.co.at

TIME+SPACE
www.timespace.com

Classic Tracks

The Knack 'My Sharona'

Artist: The Knack
Track: 'My Sharona'
Label: Capitol
Released: 1979
Producer: Mike Chapman
Engineers: David Tickle, Pete Coleman
Studios: MCA Whitney

In 1979, at the end of a 13-way bidding war, the Knack went into the studio to record their debut single — a debut single that would go gold in seven days and sell six million copies worldwide.

Richard Buskin

Today, Sharona Alperin is an upmarket real estate agent in Los Angeles, but in 1978 she was the 16-year-old inspiration behind the hit song that now serves as the intro to her web site — naturally called www.mysharona.com. The song bestowed lasting fame on a Beatles/Brit Invasion-inspired LA band who took their name from the title of a 1965 Swinging London film by Fab Four director Richard Lester: *The Knack... And How To Get It.*

Comprising singer/guitarist Doug Fieger, lead guitarist Berton Averre, Prescott Niles on bass and Bruce Gary on drums, the Knack embarked on their date with destiny when 26-year-old Fieger got the hots for Alperin after being introduced to her by his unwitting girlfriend. Indeed, he and Averre penned the number at the apartment that Fieger and said girlfriend shared.

"Berton had this basic guitar and drum riff lying around for a long time, even before the Knack got together," Fieger would later recall. "He played it for me and I really liked it. I said we would do it someday, but I didn't know how we could use it at the time. Then, at the same time the Knack started, I met a little girl named Sharona, whom I fell in love with. When I would think about Sharona, Berton's riff came to mind. So, Berton and I got together and worked out a structure and a melody and the words."

Yeah, those words, in which Fieger admits to having 'Such a dirty mind / Always get it up for the touch of the younger kind'. Many critics were not amused, but Sharona was evidently wooed, and within a year she was in the studio when the song was committed to tape.

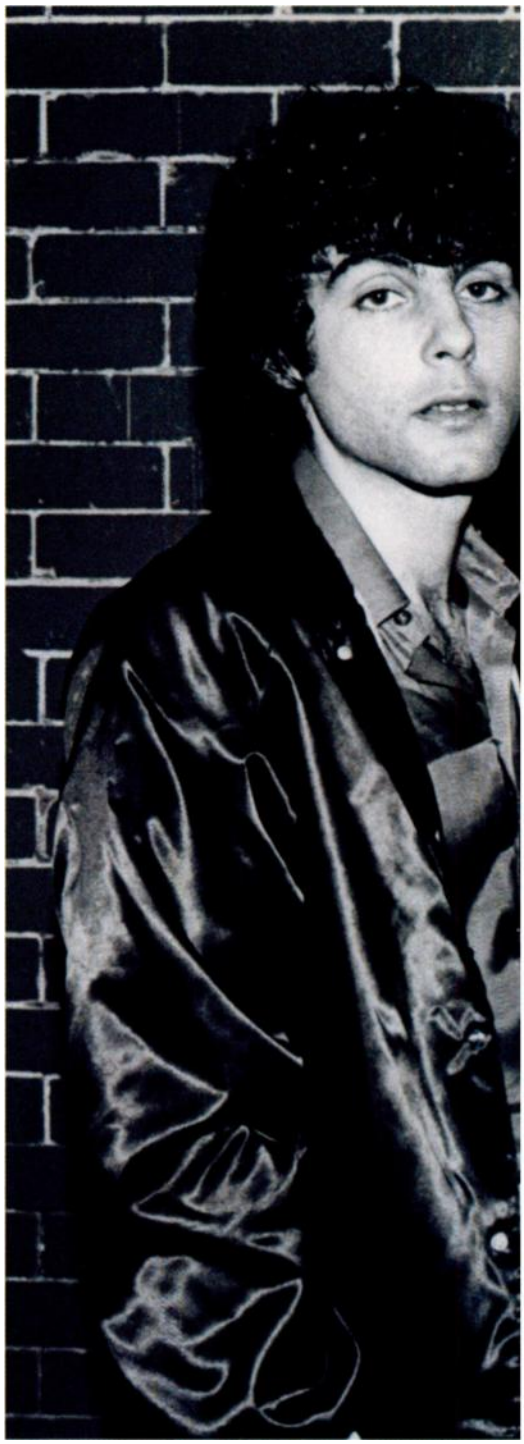
"I thought she looked about 14," remarks David Tickle, who engineered the session that Sharona attended. "She was there when we tracked the backing vocals."

Careering

Tickle has worked with Prince, Blondie, U2, Rod Stewart, Joe Cocker, Joan Armatrading, Toni Childs, Split Enz and 4 Non Blondes during a 30-year career that has taken him from Guildford, Surrey (southwest of London), to the Hawaiian island of Kauai by way of Los Angeles. However, the route that bridged the three-year gap between his first work assignment and that with the Knack was a fairly circuitous one, involving a cruel setback, some lucky breaks and a pretty weird coincidence.

A keyboard player in his school covers band, Tickle was just 16 when he began as a tape-op on May 31st, 1976, at Central London's Zodiac Studios (later acquired by producer Tony Visconti and renamed Good Earth). Hired by an engineer named Mike Cooper, Tickle took over his job behind the MCI console when Cooper departed three months later, and he was soon recording material by the likes of Liverpool Express and the Swinging Blue Jeans. However, within just six weeks of his appointment as a fully fledged engineer, David Tickle was the next employee to part company with Zodiac.

"I met this guy who was out of work, an engineer called Colin Thurston, and I felt kind of bad for him so I got him a job as our studio manager," he recalls. "Then we got this fabulous booking with Tony Visconti and David Bowie — they were working on Bowie's *Low* album — and Colin told me 'You're not engineering any more. You're gonna have to tape-op.' I said 'Why?' and he said 'I'm demoting you.' I couldn't believe it. I'd got the guy a job and now he was demoting me. Well, I lasted about a day. He just snagged the whole project from underneath me, and while he ended up working on that record I left the studio and felt totally despondent and confused. I didn't know what to do, so I went to the studio where Colin had last worked, where he'd been fired, TPA (Tin Pan Alley) on



The Knack in 1979 at the time of 'My Sharona'.

Denmark Street, and as they still hadn't filled his position I ended up getting his old job."

How's that for a spot of opportunism? Intent on taking the fast track to becoming a producer, Tickle immediately set about finding bands in the Guildford area and sneaking them into TPA on weekends and in the wee small hours of the morning to record demos. The problem was that the studio clock was rigged to the power supply, so when Tickle turned this off the clock would stop, and it wasn't long before the owner figured out that surreptitious sessions were regularly taking place until five or six in the morning. The result: Tickle and TPA went their separate ways. Next, he turned down an engineering post at a small facility in Oxford named Acorn that couldn't open until it had finished constructing its own console — instead of operating as a studio the place would eventually become known as SSL.

Short stints followed at Indigo Studios in Manchester and Ringo Starr's home setup at Tittenhurst Park in Ascot, formerly owned by John Lennon. It was there that, among various assignments, Tickle honed his production skills on some demos by a New Zealand band, and thought no more about it, before landing a job as an engineer and trainee producer for the glam-pop songwriting team of Nicky Chinn and Mike Chapman. Nevertheless, when Chapman heard the aforementioned demos, he vowed to sign the band once he and Chinn had launched their new American-based record label, with the intention of assigning the group to Tickle as his first full-scale production.

That would never happen — Chinn and Chapman subsequently changed their minds about the New Zealand band, named Split Enz, whereupon Tickle quit his job. He hired an attorney who, by sheer chance, introduced him to Michael Gudinski of Mushroom Records in Australia. Gudinski duly informed Tickle that Mushroom had released his demo production of Split Enz's 'I See Red' and that it had climbed to number nine on the Australian singles chart. Accordingly, having just left

Chinn and Chapman because they wouldn't sign the band, he was asked to produce Split Enz's next album, and this in turn launched his production career. Things had worked out for the best, and in the meantime, Tickle had engineered several Chapman-produced albums, including Blondie's *Eat To The Beat*, Pat Benatar's *In The Heat Of The Night* and the Knack's *Get The Knack*.

Getting The Knack

It was partly on the strength of his then-recent US chart-topping singles with Blondie, Nick Gilder and Exile that Chapman was chosen by the Knack to produce their debut record. He'd already heard all about the band — after having been turned down by every major label, Fieger, Averre, Gary and Niles had quickly built a following on the LA club scene and caused a buzz that prompted offers from no fewer than 13 labels and a $500,000, two-album deal with Capitol.

The first of these records helped revive power pop in the wake of the punk explosion, melding catchy Beatles-type hooks and Who-tinged vitality with New Wave sensibilities, its dozen tracks being recorded in just one or two takes, with overdubs limited to a few guitar parts and backing vocals. These sessions took place at MCA Whitney in the Glendale suburb of Los Angeles, where the large room used for tracking housed a 36-input Neve console, an MCI 24-track tape machine and Altec Lansing speakers.

"The control room was maybe 20 feet by 18 feet, fairly compact, but the studio was big, capable of taking a decent-sized orchestra," recalls David Tickle, whose home setup in Hawaii has an SSL desk and a Pro Tools rig, along with Westlake monitors.

"The entire back wall — which had a warehouse on the other side — was a pipe organ; one of those organs that had all kinds of trumpets and drums and everything mechanical inside of it. It would actually play the real instruments. The kit was probably about 15 feet from that back wall, and the room's size accounted for the drum sound that you hear on tracks like 'My Sharona'. It was actually so big that the whole back half of the studio was where we created that drum sound, with the front of the kit gobo'd off.

"To the left and right of the drum kit, about 25 feet away on each side and about 12 feet high, were two Neumann KM84 microphones, while there was another KM84 on the snare. However, the big trick was the compression — the Urei 1176s that we used on the room mics — while Mike Chapman's other engineer, Pete Coleman, had the idea of putting one more mic about 10 feet above the drummer's head, right above the snare drum. That microphone, again with plenty of compression, had a lot to do with the drum sound, and so did the room, where the ceiling must have been 30 foot high.

"The night before the album sessions started, Mike Chapman said 'Tomorrow I want the biggest drum sound ever recorded.' That was his MO to get the thing going. Still, it's one thing to use room mics and quite another to have one directly above the snare. To this day I've never seen anyone else do that, but it does work, helping to capture as much as possible of the room sound around the drums by way of keying the compressors off the snare. Every time the snare was hit, the room mics would compress along with it and suck in the sound of the room."

Doug Fieger, Mike Chapman and Berton Averre celebrating Chapman's birthday at MCA Whitney at the end of the mix.

Dreaming Of Drums

Tickle and Coleman tag-teamed as engineers throughout the project, with Tickle taking care of most of the recording and Coleman doing the mix.

"It was all so quick," Tickle remarks. "Seven days were spent recording the album and four days to mix it... After Chapman said he wanted the biggest drum sound ever, I was so nervous I could barely sleep. I kept thinking 'How am I going to do this? This is it!' Anyway, I started dreaming and I thought 'Oh yes, use this microphone over here and the room mics out here to the left and the right,' and then I started to see the EQ on the console and thought 'Huh, let's put these room mics through the 1176s.' There I was, using the compressors in my dream!

"The next morning, when I went to the studio, my hands were all clammy, aware that I was working on a big record and wondering 'Oh shit, what should I do?' Well, I started setting up everything the way I'd seen it in my dream, with the mics out to the side, and I set up the console as well as the compressors, and then Pete Coleman arrived about an hour later and he said 'Oh, good idea, I like what you're doing here. Let's set up this one extra mic above the snare drum.' Well, within a couple of hours we had the sound we wanted. The whole thing was there, with Neumann FET 47s on the kick and tom-toms, and KM84s on the cymbals. It was a great thick-sounding maple Gretsch kit.

"You have to remember, back then people weren't using room mics. Drummers were placed inside shag-carpet booths. I'd actually experimented with the room mics and compression on the Split Enz song 'I See Red', and then, when I worked with Mike and he heard that, he loved it. As a result, I used that same technique for Blondie's *Eat To The Beat* at the Power Station. However, the Knack album was particularly special because of the keying of the compressors with the snare drum. That's what kind of brought it all together."

The Beatles & NOFX

The rest of the band was recorded in the same large room as the drums, with guitarists Berton Averre and Doug Fieger standing next to each other to the right of Bruce Gary, and about 30 to 40 feet in front of him. True to their Beatles influence, they were equipped with Vox amps, each miked with a Neumann U67 about a foot away and a Shure SM57 up close.

"The Beatles' spirit was invoked in terms of their fondness for the guitar sounds, with the Vox amps and so on, but not so much with the singing and drumming," Tickle comments. "I preferred the U67 over the 87 because of its tubey quality and the bite that you get. At the same time, the 57 gives you a more direct punch out of the guitar, and since it was going to be a record with very few overdubs we wanted to make sure that each sound really stood out in a three-dimensional way. By using a couple of different characteristic microphones we were able to get that to pump out."

Prescott Niles, standing in front of the control room window, about 60 feet away from the drummer, had a little booth constructed around him to contain his bass guitar's low end, while a tube 47 was employed on his Ampeg BA115 bass amp combo.

"We also had a mic set up for Doug's vocals, although these were done separately because of the leakage," Tickle states. "Again, we used a 47."

While the 'My Sharona' guitar solo was layered in two parts, Doug Fieger insisted on

record, edit, mix, master.

PRO TOOLS HD 7

improved host processing efficiency • new instrument tracks • support for up to 160 simultaneous channels of I/O • support for RTAS plug-ins on aux inputs and master faders • automatic delay compensation

DIGIDESIGN ICON INTEGRATED CONSOLE

fully modular architecture • expandable to 80 physical faders • comprehensive monitor control section • Includes XMON remote, rack-mounted analogue I/O audio monitoring and communications system • support for up to 7.1 surround mixing

ON PERMANENT DEMONSTRATION

FREE Crane Song Phoenix TDM plug-in with all Pro Tools HD 7 systems purchased. (Time limited offer)

digidesign

KMR exist solely to provide recording tools and recording solutions to those people who care about their sound. Backed up by an extensively stocked showroom, professional recording engineers on hand to advise and support, excellent service and years of experience to create the studio ideally suited to your requirements, KMR is the UK's finest, friendliest and most knowledgeable studio equipment supplier. With an allegiance to quality over anything else, we are here to answer any recording related question you may have.

Please email, phone or drop by for a coffee and a chat about your studio.

And in the KMR showroom over 100 studio grade recording tools on demonstration from manufacturers including :

AEA, AKA, Alesis, Apogee, Apple, Audient, Avalon, Bryston, DPA, Drawmer, Earthworks, Eventide, FMR, GML, KRK, Lexicon, Little Labs, M-Audio, Mackie, Manley, MOTU, Native Instruments, Neumann, Oktava, Oram, Roland, Rupert Neve Designs, Sennheiser, Shure, Spectrasonics, Soundfield, Steinberg, TC Electronic, TL Audio, Universal Audio, Van Damme, Waves and Yamaha.

API 2500 Stereo Compressor £ 1,800.00

API 7600 Channel £ 1,800.00

Crane Song STC 8 £ 2,550.00

Crane Song TRAKKER £ 1,460.00

API 500V 10 slot rack • 512C • 525 • 550B •560 • 550A •

The sound of the classic SSL E Series console in a rack. The E-Signature Channel offers two distinctive Eq options coupled with the most musical dynamics processor available

SSL XLogic E Signature Channel £ 2,100.00

SSL XLogic G Series Compressor £ 1,900.00

SSL XLogic K Series Compressor £ 2,000.00

Coleman Audio TB4MKII £ 740.00

Coleman Audio M3PHMKII £ 640.00

Roll Music Folcrom £ 500.00

Neve 8816 £ 1,850.00

Neve 1073 DPA £ 1,599.00

Neve 1081 Stereo £ 5,797.00

Gefell M92.1 £ 1,200.00

Gefell M930 £ 636.00

Gefell UM900 £ 2,464.00

UM900 Naked Eye Studio Vocalist M930 e22S

Josephson e22S £ 940.00

Josephson C42 £ 320.00

Josephson C42MP £ 685.00

Brauner Phantom AE £ 595.00

Brauner VMA £ 3,999.00

Crowley & Tripp Studio Vocalist £ 1,000.00

Crowley & Tripp Soundstage £ 1,000.00

Crowley & Tripp Naked Eye £ 490.00

The new SSL AWS 900+ builds on the successful original formula by adding tighter integration with more popular DAW platforms, enhancements to the displays and meters, and a sleek 'platinum' styling.

Using the new control protocols, plug-ins and virtual instruments can now be controlled from the D-Pots and the faders, a hotly requested feature from many workstation users. An enhanced display allows up to 6 character labels for track/channel names and the console also now features workstation Master Fader support.

Complete with full automation, TOTAL Recall, and TOTAL DAW control the AWS 900+ is the ideal mixing and tracking partner for any computer based studio. Includes the famous G Series mix buss compressor.

ON PERMANENT DEMONSTRATION

AWS 900+

Solid State Logic

K&H O300D Active 3-way nearfield monitor of unique power and clarity. Very revealing with great stereo imaging and a solid, tight, bottom end.

K&H O300D £ 2,629.00

The largest collection of serious recording tools in the UK

KMR AUDIO

www.kmraudio.com

020 8445 2446 • sales@kmraudio.com

1375 Whetstone High Rd, London, N20 9LN

0% 6 months deferred credit available • HIRE SERVICE AVAILABLE

Dynaudio AIR 6 Developed for both stereo and multichannel monitoring in any of the most demanding environments. With both analogue and digital inputs and hardware and software remotes, setup is both easy and precise.

Dynaudio AIR 6 £ 2,000.00

Gelthain RL901K Self powered high headroom main/midfield monitors. Come and listen to tight deep bass, incredible stereo imaging, zero distortion mids and beautiful, unagressive, clear top end.

Gelthain RL901K £ 5,998.00

PMC TB2A Truly accurate nearfield monitors. Great stereo imaging, low distortion, high headroom amps lead to an engaging natural non-fatiguing sound quality that is the precursor of a good mix with modern high definition music.

PMC TB2SA £ 1,450.00

Software, Synths, Sequencers, Samplers & Sounds (Inc. VAT)

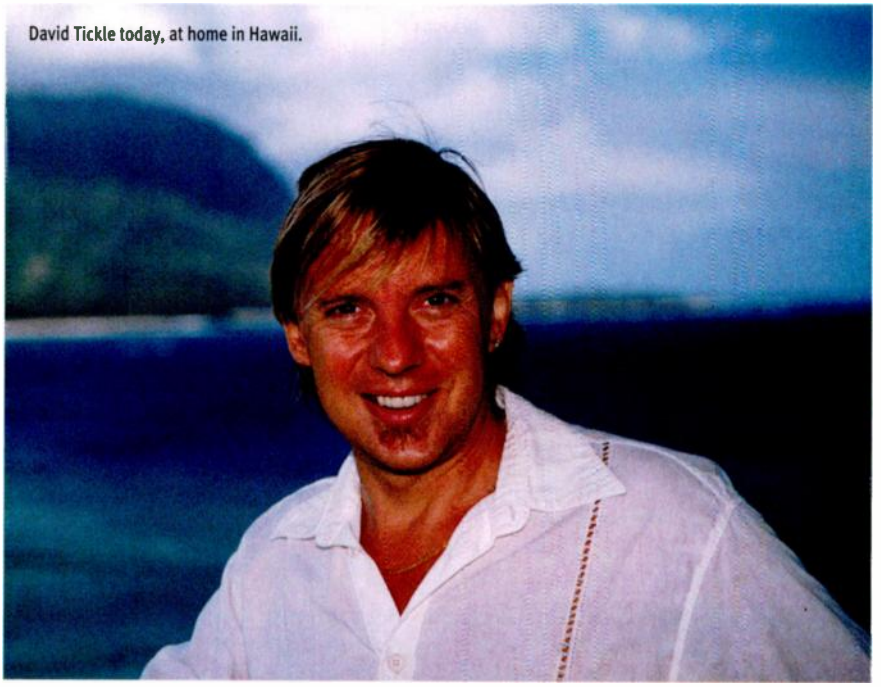
Steinberg Cubase SX 3	£ 499.00	Apple Logic Pro 7.2	£ 599.00
Steinberg Cubase SL 3	£ 199.00	McDSP Project Studio	£ 351.00
NI Absynth 3	£ 165.00	SoundToys Native Effects	£ 363.00
NI Reaktor 5	£ 269.00	Audioease Altiverb V5 Native	£ 381.00
NI Kore	£ 369.00	Spectrasonics Stylus RMX	£ 132.00

All Prices Excluding VAT except where stated • Errors and omissions excepted

recording his vocal completely dry.

"There were no effects on anything," Tickle confirms. "That's what they wanted. To get the voice thicker we would double Doug's vocal, and then when it came to the mix the latest technology available was the EMT 250 reverb. It looked like R2D2, like a little robot, and it was extremely expensive – back then it cost around $32,000 — but Chapman bought one of these things and we used it on the snare drum and the vocals. Basically, it was a very, very short reverb — you could barely hear it, but it was just enough to lift the voice and the snare.

"The guys made a point about not using any effects in the mix. To a certain degree they were in control of their own destiny, and while Chapman really put the sound together he would have done more if it hadn't been for Fieger's input. That kind of tension between Chapman and Fieger was a healthy tension, and it helped cause the sound to turn out the way it did."


David Tickle today, at home in Hawaii.

Party Mix

It was after five days of recording all the basic tracks that the sessions switched to an adjacent Neve-equipped room at MCA Whitney for the mix, as well as for the preceding addition of backing vocals. And this is where David Tickle did a star turn... well, kind of.

"When we were finishing 'My Sharona', they wanted to have a group of people sing it, so Chapman said 'Get in there, David, with the band,'" he explains. "That's when Sharona herself was there. Anyway, we did the background vocals, and what you can't quite hear is that they go 'Fuck-a-me, fuck-a-me too much, my Sharona'. That's actually what was being sung by all of us grouped around one mic. It wasn't super-loud, but it definitely lifted the song."

And clearly, what with all of the anticipation surrounding the band, the overall sense of optimism was lifting everyone's spirits.

"I was arriving at nine, 10 in the morning and leaving at midnight, and even the mix was like a party. We were mixing two or three songs per day. Pete Coleman was manning the board and I was just setting the compressors or doing whatever was needed. Four days was all it took, and that was really the band's call — 'They used to make records this way. Why can't we do it now?'"

Well, they did; not in the 17 hours that it took the Beatles to record their first album, but in 11 days and for a total outlay of less than $18,000. Certified gold within 13 days of its release on June 11th, 1979, *Get the Knack* went platinum on August 3rd and eventually sold over five million copies. Meanwhile, 'My Sharona' went gold within eight weeks of its June 18th release and topped the Billboard chart for six weeks through August and September. 'Good Girls Don't' subsequently peaked at number 11.

Knack-lash

However, the much-hyped 'Big Knack Attack' headed downhill in the wake of resentment over the band's Fab Fourish references and refusal to do interviews, as well as the perceived sleaziness of their undeniably catchy songs.

"The ugly sexism of these corrupt creeps is an affront to women," carped one critic, while San Francisco artist Hugh Brown invented a 'Knuke the Knack Kit', complete with a "Just When You Thought it Was Safe to Listen to the Radio" Jaws-style T-shirt and "Honk If You've Slept with Sharona" bumper sticker.

Not that Sharona Alperin has any complaints. Indeed, according to the statement on her web site, the song that bears her name and craves her body has been of immeasurable assistance in her daily quest to help celebs find their dream homes over the past 14 years.

"'My Sharona' has had an impact on my ability to understand the entertainer's mind," she asserts. "There's something simpatico. You've got to care to the nth degree... I sell the most emotional product on the market, because a star's home is their only safe haven."

To quote Doug Fieger, 'My, my, my, aye-aye, whoa!' SOS

Getting Sharona In Shape

The engineer now freely admits that when he first heard 'My Sharona', during week-long pre-production sessions at SIR in Hollywood, it wasn't his favourite among Knack songs that included 'Lucinda' and 'Oh Tara'.

"Chapman was all over 'My Sharona'," says Tickle, "He thought it was great. However, the structure of the cassette demo wasn't very good, and that's what Mike specialised in: coming up with the right parts where there weren't any, working out the solos. He taught me a lot about songwriting. He did most of it and I observed the process as he really knocked the songs into shape, determining where the solo came and the impact of the verse-chorus-bridge. He's a very, very good songsmith, and he basically worked out most of those structures so that they had some emotional impact.

"In the case of 'My Sharona', the demo consisted of the same drum pattern and the same riff as on the finished record, but it meandered more. And as far as I remember, in its original form it didn't have the guitar solo, or certainly not the one that's on the actual song, where it's in two layers. The original may have only had the one layer. I know that Mike worked on that quite a bit. The fact is, you would have recognised the song, but it just wasn't as tight and as cleaned-up structurally as Mike then made it.

"Most of the songs were captured first take, maybe second take once we got all of the tones together, and then it was like 'Oh, we should put an overdub on this.' We'd do that, record the vocal and then move on. That's how it was with 'My Sharona'. Everything had been worked out in pre-production and it was polished, so what you hear is the real thing, a complete take. Practically all the songs were done that way — there may have been a little bit of editing on the album, but they were getting everything down within a couple of takes, and that was due to the band's proficiency and the straightforward structures of the songs. The band was very, very tight. That was a part of their trademark."

got keys but no control?

ReMOTE ZeRO SL

Automap™ for the keyed up generation

Want to get a "jump on the future" of MIDI controlling but don't need a keyboard?

Novation's first dedicated control surface is the product you need, featuring revolutionary Automap™ technology for instant control of your software. The ZeRO SL fits neatly behind a qwerty keyboard, saving precious space and allowing quick and easy access. With huge LCD displays running across the top, parameters are displayed instantly as the SL automatically maps to popular sequencers and plug-ins, putting an end to MIDI headaches!

The SL also comes bundled with the fantastic Xcite software and samples pack; the perfect accompaniment to the controller, making a complete package that offers something for everyone, from complete beginners to the consummate pro.

automap™ REVOLUTION

SUPPORTED APPLICATIONS -

NUENDO

PC COMPATIBLE

Xcite Pack Software & Samples

live lite5 BFD SAMPLES CD GURU BASS STATION

INNOVATION | INCONTROL

t: +44(0)1494 462246 e: sales@novationmusic.com

novation

www.novationmusic.com

Way Out Ware TimewARP 2600

Software Synthesizer For Mac & PC

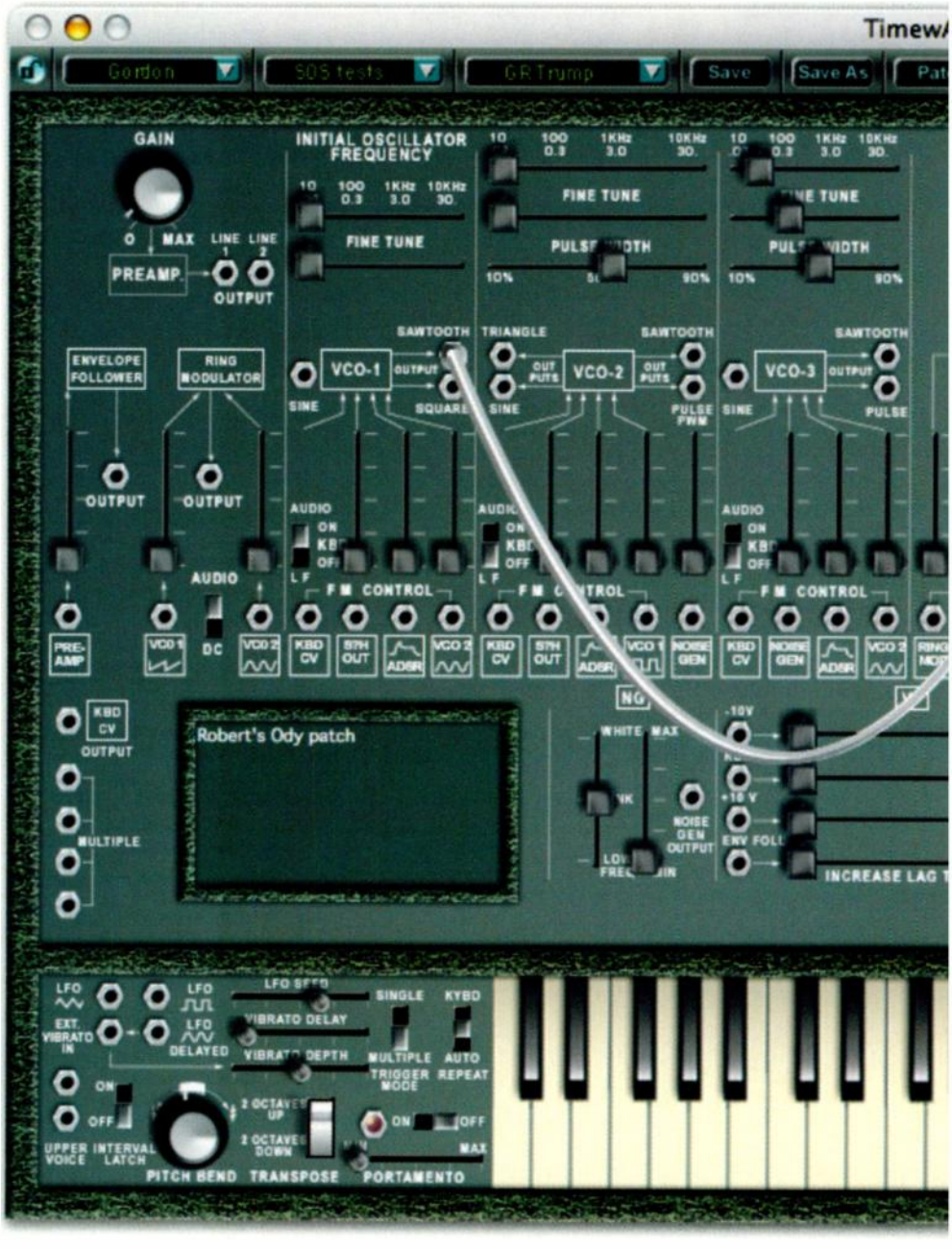
Gordon Reid

Can this second in-depth software emulation of the ARP 2600 improve on Arturia's existing *2600V*?

This new instrument from Californian company Way Out Ware joins Arturia's *2600V* in emulating ARP's 2600 semi-modular synthesizer. Installing and authorising the software is straightforward if you do it on a machine connected to the Internet, but WOW provide a couple of workarounds if you prefer to keep your studio computer solely for music use.

Oscillators & Noise Generator

As you would expect, *TimewARP*'s oscillator section is modelled on that of the original ARP



Sound On Sound

Way Out Ware *TimewARP 2600*
£170

pros

- Like its inspiration, it's a very flexible synthesizer.
- The design and implementation is a purist's delight.
- Its snappy envelopes make it useful for aggressive and percussive sounds.

cons

- It's extremely CPU hungry.
- Neither the oscillators nor the filter emulate the ARP 2600 as closely as WOW would have you believe.
- With a few honourable exceptions, the factory sounds do not impress.
- It lacks the competition's enhancements.
- There is no paper manual, only PDF documentation.

summary

With a sound that is similar, but not identical, to that of the ARP 2600, *TimewARP 2600* is faithful to the philosophy of the original, adding little in terms of extras. You might feel that this approach lacks imagination, or that it's superior because it remains truer to ARP's vision. Either viewpoint is justifiable. However, its hunger for CPU power will not win it many friends, and I fear that it has arrived too late to make the impact that it might once have done.

2600, the only differences being that VCO 1 and VCO 3 each have a sine-wave output. (This is in sharp contrast with Arturia's *2600V*, which offers additional octave selectors, oscillator sync, and scaling controls for the keyboard CV inputs.) So how true to the original do *TimewARP*'s oscillators sound? Lining up my own ARP 2600 and 3620P keyboard alongside my Mac running *TimewARP 2600*, I started by comparing the the sawtooth waves, and was encouraged to find that the two instruments sounded very similar. This is surprising, because the waveforms show marked visual differences on an oscilloscope. Indeed, WOW seem unable to tell the difference between a sawtooth wave and a ramp wave. Immaterial at audio frequencies, this makes a huge difference when using the oscillator as an LFO. However, don't be too harsh on the developers — I only recently discovered that the ARP 2600's oscillators produce sawtooth waves at audio frequencies, but a misshapen ramp (that changes markedly when you start patching it to different destinations) in LF mode.

The sine waves appear more similar visually, but listening to them reveals that the ARP 2600 is bright, whereas *TimewARP* sounds like the 'pure' tone that a sine wave should be. Repeating these experiments with the triangle wave, I found that *TimewARP*'s is again more 'true' than the ARP 2600's, and I obtained equivalent results for the square and pulse waves. This implies that WOW have modelled their waveforms on ideal waveshapes, not those derived from real ARP 2600s with real components and less than perfect calibration.

Test Spec

- Way Out Ware *TimewARP 2600* v1.10
- 1GHz G4 Titanium Powerbook with 1GB RAM running Mac OS 10.3.9 and 10.4.3, *Plogue Bidule* 0.8501 and 0.9, and Korg Legacy MS20 keyboard controller.
- ARP 2600 synthesizer, serial number 1618.

However, there's an even more significant way in which *TimewARP 2600* differs from the real thing. When tuned to the same pitches, the phases of all three oscillators are always the same! Without phase-locking (as occurs by accident in some vintage synths) this would only happen by chance on an analogue synthesizer, even if you could tune the oscillators perfectly. By locking the oscillators' phases and eliminating any movement, WOW have significantly affected the nature of any two- and three-oscillator sounds, eliminating the warm, organic movement that occurs on a true multi-oscillator vintage instrument.

Filter Section

The *TimewARP 2600* filter section sticks to the configuration of the ARP 2600, with the exception of an added KBD CV Amount control. (On the ARP 2600 itself, this is always 100 percent). Testing the ranges of the two instruments' filter cutoff frequencies, I found that the ARP's filter ranges from 12Hz at the low end to way, way off the scale at the upper end. (When last I saw it, it was heading up toward 30kHz and still going strong.) In contrast, *TimewARP*'s filter ranges from 10Hz at the bottom to approximately 10.2kHz at the top. Despite WOW's claims to the contrary, I was unable to make it increase beyond this, no matter how many positive voltages I applied to its CV inputs. This means that at maximum cutoff frequency with no resonance the soft synth audibly attenuates high frequencies to a far greater extent than the hardware. I suppose that you could hypothesise that WOW modelled *TimewARP 2600* on a later version of the ARP 2600, one that used the ARP4072 filter. Due to a miscalculation, the 4072 had a bandwidth of about 12kHz, so ARP 2600s from 1976 onward had a duller sound than earlier units.

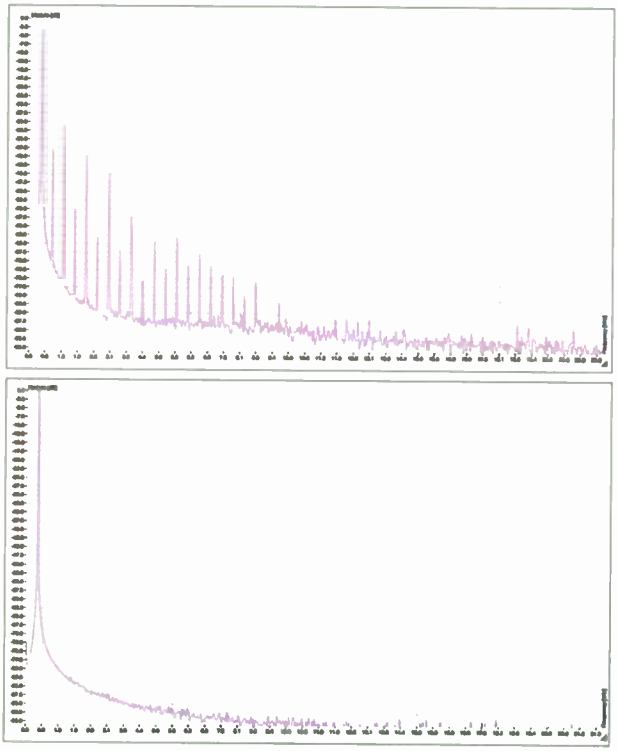
It was while making these tests that I noticed that the cutoff control's Tool Tip shows the wrong frequency values. When I set up the filter resonance so that the filter was self-oscillating and the Tool Tip read 440Hz, the actual output measured at 621Hz. That's six semitones too high! To obtain a true 440Hz output, I had to set the filter cutoff frequency to 309Hz.

Transient Generators, VCA & Output

TimewARP's transient generators, VCA, output mixer, and reverberator follow the design of ▶

the ARP 2600 precisely, with no additional frills. However, WOW have taken the opportunity to extend the envelope times of *TimewARP 2600*. I measured the longest Attack as 56 seconds, but more important, of course, are the minimum Attack, Decay, and Release times. With all the ADSR controls set to minimum, the *TimewARP*'s transient was quicker and better defined, which resulted in more precise clicks and snappier transients.

To the furthest right of the control panel you'll find the output Mixer and the Reverberator. Again, I can see no difference between WOW's implementation and the original synth. However, the sound of the reverb is nothing like that of a spring. Don't get me wrong... it's a usable lo-fi reverb, but the definitive 'boyoyoyoyoinnggg' of the ARP is completely lost.



The *TimewARP 2600*'s sine-wave spectrum (bottom) and the ARP 2600's sine-wave spectrum (top) for a 440Hz tone. While the emulation is almost mathematically pure, the hardware synth is much brighter.

Modular Toys

We've now come to the end of the ARP 2600's conventional signal path but, of course, this is only part of the story. Underneath these modules lies a set of Voltage Processors and a powerful Sample & Hold and Electronic Switch section, all of which are recreated accurately on *TimewARP*.

To the far left of the panel lie the Ring Modulator and the external signal input modules — a Preamplifier and an Envelope Follower. I rather like the Ring Modulator on *TimewARP*, as it offers both AC- and DC-coupled operation, and sounds much like the device on the ARP 2600. The external signal input is, however, an improvement on the original. When you launch *TimewARP 2600* in a suitable application, you can choose from none, one, or two signal inputs. If you choose the last of these options, the two channels of audio presented to *TimewARP* appear independently on the Preamplifier's Line 1 and Line 2 outputs, ready for directing anywhere within the synth. If you like to use synthesizers as signal processors, you'll like this a lot.

Keyboard & LFO

If we ignore the Blue Meanies and Grey Meanies, the earliest ARP 2600s were supplied with the ARP 3604P keyboard, which provided nothing more than pitch, trigger, and gate information. ARP later replaced this with the ARP 3620, a sophisticated duophonic unit that provided single- and multi-triggering plus a flexible triple-output LFO with delay. As you would hope, *TimewARP*'s keyboard section emulates the ARP 3620, although it does so with 60 keys rather than 49. The layout is totally different from that of the original, but the facilities appear to be the same, with the unavoidable absences of the footpedal inputs for portamento on/off and the interval latch. This means that the ARP 2600's unusual duophonic mode is retained, allowing you to play two pitches (although only one underlying timbre) using a single voice.

TimewARP's keyboard LFO is, in one way, a significant improvement on the ARP 3620's, because its maximum frequency is 100Hz rather than 20Hz. However, as found elsewhere in *TimewARP 2600*, the waveforms are a bit too close to the ideals and, unlike the 3620, *TimewARP*'s LFO sine wave is gated by the keyboard — on the original, it is free running and ever-present at its output, which is much more flexible.

New Friends

In common with other soft synths, *TimewARP* allows you to control parameter values (but not patching) over MIDI using velocity, aftertouch, the mod wheel, and one assignable Continuous Controller. You can determine the minimum and maximum ranges of each controller, select one of three response curves for each, and invert the effect if you wish. While this system does everything that you might reasonably wish, only three-quarters of its settings are saved in the patch — the Continuous Controller is part of a global map that you must save and load independently. That's daft.

Click on the *TimewARP 2600* logo to the bottom right of the panel, and a small menu appears. This allows you to save and load the global MIDI maps that you create, as well as providing access to the micro-tuning options and the MIDI Clock synchronisation window. The micro-tuning has some interesting scales, some of which I have not seen before. Unfortunately, you cannot create new scales of your own.

At the top of *TimewARP 2600* you'll find the inevitable menu bar, and this gives you access to the final set of facilities. Most of these are concerned with the saving and loading of patches. The Patch Manager is a hierarchical (Group-Category-Patch) system for importing, exporting, organising and, if desired, reordering your sounds. Unfortunately, it does not lend itself easily to selection with MIDI Program Change commands, because only the first 128 patches in any Group are available. This means that you have to limit the number of patches in upper Categories if you want to be able to 'reach' patches lower down the list. Furthermore, despite allowing you to move around the lists using the cursor keys, the highlighting does not move with you, which instantly leads to confusion. Alongside the patch management controls, there's the polyphony selector (one to eight voices), a reset button that re-patches *TimewARP 2600* to a basic default and silences the thing, a MIDI indicator, output level meters, and a CPU usage meter.

Absent Friends

Unlike the Arturia *2600V*, *TimewARP* does not dispense with the ARP 2600's (in)famous VCA thump, which occurs when you play notes with the Attack and Release set to their

Documentation

When you open the cavernous box, you find that Way Out Ware provide just a CD and a minuscule *Quick Start Guide* that tells you how to load and authorise the software. In other words, there's no manual. Inserting the CD demonstrates that the instructions — written by Jim Michmerhuizen, the man who wrote the original ARP 2600 manual — are supplied as a pair of PDFs. Bizarrely, the first of these provides chapters one to four, and six. The second provides chapter five (a tutorial that encourages owners to do more that just select factory patches) as a separate document. Call me old-fashioned, but I think that this is naff.

NORD STAGE 88 / 76 / COMPACT

THE PROUD WINNER OF THE BEST STAGE PIANO 2006 M.I.P.A

Featuring the best of Clavia's award-winning technologies, the Nord Stage hosts three different instrument sections - Organ, Piano and Synth - plus an extensive effects section. You can use two sounds of each Section simultaneously - that's two Organ manuals, two Synths, and two Pianos. These can be combined in any split or layered configuration you need for your performance.
The Stage's user interface is designed from a basic concept: All vital functions are represented on the front panel - easy to access, and never further away than a press of a single button. After more than 20 years of experience in designing instruments to be used by professional musicians, we know how important it is to make the technology completely transparent for the user. With the Nord Stage you only need to focus on your performance, which means never having to struggle with the technology.

Sound on Sound - February 2006
"THE STAGE IS ABSOLUTELY ON THE MONEY"

NORD MODULAR G2X

G2 ENGINE NOW ONLY £595 INC VAT

Clavia proudly presents the second generation of the Nord Modular - the Nord Modular G2. Nord Modular G2 offers you the better of two worlds and provides an effective solution for the working musician. With the G2, Clavia introduces a truly innovative hardware platform with focus on live performance features. The G2 offers fully assignable controllers to tam, bend and tweak. In fact, the ultra-flexible user interface of the G2 takes real - time editing to a higher level. While remaining true to the traditional modular synthesis concept it manages to go where no modular synthesizer has ever gone before...

NORD ELECTRO 2

VOTED BEST ORGAN - M.I.P.A 2003/04

Clavia's goal when developing the Nord Electro 2 was to create the best emulations of traditional electromechanical keyboard instruments on the market, and to make the instrument compact and lightweight. Nothing else. No brass banks, no orchestral sounds. No compromises. Just outstanding electromechanical keyboard sounds with true feel, from fast keyboard response to authentic sound. The organ in the Electro 2 is based on a digital simulation of the mechanical tone wheels of the B-3 organ. It offers a number of very innovative solutions to mimic the typical B-3 sound.

NORD LEAD 3

PERFORMANCE SYNTHESIZER

The Nord Lead 3 Clavia takes virtual Analog to the next level and introduces the Advanced Subtractive Performance Synthesizer. 'Advanced Subtractive' because of the extremely flexible sound engine, featuring a combination of traditional shapable wavetables and a very intuitive 2 or 4 operator FM engine. All this can be routed through the top quality multi filter for further shaping. 'Performance Synthesizer' because of its unsurpassed user interface and play control functions such as keyboard hold, parallel polyphonic glide in legato mode, single-key triggering of whole chords and even clusters of slightly detuned voices.

Roth
AUDIOBLOB 2

"SOUNDS LARGER THEN IT IS IN REAL LIFE"
WHAT HI-FI 2006

2.1 MULTIMEDIA AUDIO SYSTEM

MP3, CD, DVD, COMPUTER AND PRO AUDIO RECORDING

Roth. Not a name you may be familiar with. It is certainly a name you will remember, especially when you hear the sound that comes from our first audio system. The Audioblob 2 has been developed by a highly regarded team of British sound engineers. It has been assembled in a state of the art production facility. We realised that it was no good just bringing another speaker system into the burgeoning speaker market. If the product is to succeed, it has to be good, it has to deliver a sound that's true, and last but not least, it has to be affordable. We are very proud of this product, it delivers everything we set out to achieve. We hope you enjoy the Audioblob 2.

"THE ROTH IS ONE OF THE BEST-LOOKING SYSTEMS IN THIS TEST, AND FEELS REASSURINGLY SOLID."
WHAT HI-FI 2006

*ALSO AVAILABLE IN WHITE

HOSA TECHNOLOGY

> FULL RANGE AT HANDINHAND.UK.NET

For 20 Years, Hosa has provided a cost effective solution to the cabling needs of musicians, engineers, and audio and video enthusiasts the world over. In an effort to further meet those needs, Hosa continues to add new products that make the creative individual's life easier. Indeed, much of what we offer is the direct result of suggestions made by customers and retailers.

HIGH QUALITY - OUTSTANDING VALUE

ZAOLLA SILVERLINE

FULL RANGE AT HANDINHAND.UK.NET

Thought about your cables lately, no? well maybe you should. You spend thousands of pounds on state of the art, hi tech equipment, and yet like thousands of musicians, listen to it through cheap cables (copper). It's like owning a top of the range high performance sports car and putting two star petrol in it. You wouldn't do it would you. And yet, when it comes to getting the best out of your, hi tech music kit you are doing exactly that.

"YOU WANT THE BEST? ALL THINGS CONSIDERED THIS ISN'T FAR OFF"
Guitarist - March 2006

minimum values. Likewise, the Initial Gain at the top of the VCA section is retained. I was also pleased to see that I could use *TimewARP*'s Voltage Processor sliders as CV generators (which was a trick missed by Arturia). Everywhere you look, it's apparent that *TimewARP 2600* is more faithful to the ARP 2600's 'look and feel' than is the Arturia *2600V*. But this also means that the Arturia synth offers far more in the way of goodies and extras not found on the original.

These 'extras' are non-trivial. I've already mentioned those found in the *2600V*'s oscillators, but *TimewARP 2600* also loses the Arturia's four additional filter profiles (12dB/octave low-pass, high-pass, and band-pass, plus notch), the dedicated chorus/delay effects section, and powerful extensions to the Voice Processor architecture. Likewise, the Tracking Generator (a programmable modulation generator that allows you to create unusual modulators and direct them anywhere you choose within the *2600V*) is gone. Then there's the loss of the *2600V*'s 10 additional patching VCAs.

But much as you may (or may not) miss these, the absence that will probably cause WOW the greatest headache in terms of competing for your cash is that of the ARP 1601 sequencer. Not just confined to producing Giorgio Moroder-esque bass lines of the 'I Feel Love' variety, this offers all manner of polyrhythmic effects and complex sound-shaping possibilities. Now that the world has become accustomed to it being there, I can't imagine many people being blasé about losing it.

Let's Do The *TimewARP*...

To test *TimewARP 2600* in a musical context, I ran it as a stand-alone synth on my G4 Powerbook and as an AU plug-in within *Plogue Bidule*. Having launched the program, I set up the obvious stuff like MIDI channel and polyphony (all easily done), and immediately discovered that the CPU meter is an ever-present worry. This is because *TimewARP 2600* — especially in its stand-alone incarnation — has a humungous appetite for CPU power. On a good day, I might suggest that this is due to a careful implementation of every nuance of analogue circuitry, resulting in extremely power-hungry algorithms. Alternatively, I might suggest that it smacks of badly optimised code that chews power unnecessarily. Either way, the result is the same. If I wanted absolutely reliable and glitch-free results on a 1GHz computer, I had to set the polyphony to... one!

"The sawtooth waves of the two instruments sounded very similar to one another. This is surprising, because the waveforms show marked visual differences on an oscilloscope."

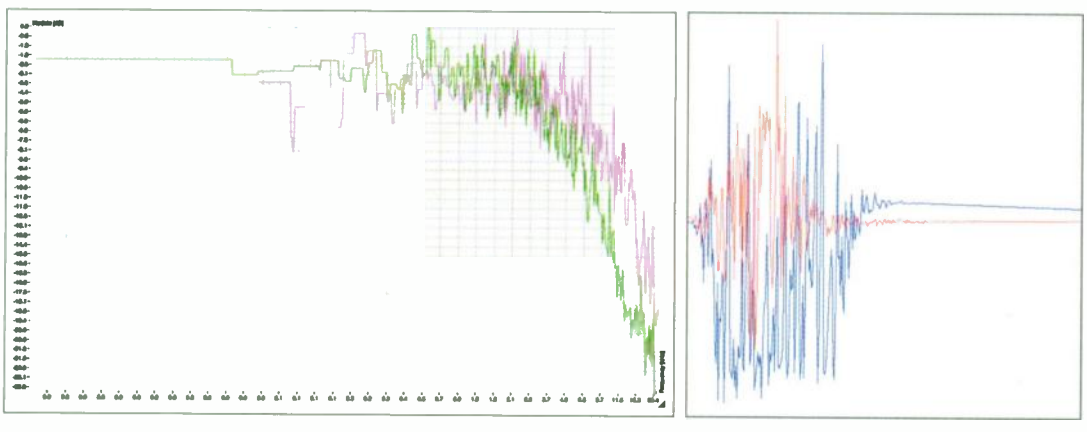
To be fair to WOW, my Powerbook is close to the minimum specification recommended for *TimewARP 2600*, but I was still shocked. With simple patches, I tried to get as many as three notes, but even then I couldn't use long release times, because the software generated digital hash as it tried to arbitrate between the tails of previous notes and the onsets of new ones. There was also the digital equivalent of mistriggering, as the software tried to decide which new notes to play! Strangely, the program proved to be less greedy when running within *Bidule*, and I could run multiple soft synths with less glitching than generated by *TimewARP* alone in stand-alone mode.

When I set up the same simple patches on both hardware and software synths and listened to the results, there was little similarity. I had the original ARP 2600 patch book to hand as I wrote this, so I tried programming its patches on *TimewARP 2600*, and these did not result in the correct sounds. Nevertheless, as I tweaked patches — tailoring the envelopes by ear and adjusting the filter settings to match the response of the real ARP — the timbres started to move closer together, as did the articulation of the notes. Before long, I found that I could make the ARP 2600 and *TimewARP 2600* sound quite similar, even though *TimewARP* lacked the brightness and sparkle of any sounds that were supposed to be extra bright and sparkly.

As I started to programme complex patches, the differences between the original and the copy became more obvious, so I next attempted to increase the polyphony — such as I could — and to create some interesting polyphonic patches. I found that *TimewARP 2600* is rather good at percussive sounds such as electric pianos, which is to be expected given the snappiness of its envelopes. However, I failed to obtain the warmer ensemble sounds that I had hoped. The factory sounds bear this out, with a heavy concentration of leads, basses, and effects, but few polyphonic/ensemble offerings.

It was around this time that I really started to miss the additions provided in Arturia's *2600V*. Processing external signals is less fun when you don't have a MIDI-sync'ed sequencer to manipulate complex parameter changes in synchronisation with incoming sounds, and the loss of oscillator sync and the additional filter profiles (while irrelevant in any comparison with the original 2600) is a shame.

I also found *TimewARP*'s user interface not always to be as slick as the *2600V*'s. For example, there is no equivalent of Arturia's Move Away Cables facility, which causes patch cables to move out of the way of the mouse pointer so that you can adjust parameters that would otherwise be hidden by electronic spaghetti. Neither are there the useful colour-coded patch cables and patching menus. But I'm also worried about WOW's quality control. Has anybody else noticed that LFO Speed is written as LFO Seed on the keyboard section? This makes me worry about other oversights.



The graph on the left demonstrates the differences between the low-pass filter responses of the ARP 2600 (pink trace) and *TimewARP 2600* (green trace) using the same noise signal— the modelled processing audibly attenuates the high frequencies when compared with the original hardware. The amplifiers also respond differently, with the *TimewARP 2600* (red trace) giving faster envelope times, and therefore better defined transients.

Evolution

When *TimewARP 2600* first appeared, it was only as an RTAS plug-in for *Pro Tools*. This frustrated many would-be buyers, and ensured that the Arturia *2600V* stole a march in the marketplace. In many ways, RTAS was a bizarre choice. Sure, *Pro Tools* dominates some areas of the audio industry, but it would have seemed more sensible to release a VST version first or, at a pinch, a stand-alone version. Happily, all of these different versions are now available, although they have appeared nearly a year later than expected. The minimum specifications for the original release and for the latest version are as follows. Note that HTDM and DXi versions are not provided.

• RTAS-only version:
PC: 1GHz processor with 256MB RAM running Windows 98SE/2000/XP.
Mac: 800MHz G4 with 256MB RAM running Mac OS 10.2 or higher.

• AU/RTAS/VST/stand-alone version:
PC: 1.5GHz processor with 256MB RAM running Windows XP.
Mac: 1GHz G4 running Mac OS 10.3 or higher (RAM not specified).

Happily, the operation of *TimewARP 2600* was pretty robust, with just a couple of freezes during the review period. These tended to happen when I was doing something innocuous, such as selecting patches using the Patch Manager. In contrast, over-stressing the CPU usually just caused ghastly noises.

Conclusions

Way Out Ware were going to have to work hard to live up to their claims that they have 'perfected the software synth version of the ARP 2600'. The company's marketing blurb also claims that its developers started 'with an accurate digital emulation of every module from the original ARP 2600, and then added new features', but given the differences between the original synth and their emulation, I'm not sure that they were accurate enough.

I'm also unhappy with WOW's claims that they 'spent two years reinventing the mathematics behind analogue emulation, developing proprietary and optimised virtual analogue oscillators and filters'. It seems far too grandiose to claim that they have reinvented modelling, and, given the CPU drain occasioned by *TimewARP*, the word 'optimised' appears to be misused.

Some people have suggested that, while less enhanced than the Arturia *2600V*, *TimewARP 2600*'s sound is closer to that of the original ARP. I beg to differ: while the graphic design of *TimewARP* is truer to its inspiration, it exhibits a similar degree of sonic authenticity, or lack thereof, as other soft synths. Whether this is a problem or not is open to debate — many of the issues I've raised have no bearing on whether *TimewARP 2600* is a good synth or a bad one; they simply tell you whether it is an accurate recreation of the ARP 2600 or not. So let's view *TimewARP 2600* as a synthesizer that has the look and feel of the ARP 2600, and which shares much of its character. It then fares quite well, although I can't help feeling that the better-specified Arturia *2600V* fares rather better. SOS

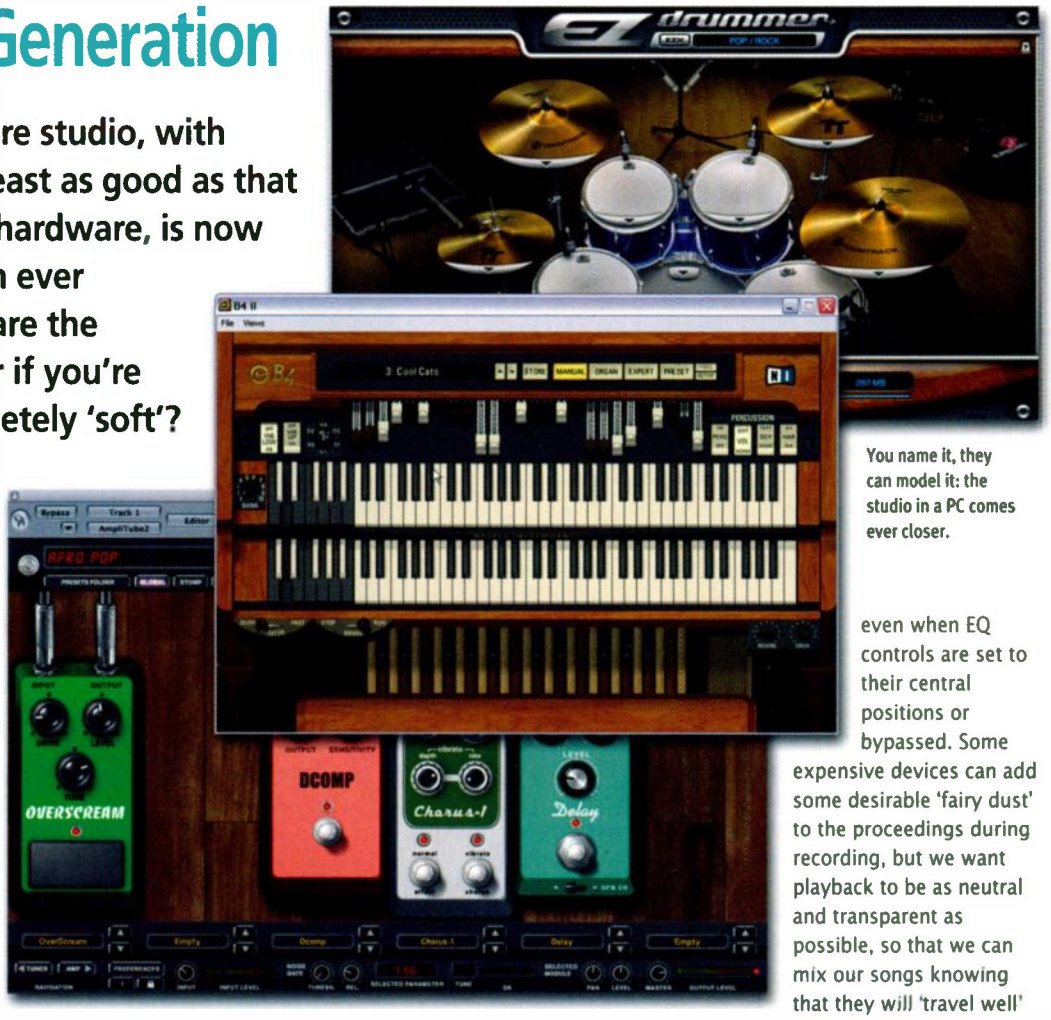
information

£ £169.99 including VAT.
T M-Audio +44 (0)1923 204010.
F +44 (0)1923 204039.
E info@maudio.co.uk
W www.maudio.co.uk
W www.wayoutware.com

Software Studios

The Next Generation

The totally software studio, with sound quality at least as good as that offered by studio hardware, is now more feasible than ever before. But what are the factors to consider if you're going to go completely 'soft'?



You name it, they can model it: the studio in a PC comes ever closer.

Martin Walker

A few years ago lots of PC musicians started to get excited about the possibility of being able to run all the audio and MIDI tracks required by a song, plus effects, on a single PC. After all, we were being offered lots of new software synths, samplers and plug-in effects, all claiming to sound as good as their hardware equivalents. At the time, the only thing holding this concept back was lack of CPU 'grunt' — it was just so easy for everything to grind to a halt when a decent reverb plug-in could consume 50 percent of your entire processing capability. However, the arrival of dual-core PCs has resulted in a colossal leap in processing power, so is it now time to sell our remaining hardware and go totally 'soft'?

Audio Quality Benefits

Until a few years ago, most musicians started their musical journey with an instrument, such as a guitar or keyboard. When they wanted to record some music, they bought a reel-to-reel tape deck or cassette multitracker. Once they had two two or three instruments to record, they bought a small mixing console, and perhaps a microphone or two for recording vocals or other acoustic instruments. The mixer allowed them to connect all these sources, tweak their sounds using its EQ controls, plug in hardware effects units to add reverb, chorus, and so on, and then mix the sounds together, using the channel faders, to produce the final stereo output.

Today, many musicians start this process in reverse, by first buying their PC and then looking at what needs to be added to it for making music. Ironically, freed from the preconception that a mixing console is required, many adopt the much simpler approach of relying totally on the facilities of their audio interface, which may result in better audio quality. When I started recording my music, many years ago, I learned the hard way that even keyboards and hardware synths can sound significantly better if you listen to them plugged straight into a power amp/speakers rather than first passing them through a budget mixing console, as handy and versatile as these are.

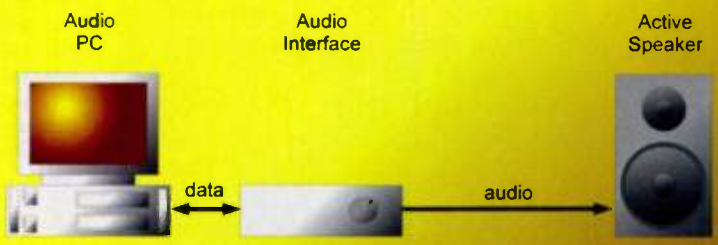
Every device you pass your audio through affects its sound quality slightly, even when EQ controls are set to their central positions or bypassed. Some expensive devices can add some desirable 'fairy dust' to the proceedings during recording, but we want playback to be as neutral and transparent as possible, so that we can mix our songs knowing that they will 'travel well' and essentially sound the same on other systems.

Thus one of the incidental advantages of recording directly to the inputs of your audio interface and playing back directly from its outputs to a pair of loudspeakers or headphones is that the audio signal path often ends up considerably less complex, and this may mean that the end result sounds cleaner and more transparent. Some musicians also find that replacing a large mixing desk with a comprehensive multi-channel audio interface gets rid of nasty early reflections between their speakers and listening position, thus improving the acoustics of their studio! You can see various suggested setups with simple audio paths in the diagrams opposite.

Soft Synths: Poor Relations?

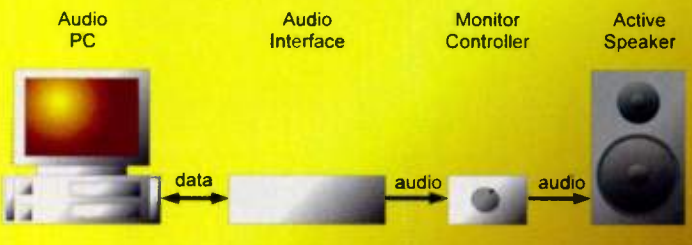
If you're considering creating all your music inside a PC, it's important to make sure that ▶

The Simplest Audio Path



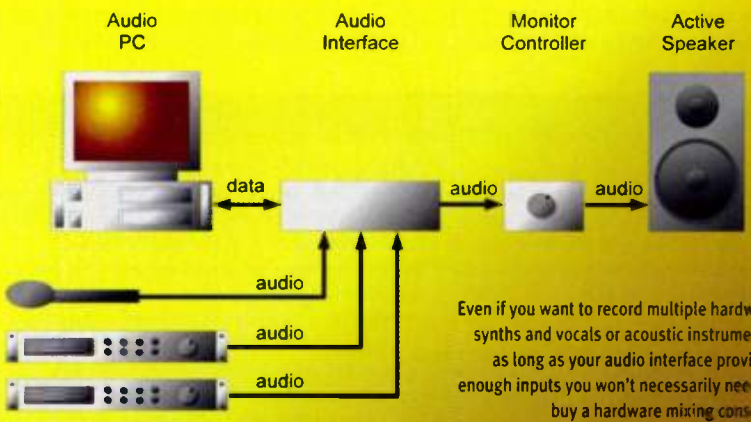
The simplest possible audio path requires a single stereo pair of cables between the output of your audio interface and your active speakers (or power amp and passive speakers), but relies on your audio interface providing an analogue output-level control to easily adjust speaker volume.

Adding A Monitor Controller



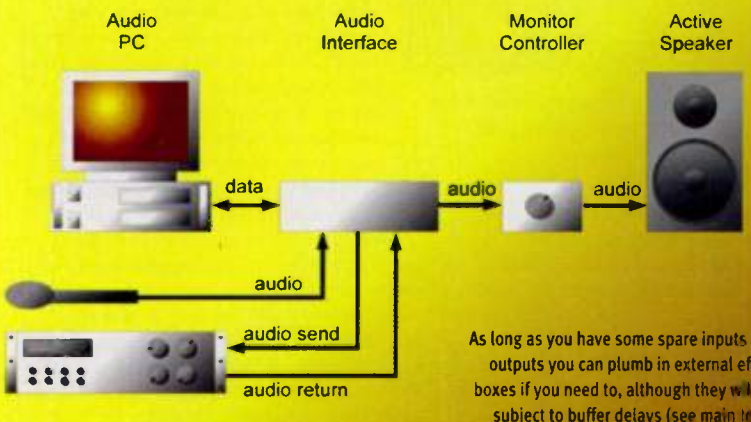
If, like many audio interfaces, your model doesn't provide a handy analogue level control, you'll really need some sort of monitor controller, connected as shown here.

Additional Recording Options



Even if you want to record multiple hardware synths and vocals or acoustic instruments, as long as your audio interface provides enough inputs you won't necessarily need to buy a hardware mixing console.

Adding An External Hardware Effect Unit



As long as you have some spare inputs and outputs you can plumb in external effect boxes if you need to, although they will be subject to buffer delays (see main text).

With the audio quality from soft synths such as Arturia's *Moog Modular V* being so good, is there any longer a need to worry about replacing hardware with software?

the software synths and plug-ins you choose provide audio quality on a par with their hardware counterparts. When I first broached the subject of the software studio, way back in *SOS* November 2000, many musicians remained unconvinced that a software version could ever sound as good as the 'real thing'. However, my view remains exactly the same as it did then: there's absolutely no fundamental reason why a software synth should sound any different from a hardware synth containing DSP chips.

My only reservation at the time was that, unlike developers programming for

Monitor Controllers: Some Options

If your sound creation is done entirely in software the very simplest hardware setup is to connect the output of your PC's audio interface directly to a pair of active monitor speakers, or to a power amp and pair of passive speakers (see the first diagram on the previous page). With some acoustic treatment in your studio, the results obtainable via this simple audio path can be superb.

However, such a minimalist approach has one possible drawback: your audio interface really needs an analogue output-level control so that you can easily adjust the speaker volume. Using the digital faders found in the software Control Panel utilities included with many audio interfaces not only throws away digital resolution, but also runs the risk of a full-strength signal accidentally reaching your speakers or your ears. Some active speakers do have level controls, but jumping up and down to adjust them is tedious, especially when there are usually two controls (one on the back panel of each speaker) whose positions you have to keep in sync to maintain a balanced stereo image.

Sadly, few audio interfaces currently provide a dedicated analogue output-level knob, so many musicians are instead buying a 'monitor controller' to perform this function. Recommended models include the Samson C-control (www.samsontech.com) for about £80, Mackie's Big Knob (www.mackie.com) for about £250, and the £400 Presonus Central Station (www.presonus.com). All offer three sets of switched outputs so that you can connect up to three pairs of speakers to hear how your mixes translate to other systems, plus talkback functions with built-in microphones, allowing you to talk to musicians in your live room, if you have one. The

Few audio interfaces provide an analogue output-level control, so if you dispense with a hardware mixer you may need a monitor controller to provide this function.

Presonus model also features a totally passive audio path (no power supply required, and containing no transistors, FETs or integrated circuits), so it can't add any background noise, distortion or other colouration to your audio as it passes through — although the relatively simple active path of the other two should only have a tiny effect on your sounds.

If you don't have multiple speakers or require talkback functions there are several simpler products available, all of which feature high-quality passive components. The £99 PVC (Passive Volume Control) from NHT (www.nhthifi.com) features a clearly calibrated, large rotary knob in a small one-third-rack width or desktop unit. It allows precise 1dB level adjustments over a 40dB range and features completely balanced operation via its Neutrik combo XLR/TRS input jacks and XLR outputs. Another, slightly more sophisticated, alternative is SM Pro's M-Patch (www.smproaudio.com). Once again, this one costs about £99, and it has identical I/O socketry, but also offers two switched inputs, each with individual rotary level controls but smaller knobs, and two switched outputs, all housed in a half-rack-width case. SM Pro have also just launched a new M-Patch 2, at about the same price, with the same passive main signal path, but with larger knobs, additional stereo/mono and mute switches and a budget headphone amp with its own level control.

While we're talking about hardware, there's also a lot to be said for the dedicated control surface. A knob for each function makes real-time sound-tweaks far more intuitive. Once audio reaches your MIDI + Audio sequencer, you can use one of the many available control surfaces to do your mixing, sound editing, synth tweaks, and so on, just like you did with hardware.

QUESTED

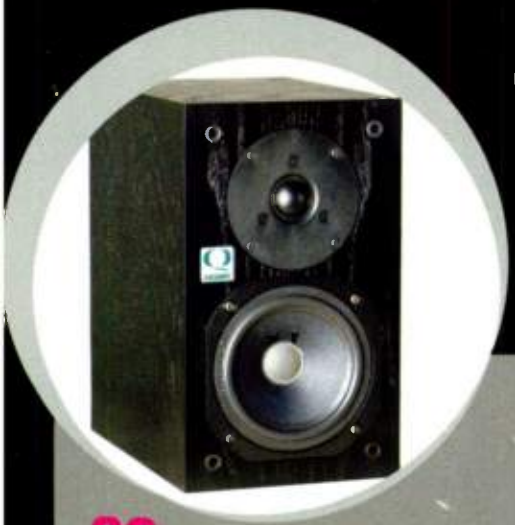
The truth is out there...

... just listen!

QUESTED, whose origins go back to the recording industry in the early seventies, has a world-renowned reputation as a company that designs and installs monitoring systems in major studios.

With such renowned studios and producers as Abbey Road, Sarm, Crescent Moon (the Estefans), Remote Control (Hans Zimmer) and Michael Cretu (Enigma) this reputation is well deserved.

The new S-series active monitors brings this quality of sound to any studio.



S6

2-way, active, self-powered monitors

The smallest monitor in the S Range. Its integral electronics and compact size make it an ideal choice where space is at a premium. It is fully magnetically shielded, so can be used in close proximity to CRT monitors and TVs. The S6's infinite baffle design and excessive amplifier headroom accumulate to provide a sonic performance that is truly remarkable for a box this size.

inc VAT £763.75 ex VAT £650.00 price per pair

S8

2-way, active, self-powered monitors

The new S8 is a self-powered, two-way active monitor suited to a wide variety of applications including near to mid-field monitoring for recording and mixing, programming, post production suites, broadcast and surround sound. The 370 Watts from its two integral amplifiers makes the S8 a powerful choice. It's high power output and extended low frequency response makes it suitable for reasonably large rooms.

inc VAT £1633.25 ex VAT £1390.00 price per pair

S7

2-way, active, self-powered monitors

The S7 is a self-powered, 2-way active monitor. Its integral electronics make it a compact yet powerful choice providing a combined output of 190W for each monitor. It is fully magnetically shielded, so it can be used in close proximity to CRT monitors or TVs. The S7 is a finely tuned reflex port design. Extensive work has gone into the amplifier and active cross over, employing world leaders in amplifier design.

inc VAT £1128.00 ex VAT £960.00 price per pair

SB10

Active sub-woofer

The SB10 is the latest edition, a powered sub to complement the above and comes complete with controller for phase, volume and crossover control.

inc VAT £1169.13 ex VAT £995.00
price for system: controller and sub

Exclusively distributed in the UK by SONIC8

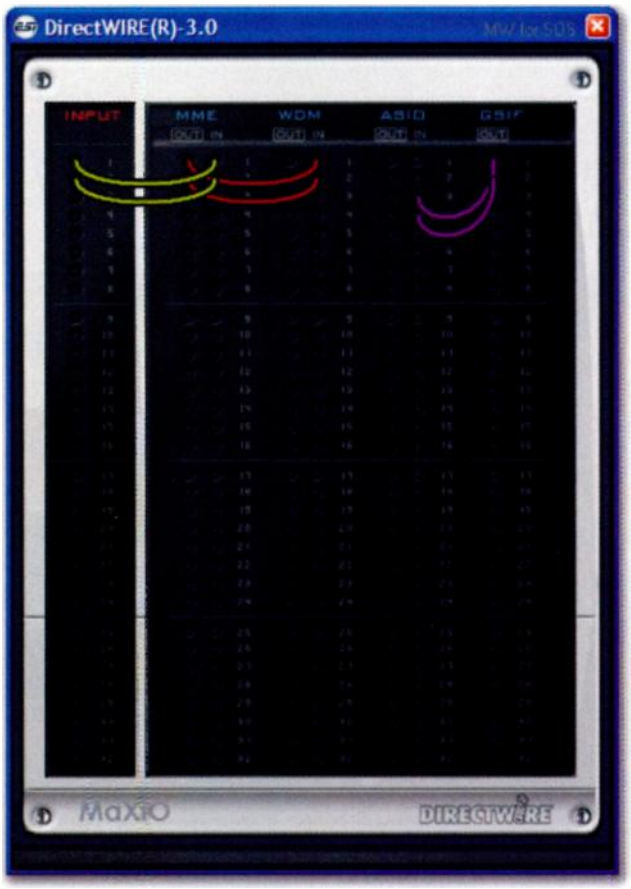
QUESTED
www.quested.com

SONIC 8
www.SONIC8.com
08701 657 456

dedicated DSP chips, who could splurge processing power on a few luxury plug-ins, those designing for native (computer) processing had to keep overheads as low as they reasonably could, to avoid accusations that their products were 'CPU hungry'. This could result in corner-cutting. However, whereas in 2000 we were getting excited about 750MHz processors, we now have dual-core versions wherein each core is clocked over four times as fast. And with this ability to compute eight times faster, cutting corners can become a thing of the past.

However, one area does catch people out. While many hardware synths incorporate multi-effects that result in all their presets sounding polished and complete (but which sometimes have to be disabled because they can be overpowering when heard in the context of a complete mix), software synths can sometimes sound raw in isolation (other descriptions I've noticed on forums include "tinny" and "thin", or just "bad", but they are all referring to the same thing). This is simply because their developers expect that you'll mostly be using plug-in effects to mould them to perfectly fit your songs.

Moreover, polished tracks generally require a lot of creative input, so it's perhaps not surprising that some beginners feel disappointed when they create a track featuring a clutch of soft synths and it sounds a bit flat in comparison with commercial music. This isn't because software "sucks"; it's more the case that a little more effort may be required in honing the sounds and effects.



If you want to run several music software applications side by side, there are various ways to combine their outputs. One of the most elegant is ESI Pro's *Direct WIRE* utility, which lets you plumb the output of one application into the input of another, allowing one app to record the contribution of the other.

Software Studio Advantages

The biggest advantage of the software studio is undoubtedly that each song is stored along with all the sounds it uses, plus effect settings, mix settings, and so on — in other words you have total recall, just like really expensive analogue mixing desks with motorised faders that remember their positions. Your entire sound library also resides on the computer, making it very easy to audition many different sounds with a few mouse clicks, instead of laboriously loading sample and synth banks into your hardware. Nearly all software sequencers also let you capture real-time movements of on-screen faders and other controls, alterations to sounds, or tweaks to plug-in effects. This automation goes far beyond what was possible a few years back, and enables far more creative possibilities.

Being able to memorise absolutely everything means that you never have to worry about losing track of a song — even if you have to leave it for months, when you come back to it every setting and sound will be perfectly preserved. This is great for not only the home studio owner, but also for professional bands, producers and recording engineers. Anyone who requires lots of different mixes and remixes to suit different markets round the world also benefits: they can load up any song from any album within seconds, start on a new version and save it under a different file name, without disturbing the original.

One-box Songwriting

Many PC musicians prefer the all-in-one approach of 'software studio' applications such as Propellerheads' *Reason*, Cakewalk's *Project 5*, Arturia's *Storm* and Image Line's *FL Studio*, where a single programme contains a virtual version of everything you might find in an electronic music studio, including synths, sample players, drum machines, effects, a sequencer to record and play back the notes, and an audio mixer to mix them all together. However, for a more open-ended approach, the flexible MIDI + Audio sequencer still reigns supreme, *Cubase* and *Sonar* being probably the most popular on the PC.

Some PC audio applications, such as *Gigastudio* and *Audiomulch*, prefer to run in stand-alone mode rather than inside a sequencer, so if you have to run them alongside a MIDI + Audio sequencer you'll have to find a way of combining their audio outputs. The traditional way is to allocate the final stereo output of each application to a different stereo output on your audio interface and then plug them into different channels on a hardware mixing console (I've worked in this way for years).

However, there are various options if you don't have a mixing console and want to keep your signal path as short as possible. One is to buy an interface with multi-client drivers that allow different applications to be allocated to the same pair of stereo outputs. You can't always tell this is possible from the documentation, but I've done it with quite a few interfaces over the years.

Other interfaces may provide several stereo playback channels that can be individually allocated to different applications, but have a DSP mixer utility that lets you mix these playback outputs to emerge from a single physical output socket on the interface. Examples of these on the PC include most models from Edirol, Emu, ESI, M-Audio and Terratec. A few, like Echo's Mia, provide several virtual outputs that you can allocate to different applications and then mix, to emerge from its single physical stereo output.

The only disadvantage of these approaches is that although you hear all the applications mixed together through your speakers, at some stage you'll need to record this combined output. Some interfaces provide very useful internal loopback options, the best known being the *Direct WIRE* utility functions of the ESI range (see screen above), which allow you users route the inputs and outputs of any of its MME, WDM, ASIO or GSIF driver formats to each other, so they can record the output of one application in another, to capture it as an extra audio track.

Other interfaces with integral 'zero latency' DSP mixers may let you re-record the output of their mixer, so you can capture any combination of the software playback channels from different applications, plus additional signals arriving at the interface inputs, as a new 'input' in your sequencer. Failing this, you can nearly always loop cables from the interface outputs back to its inputs, to record the combined signal, although if you have to rely on analogue I/O your final audio mix will have to pass through both D-A and A-D converters, lengthening the audio path and compromising audio quality slightly.

If your particular interface doesn't provide any of these options, there may yet be another internal mixing solution, this time thanks to software. Rewire is a technology introduced by Propellerheads that's now included in quite a few audio applications, including Ableton *Live*, Adobe *Audition*, Apple *Logic Pro*, Arturia *Storm*, Cakewalk *Sonar* and *Project 5*, Digidesign *Pro Tools* 6.1, MOTU *Digital Performer*, Propellerheads' own *Reason*, RMS *Tracktion*, Sony *Acid*, Steinberg *Cubase* and *Nuendo*, and Tascam *Gigastudio 3*. If your audio applications support Rewire, one can be the 'synth' to generate sounds, while the other becomes the 'mixer' and receives one or more output streams from the synth while simultaneously locking them in perfect sync.

Outside Help

Even if you want to generate the bulk of your sounds using software applications, you may still want to occasionally plumb in some hardware. If you've got a hardware synth that offers unique sounds, you can record its output directly into your PC's audio interface as an audio track. If you've got several such synths, you simply need an extra mono or stereo input for each one (see the third diagram on page 161) and can then record or audition them all simultaneously, in real-time, with added plug-in effects.

There may also be times when you want to treat one or more software-generated tracks with hardware effects that you can't duplicate in software — perhaps a favourite EQ, compressor or reverb, for instance. To connect external effect boxes, you once again need some spare inputs and outputs on your audio interface to act as send and return channels (see the last diagram on page 161). I covered this topic in more depth back in *SOS* March 2004, in a feature on using hardware effects with your PC software studio.

If you like the idea of the software studio but want to give your PC a helping hand with effects processing, you could consider installing one or more dedicated audio DSP cards, such as the TC Powercore or Universal Audio UAD1, inside your PC, to provide 'hardware quality' effects that simply appear to your PC as extra VST plug-ins.

If you find that a single PC still doesn't have enough clout to run all the software synths and plug-ins you fancy, you could create a PC network to share the load, as described in *SOS* August 2005 in my feature on spreading your music across networked computers. One of the options outlined there, using *FX Teleport* (www.fx-max.com), relies on LAN (Local Area Network) connections to ferry both MIDI and audio to your secondary and additional PCs, with all the mixing still being carried out digitally by your main PC sequencer application, so your analogue audio path can still remain simple for optimum quality.

Finally, with sufficient spare I/O channels on your audio interface you could even patch in an analogue summing mixer such as the Audient Sumo (reviewed *SOS* February 2006), to replace software digital mixing of all your audio channels with analogue hardware mixing, while still avoiding a traditional mixing console. However, I suspect this is going too far. In most setups your audio will sound significantly better if you instead spend the money you might have spent on a summing mixer on an interface with better converters in the first place, and keep the shortest possible analogue signal path. SOS

The Other Side Of The Coin

It's only fair to point out that the rise in popularity of the software studio has had some repercussions in the wider musical world, both good and bad. Those who have started to replace their old hardware synths with software equivalents and now want to sell the hardware to make more space in the studio will find that second-hand prices seem ridiculously low (I just can't bear to part with my immaculate Korg M1 for £50, for instance). On the other hand (as you'll see if you visit the *SOS* Vintage Gear forum), there are still plenty of musicians who swear by hardware rather than at it, so now's the time to snap up lots of bargains.

Software Studio Monitoring

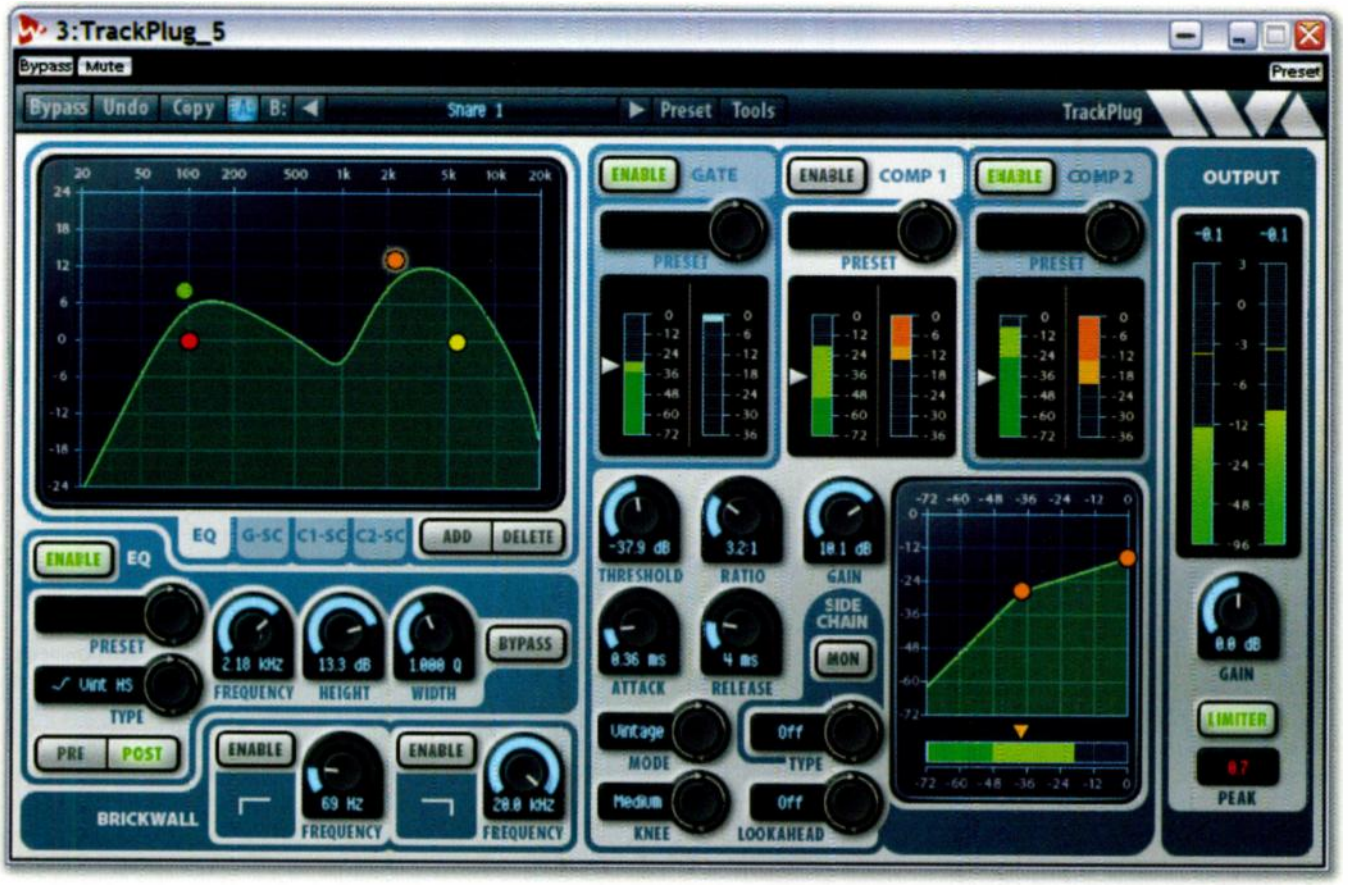
Nowadays, nearly all audio interfaces offer monitoring functions, so you needn't be too apprehensive about abandoning a traditional hardware mixer when 'going soft'. So-called zero-latency monitoring basically lets you hear the signals you're recording almost instantly, by providing a direct link between the output of the audio interface's A-D converters and the input of its D-A converters. Incoming audio is therefore only subject to conversion delays, which generally amount to about 2ms (1ms for each converter). This is negligible, being identical to listening to audio from a loudspeaker just two feet away, and even vocalists shouldn't find many problems listening to their voices on headphones with such a short delay.

Unfortunately, if you want to add plug-in effects to the signals being recorded (for instance, many vocalists like to hear some reverb on their performance to help them with pitching, while electric guitarists may want to pass their instruments through distortion or amp-modelling plug-ins), you face the additional path of getting the signals into the PC's CPU and back out again. With a fairly typical audio-interface latency setting of 6ms, the total delay becomes 14ms (1ms through the A-D converters, 6ms through the interface's input buffers, a further 6ms through its output buffers, and a final 1ms from the D-A converters).

This 14ms delay is on the borderline of acceptable for the majority of fretted and keyboard instruments, but way above the comfort zone for vocalists. You can try reducing your audio interface buffer size to 3ms, for a total path delay of around 8ms, while some musicians even try a 1.5ms setting, for a total delay of around 5ms. The only disadvantage of this drop in latency is that your CPU overhead may rise considerably.

One way around this situation is to choose an audio interface with built-in DSP effects, such as Focusrite's Saffire or one of Emu's DAS models. The beauty of these effects is that they also have zero latency, and you can apply them to vocals and instruments when monitoring, even if you intend to actually record the signals dry. Another clever twist can be found in M-Audio's Firewire 1814 interface, which features an Auxiliary buss that you can either use for setting up a separate monitor mix to send to headphones, or for adding external hardware effects to live inputs.

For the very simplest audio path, look for an audio interface that provides a dedicated and extremely handy analogue output level control, like the Focusrite Saffire shown here, which also offers built-in DSP effects that you can use during recording, with no latency.



Plug-in Bundle For Windows & Mac OS X

Wave Arts Power Suite 5

The latest incarnation of Wave Arts' plug-in suite is a treat for the eyes and the ears.

Martin Walker

Last time I looked at the *Power Suite* bundle of processors and effects it was up to version 4.06, but Wave Arts don't rest on their laurels, and version 5 adds a significant number of new features to the package, not least a new and more sophisticated graphic look, enhanced preset management and 192kHz support. It still comprises five plug-ins — *Trackplug*, *Final Plug*, *Multi Dynamics*, *Masterverb* and the new *Panorama*, which replaces the previous *Wave Surround* — and formats supported are VST and RTAS on Mac and PC, plus AU and MAS on the former and Direct X on the latter.

Trackplug combines a clean and versatile 10-band EQ with a comprehensive compressor and gate, and is already very popular around the world for its low CPU overhead, as this often lets you run significantly more instances than many competitors. However, this time around almost every aspect of it has been improved. A new brick-wall filter is now present at the beginning of the signal chain to remove hum, thumps and hiss before further processing, and the EQ section now has 11 options for each band instead of the previous seven, adding variable-Q resonant options to the existing fixed-Q low shelf and high shelf, plus the asymmetrical transition of the vintage 'analogue' low and high shelf.

Another more fundamental change is that there are now two compressors in series rather than one, each with five modes. The

SOUND ON SOUND

Wave Arts *Power Suite 5* $599

pros

- *Trackplug* now has vintage-style compressor options which sound good.
- *Masterverb* is far more sophisticated than previous versions.
- Elegant new graphic interfaces.
- Low CPU overheads throughout.

cons

- Some *Panorama* movements may not work for everyone.

summary

Wave Arts' *Power Suite 5* is the most significant upgrade yet, and adds a new level of sophistication to this already popular bundle of plug-in processors.

Clean Peak and Clean RMS modes from earlier versions offer transparent results, while the Vintage Peak and Vintage RMS modes add odd harmonics when driven hard, and Vintage Warm offers both even and odd harmonics. The comprehensive manual explains the differences well, but there's no substitute for hearing them in action — the new vintage modes offer plenty of analogue colour and warmth, and the ability to chain two compressors with different settings increases the possibilities still further.

There are also new side-chain modes for more specialist compressor duties, such as de-essing using a side-chain EQ. More exotic possibilities include the twin side-chains of EQ Compare mode, so that you can, for instance, have a band-pass filter detecting sibilance, and a notch filter at the same frequency in the alternate band. The compressor then only operates when there's sibilant energy, so *Trackplug* doesn't mistakenly remove any other parts of your track. Experimentation is the key, but there are loads of presets on offer for each *Trackplug* section, to get you into the right ball park. Finally, *Trackplug* has a new Peak Limiter on its output to help avoid nasty digital clipping. It's certainly handy, and is fairly transparent too, unless you whack up the gain.

Plugging Away

Multi Dynamics is a versatile plug-in offering up to six bands of compression or expansion/gating, and apart from the graphic makeover is essentially the same as before, except for a new Clean/Vintage compression-mode option. Vintage works like a classic analogue compressor and can, at extreme settings, create low-end fattening and transient shaping, while Clean retains the transients and avoids distortion where you require more transparency. I found the difference on some material quite profound, and overall I think the new modes are a useful and creative addition.

Final Plug is designed for peak limiting with optional bit-depth reduction and noise-shaped dithering, to allow you to increase the final level of a song in a transparent manner, unless you want to use it as an effect (think of Waves' *L1 Ultramaximizer* and you're in very similar sonic territory). Version 5 sports a new Auto Release mode, like that of Waves' *L2*, which will calculate the limiter's release time automatically. I could immediately hear the improvement, with significantly fewer artifacts such as pumping or distortion on some material.

Old Master, New Tricks

When I first reviewed *Masterverb*, way back in *SOS* February 2002, I was impressed by its smooth sound and low CPU overhead; each new version since has brought a few refinements, but version 5 is the result of a major overhaul that has added a huge number of new features. There's still a draggable 'ball' to set room size (horizontal) and decay time (vertical), with the frequency response over time being displayed graphically in a 3D window, but now there's a new section devoted to early reflections. Here you can choose from 13 characteristic reflection patterns pertaining to various rooms, chambers, plates, halls, churches, and even single and ping-pong echos. Some musicians haven't yet realised how essential these can be to 'place' a sound within a space. Setting the new Late/Early mix control to '0 percent Late' allows these early components to also be used in isolation, to thicken up sounds without giving them obvious reverberation.

Both early and late portions now appear in a new Time Response display, while the Late reverb algorithm now has a plate as well as a hall option, and early and late damping is now considerably more sophisticated, with three-band EQ controls. There's also a new Envelope section with classic threshold, attack, release and hold controls, so you can create gated and 'reverse' reverb.

While importing *Masterverb 4* presets shows that the basic hall algorithm is the same, the new *Masterverb 5* preset

collection offers far smoother rooms and halls, shimmering plates, huge gated drum sounds and exotic special effects. The new features have totally transformed the overall sound into something decidedly more versatile and sophisticated. Although many of us now reach for realistic convolution reverbs, *Masterverb 5*'s traditional algorithmic reverb sounds excellent, is much quicker to tweak, is more versatile, and used less than half the CPU overhead of any convolution reverb in my collection.

A Panoramic View

Panorama takes the previous *Wave Surround* plug-in to a completely new level. Still based on HRTF (Head Related Transfer Functions) and binaural synthesis, and with playback modes to suit headphones or various speaker arrangements, it once again allows the user to move the virtual speakers in the graphic window to place sounds in between the speakers, beyond them, and (on some systems) make them appear to move behind the listener, with or without optional reverb to make sounds fit into a real acoustic space.

However, the new version not only gives you control over the width of your image, from mono to super-stereo, but also the power to position your sounds in 3D space, in either Polar co-ordinates (azimuth and elevation in degrees, and distance in feet or metres), or Cartesian co-ordinates (X, Y and Z, all in feet or metres). The Top (plan) view can also be switched to a rotatable 3D box view, and there's a new MIT KEMAR HRTF head model to supplement the previous 'Human' one.

The reverb section has also been enhanced by a choice of room algorithms, and there's a completely new Reflections section. Once you click on that section's title bar you find a comprehensive set of extra controls that let you set the distance and material of each of the six faces of your virtual space. The most exciting aspect is that you can grab the source sound in the overhead view window and drag it around in real time between the speakers, behind them, closer and further away, and even up and down, with optional Doppler pitch processing for generating realistic motion effects. As with *Wave Surround*, I found the 'beyond the speakers' effects worked well, as did the new width and distance parameters, particularly with the Reflection and Reverb sections on, so that you could hear more reflected sound as it moved further away.

However, for me the 'behind' positions still didn't work on either headphones or speakers, becoming impossible to pinpoint, and I also had difficulties believing the up/down of the elevation parameter — you may have more luck, though. Nevertheless, the draggable source is totally automatable, letting you record complex moves in real time, and with the Doppler mode on I began to experience motion sickness within a few seconds, particularly in speaker mode with percussion tracks. For anyone wanting to create movement in their mixes, this plug-in is great fun.

Final Thoughts

Overall, Wave Arts' *Power Suite 5* is a huge leap forward from previous versions, particularly in the areas of compression, reverb and spatial effects. Audio quality is on a par with most of its rivals (as you'll hear if you take advantage of the demo version downloads), processing overhead is lower than many competitors, and at $599 (about £344) it's also good value for money. If you don't need all five plug-ins you can buy any three for just $349 (£200), any two for $299 (£170), or each one individually for $199 (£115). SOS

information

£ $599.
E info@wavearts.com
W www.wavearts.com

PHONIC
www.phonic.com

LOUD AND CLEAR

HELIX BOARD 24 FIREWIRE

Multi-track recording and mixing via firewire along with standard analogue connections

PERFORMER SPEAKERS

90-275watt professional active and passive loudspeakers

XP POWER AMPLIFIERS

140-3200watt stereo power amplifiers with high power to value ratio

SHURE®

Contact exclusive UK distributor, Shure Distribution UK on 020 8808 2222
email info@shuredistribution.co.uk or visit www.shuredistribution.co.uk

Photos: Richard Ecclestone

Jim Moray

Folk Music For The 21st Century

The idea of bringing folk music up to date is not a new one, but few people have taken it quite as far as Jim Moray. His material may be traditional, but his approach to music technology is as modern as it gets.

Sam Inglis

To some, he's breathed new life into a stale folk scene. To others, he's a vandal whose approach to our musical heritage will have Ewen MacColl spinning in his grave. Either way, there's no denying that Jim Moray's take on British folk music is a pretty radical one. His first album, *Sweet England*, saw him framing traditional ballads in an elaborate web of Marius de Vries-style orchestration, electronic percussion and heavy metal guitars. The follow-up, *Jim Moray*, replaces the florid orchestral arrangements with an equally startling indie-rock slant on things, and was released on May 1st this year.

"I don't think you get up in the morning and think 'I'm going to do something really innovative,'" insists Jim, when asked how he ended up taking this particular musical path. "You just do what comes naturally. I don't think what I'm doing is particularly revolutionary, it's just playing the songs in the way I want to play them. There's a lot of things that are present in traditional music that really get me going, but there's a lot of things that annoy me, and there's a lot of things that I miss from other music that I like, so it's trying to make a big pot of all the things that I like."

So why has his music upset so many in the folk scene? Partly, it's the scene itself ("Folk music thrives on being very nice, and everyone liking everyone else, but in a way that's not really true"), and partly it's his carefree attitude towards the songs. Jim Morays sees folk songs not as sacred relics, but as creative launching pads for his own musical and lyrical ideas; many of his tracks are not so much recordings of the traditional songs, but new songs that build on fragments of old ones. "A lot of this new album has been taking four lines from a traditional song and then writing another eight verses of my own. I use the original tunes quite a lot, but the original tunes are quite bendable, and also, a lot of them are related to each other. There are a lot of similar tunes that you can mix and match phrases from, and I do quite a lot of that. In a way, that's the most traditional thing you could possibly do. That's what traditional music is, it's not rigid or set in stone. The idea is to put your own spin on whatever's there. English folk music is my cultural heritage, it's like a gift that's been passed down, and it's as much my right to decide what I do with it as it is anyone else's to not like it, or to do their own thing."

Variety Show

The new album was written and recorded over the course of 18 months at Jim Moray's large Bristol studio, which is home to an impressive and sometimes oddball selection of instruments and recording gear. He plays most of the instruments on his albums himself, including drums, keyboards, guitars and basses, and he's very sensitive to the possibility of relying too heavily on any one

Jim Moray's Bristol studio.

thing. Most of the first album was written using *Logic* as a songwriting tool, and in many ways the rockier feel of the second is a reaction against this. "The first record was a lot more based around the backing, and the problem with that is that when you strip it down, the song disappears. If you try to play it on an acoustic guitar, the song evaporates.

"If you write on guitar in standard tuning all the time, you fall into familiar shapes and patterns, and you get stuck in a bit of a rut of playing your favourite chords. If you try writing on another instrument, it quite often opens everything out, and it's got a bit like that with the computer for me. I don't tend to write as matrix-based tracks, it's all as staves, so I go 'Right, I'll start with four-part strings,' and then you end up writing all the same notes. So I've done quite a lot of writing recently without an instrument at all, just trying to sing or hum, focusing on the melody and letting everything else come later."

As you might expect, there's rarely a clear distinction between writing and recording in Moray's studio. "A lot of the stuff on my albums is orchestral-based, or programming-based, or comes from a tiny little sound. On the first album there's a song called 'Two Sisters' that's all around a loop — it was the end of a guitar take; after the take had finished I accidentally hit a harmonic and muted it. I cut it out and then looped it and it took on a life of its own, and then when it was time to do other bits, the song was there and I was adding to it, just doing overdubs. I quite enjoy those afternoons when you're trying to make a sound, but you've no idea how, so you're playing a radiator with a contact mic on, or scraping a mug across the bristles of a brush.

Some of Jim Moray's guitars, including a custom Atkin acoustic (right) and 'violin' bass (centre).

"When I first started recording stuff on four-track, I had a Yamaha keyboard with an inbuilt sampler, which had really grainy-quality sampling. What I used to do was record myself singing and my playing as parts, and I still do that a lot with *EXS24*: sample things in and make my own patches, rather than just using generic ones. There's quite a few things on the album that are sounds I've recorded on Minidisc or Dictaphone, transferred into *EXS24*, looped and used as a sound bed — at the start of the song 'Nightvisiting', there's a sample of some children in a playground, and I've just fiddled with the pitch-bend. Generally, I often use instruments as sound generators, like the upright bass — if you bow it, you've got a pretty pure waveform to stick plug-in effects on. I quite like combining effects in a different order to make something new, like delays where you plug the return into a pitch-shifter and every time the delay comes around again it's gone up a tone."

Orchestral Manoeuvres

"I use a lot of the East West *Symphonic Orchestra Gold Edition*, for mocking up strings and brass — about a third of the stuff on the album is from that rather than real players. Sometimes it gives better results than real players: if you've got something that needs really precise repeatability, it's very good, but there's a bit of an art to programming those things."

The real orchestral parts are one of the few areas where Moray hires in other people to play, but his method owes a lot to the sequencer-based approach to recording. "I tend to put the orchestral parts on quite early, because the way I do those parts is quite indecisive. Rather than doing the arrangement and getting it all right on paper, then overdubbing it in an orchestral session, what I tend to do is write out bits that I know I want to be in there, and I put them on separate pieces of paper as eight-bar sheets. I get the string players in and physically conduct them through all the bits, but they can't actually hear what they're doing, they're just playing all these little eight-bar bits. So

I'm almost making my own sample kit for each song, that I can then cut out and put in a different order and mix and match sections. I kind of make my own phrase kit for each song, and then sort out the arrangement on the computer afterwards."

The method might sound haphazard, but, as with many of Jim Moray's unusual ideas, there are sound reasons for doing it this way. "It's a lot easier, and also you tend to get one really good performance of each bit. There's something your ear latches onto. If I, for example, have a rising movement in the first violins and the cello is holding a note, what I can do is take those violins and put a different note underneath them, and they're looping but the rest of the strings are moving. It gives this kind of symmetry. What I like about electronic music is the repeatability of it; there's something hypnotic about it, and I think that applies as well to really good orchestral playing, but I'm generally working with people who have never seen the music before.

"With production, the job is to get the best possible performance out of people without them knowing what you're doing. So with strings I tend to not come clean about what it is I'm trying to achieve. I just tell them what's right and try to give them pointers. I usually do four takes of a quartet, and sometimes extra cellos, and I usually do double-bass myself, here. I find that if the bass is out of tune or out of time, that's the thing that has ruined a lot of tracks for me. I don't know if you have ever made a demo and you can't work out what's wrong with it, everything's there but it just doesn't work? I've found that when I've gone back with a more critical ear, it's the bass end, so I tweak the timing and

The 'stick' electric double bass, played through a hardware Auto-Tune unit, is often doubled with a Hofner violin bass.

the tuning... I use a stick electric double-bass. I've got a rack Auto-Tune unit, because I'm not a double-bass player, but it's great for bowing. You can even distort it and compress it a lot and it sounds almost like a triangle-wave synth.

"A lot of tracks on my albums don't even get a bass guitar, but when they do it's this one," he says, indicating a copy of a Hofner violin bass. "It's a lot less boomy, you get a lot more plectrum attack and it doesn't get in the way so much. I usually have a sustained note on the double bass and I'm doing a rhythm underneath it on this."

Keeping Music Live

Although Jim Moray's studio space would probably be large enough to swallow the average home studio four or five times over, he's chosen to keep it as one large and very live-sounding room rather than build partitions, booths or a control room. "Here isn't actually the easiest place to mix," he admits. "I have, before, put up duvets hanging off mic stands and so on, which helps a lot, and I check things on headphones. I'm quite used to the room, because it's quite live, and the reason the foam is there is because it's just the right amount to make drums sound nice. I did a bit of recording orchestral stuff, sending it back out through PA cabs and re-recording it — this room just glues everything together in a nice way. A good rimshot on the snare drum makes this room sound really nice. The way it projects down the room is just perfect."

Some people would automatically expect to use a dead-sounding recording area for vocals, but Jim is happy to exploit the sound of his room. "I don't put up any trapping or anything, it's just a BLUE Kiwi mic in omni mode, so I get the whole room. I ride the levels fairly early on, and if you do that early, you can get the rest of the instruments to fit around the vocal. When there's loud bits, if you record in omni mode, the room really comes out at you. I'm quite a fan of that swamped-in-reverb sound — all the strings are like that — but for vocals, there's a UAD1 plug-in called *Reflection Engine*, which is

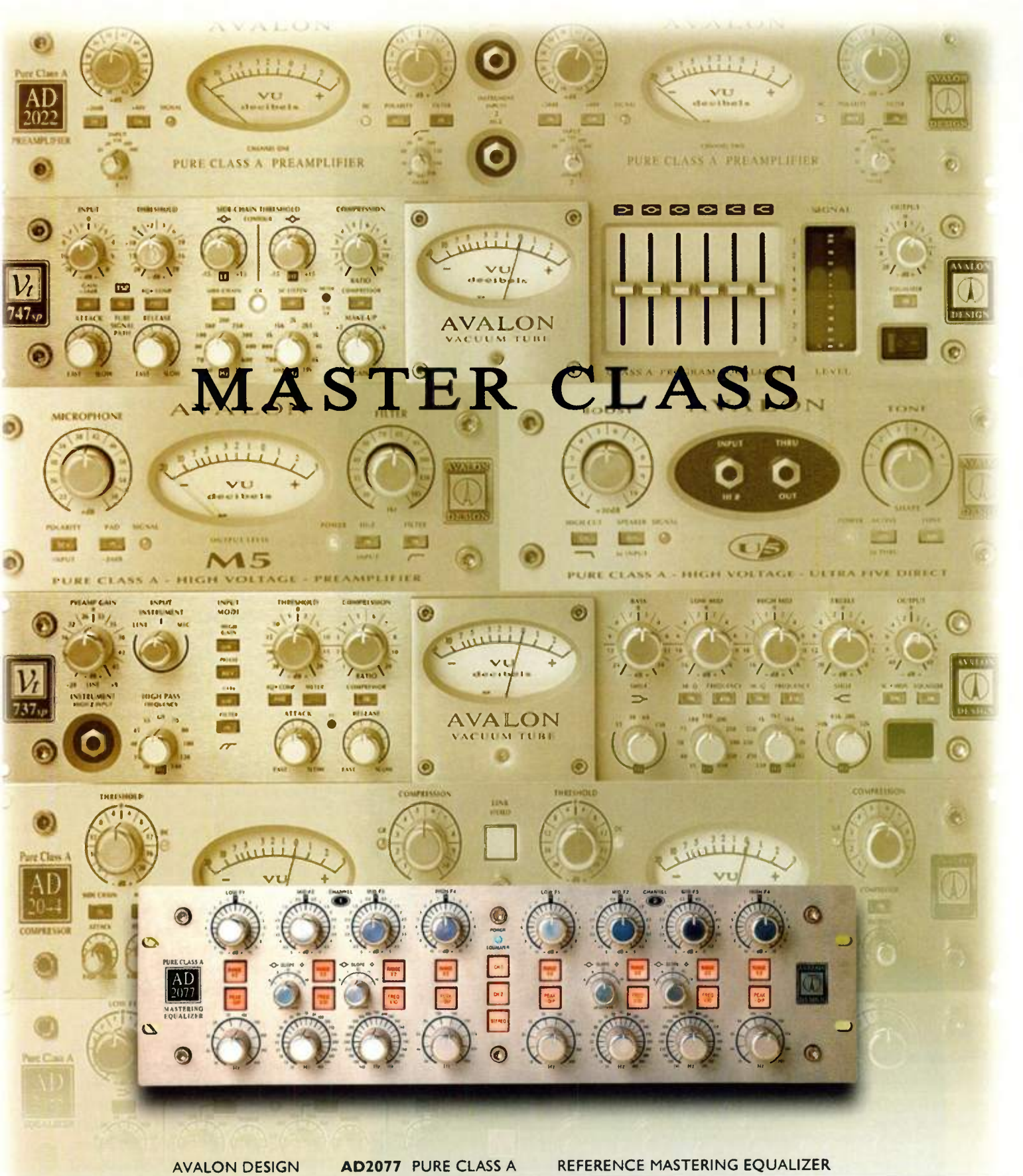
Tuning In

One of the many interesting features of Jim Moray's music is his guitar playing, which combines the sort of sound palette you'd expect in modern rock music with chord patterns and tunings drawn from folk artists like Dick Gaughan. "This album has been a return to standard tuning, a bit, but I still used about 10 different tunings for different things," he says. "The folk style of guitar playing has always been about alternate tunings, so I grew up trying to emulate that, and because it was easier, I transferred over to electric guitar, still taking the tunings.

"I change strings a lot, because if you get the gauges of strings right for alternate tunings it just sounds a million times better. What you're trying to do is keep the string tension the same as it would be for a standard set, generally, because you lose sustain and you lose a lot of tone when you have a floppy string going 'bong'. I try to match the tension as much as I can so all the bends come off exactly the same, and when you hit a chord there's no one string that dies away before the others. Also, if you palm-mute parts, some notes will ring out and some won't if you've got them all at different tensions.

"You end up changing a lot of strings, and I do a lot of changing halfway through a song, as well, so for live I use this [*Line 6*] Variax. It's a home-made job: Variax guts inside a Japanese Telecaster with an early '80s Japanese Strat neck. The Line 6 guitars are all a Fender scale length, so you can actually take the neck off and bolt on whatever neck you want. I'm still tidying it up — these are just the tops off pickups, cut with tin-snips and glued on! I tend to leave it on one pickup sound, so it's got a Telecaster sound, but I change all the presets to be different tunings. Before this, as I can't take this many guitars to gigs, I used to use a Roland VG8 system, which had more latency. I did have one of the acoustic Variaxes, but I didn't get on with it very well, I think mainly because I've got some very nice hand-made acoustic guitars, and I wanted it to sound like those!"

A Variax in disguise.



AVALON DESIGN

EURO AVALON
T +49 89 818 86 949
F +49 89 818 93 485
euroavalon@aol.com

AVALON USA
T +1 949 492 2000
F +1 949 492 4284

AVALON UK
T +44 (0) 207 231 9661
F +44 (0) 207 231 3002
avalon@asapeurope.com

www.avalondesign.com

almost a slapback thing, but more subtle than a delay. I also use tape delays, some from the Line 6 DL4 and some from plug-ins. I try to use delays on vocals, to keep them fairly up-front but with some detail on the edges. I don't usually double-track with my voice: my voice just doesn't seem to take to it. I double-track a lot of harmonies, though, to the extent of doing six or seven different harmonies with four or five takes of each.

"On this album I've gone really big on Soundfield miking. It was an eBay purchase which I'm still learning how to use; sometimes you almost don't need to do anything — you could just hold it in your hand and it would sound great — and sometimes it sounds abysmal. I've gone really big on Mid & Sides miking, so all the guitars were done like that, and all the drums. What I tend to do is use the same BLUE Kiwi or a TLM103, at head height about four or five feet in front of the kit, and then I've got a little Beyer ribbon mic. I got a hypercardioid ribbon and a figure-of-eight ribbon as a set, but for some reason it works a lot better with non-matching mics, so I use the ribbon as the figure-of-eight for the Sides,

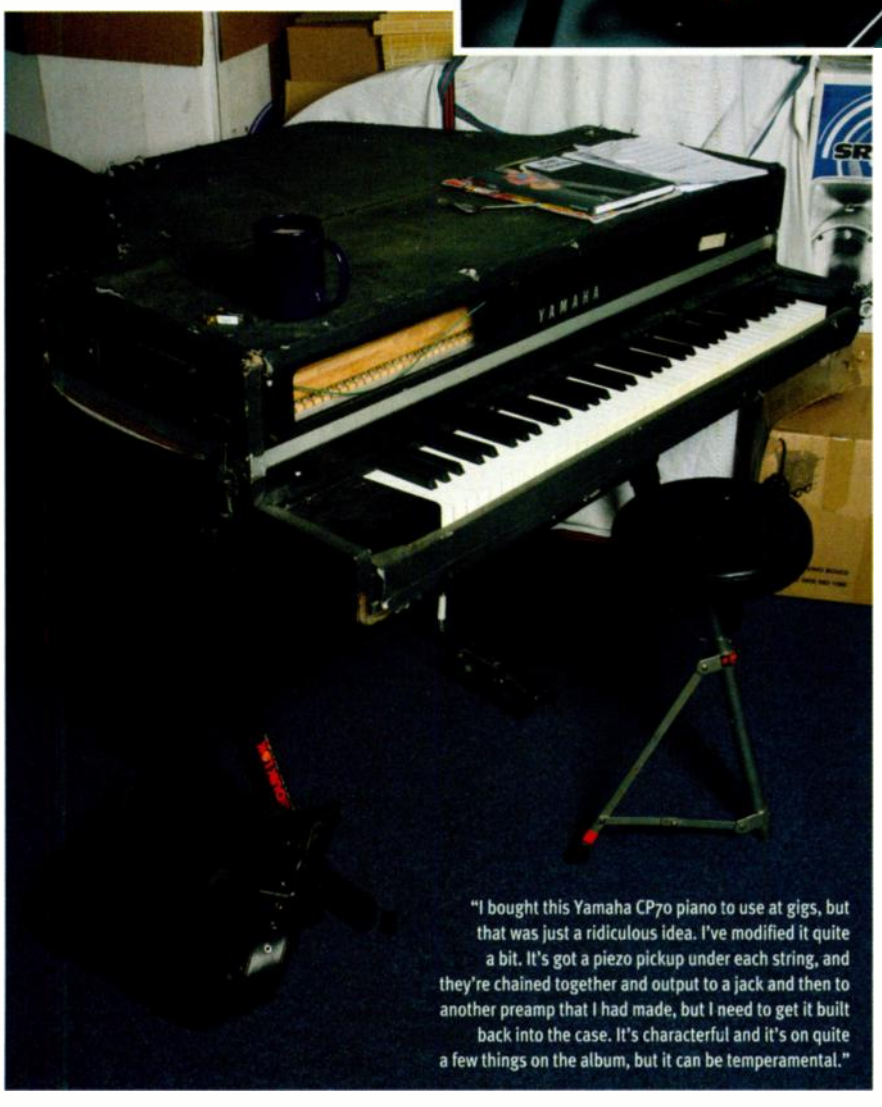
"This is a Fender electric mandolin from 1966. I think they made them and gave them away as a set with Strats from 1966. I tune this to my own tuning. Instead of being in fifths, the bottom three strings are in fifths and the top is a fourth. It's quite good for overdubs, little twinkly things."

and then a mono mic which I move about to get the best sound, and that's my main Side mic. The mono mic goes through the Liquid Channel, using something quite distorted and sucky — one of the Universal Audio channels, or driving one of the Neve simulations really hard, so you get it not distorted, but quite excited and crushed. Then I roll a lot of the bass off the Sides, and they go quite high in the mix, because with M&S your width control is the Sides level. I have quite a lot of those in the mix, but they just tend to get the cymbals and the top end of the snare; that gives the width, and then you've got this massive, thick thing down the middle. Generally all I have in the mix is the Mid & Sides mics, the kick mic and a bit of top snare mic. I hit drums quite hard when I play, though, and sometimes it can get a bit overpowering if you use just the top mic — you get too much of that tubby thump and not enough 'crack' from the bottom, so I'll sometimes blend those.

"Then I use an Electrovoice RE20 inside the kick, just off centre, and I've got the same mics I use for drums live; they're Electrovoice ND408s with swivelling heads. I have these rim-mounted. We did a tour with films, and I wanted the drums to be quite low-profile so that we didn't get a shadow on the screen. Instead of overheads we kind of had 'underheads', clipped on the cymbal stands facing up. You tend to get more cymbal than drum that way. It's nice and crisp."

Jim's other mic placements are also often unconventional. "I tend to use a Sennheiser E609 on [*electric guitar*] cabs, hard against the grille, or the EV ones again, or possibly an SM57. SM57s sound really nice, but I just tend to use other things — I don't know why. If you've just got a three-piece guitar band you can quite often stick an extra mic somewhere, say underneath the ride cymbal, to open up at a certain point, like a chorus or a middle eight or something, and then bring down again, and it makes the drums breathe a bit. I've got a couple of '57s that I tend to put in random places for things like that.

"Acoustic guitars I do either with the Soundfield or with the Mid & Sides arrangement. I tend to mic them the length of a string away from the mid-point — quite far off. When I went into the BBC for the first time, they miked my guitar in this particular way that sounded amazing, with a U87 about four feet away, pointing at the bridge, and



"I bought this Yamaha CP70 piano to use at gigs, but that was just a ridiculous idea. I've modified it quite a bit. It's got a piezo pickup under each string, and they're chained together and output to a jack and then to another preamp that I had made, but I need to get it built back into the case. It's characterful and it's on quite a few things on the album, but it can be temperamental."

another at right angles to it, looking down the neck at the nut. Those, panned 25 percent left and right, sounded really nice. The guitars on a song called 'Barbara Allen' were done like that, tracked up four times, with every other take panned the other way, so you've got two right-handed guitarists and two left-handed guitarists playing it, and it makes this big, solid, happy, thick, bright sound."

On Your Own

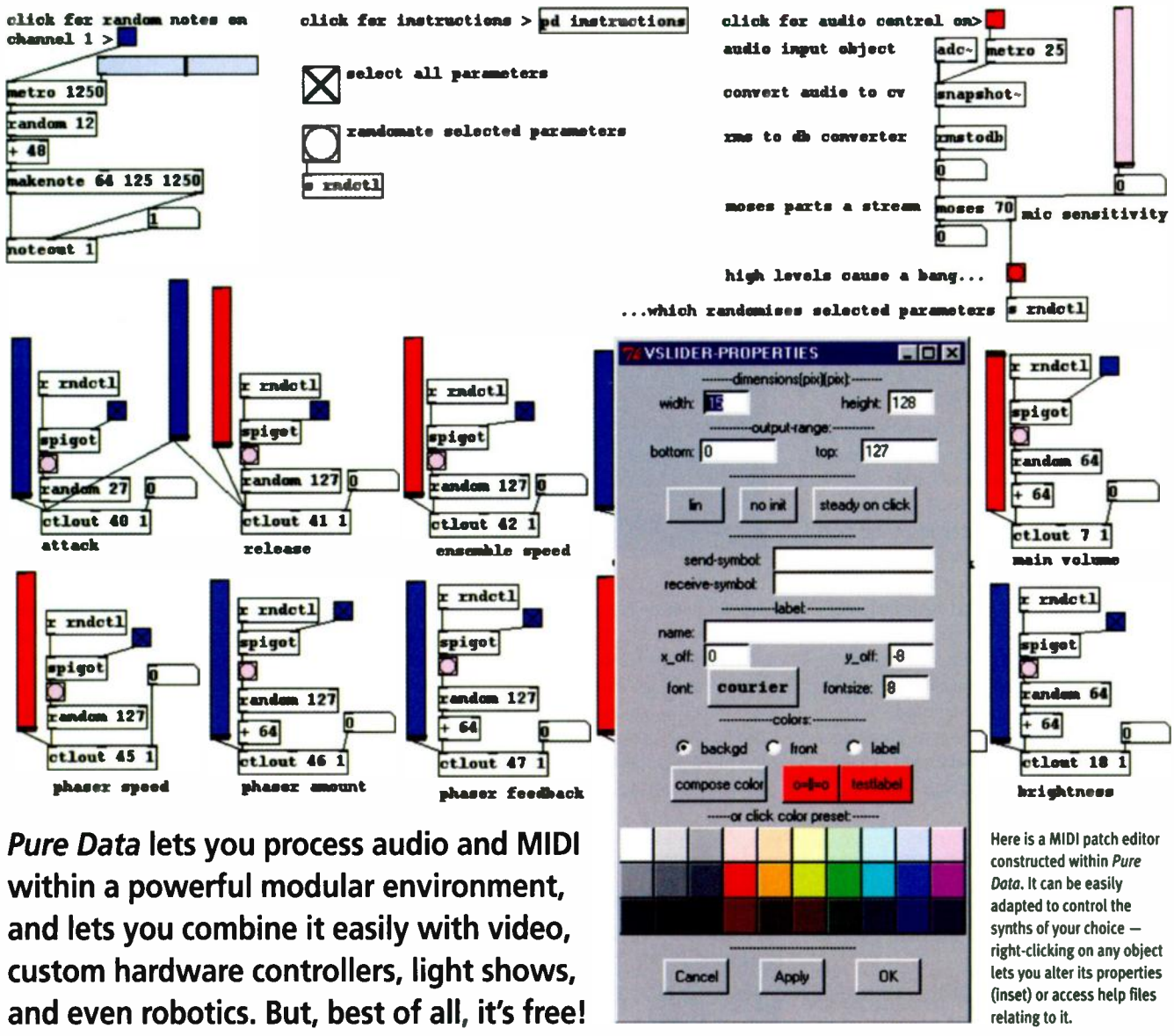
So far, Jim Moray's musical career has been a one-man show, at least as far as recording is concerned, but he's painfully aware of the limitations that brings. "To be honest, I think saying you produce yourself is a bit of a misnomer. I don't really think you can produce yourself. Production is buoying people up, setting the mood. It's making everyone feel at ease, so that it just comes out and you haven't had to do anything, but you can't really do that if you're the artist as well. I think 90 percent of what I do when I'm producing other people is talking to them, making them cups of tea, making them feel comfortable, and picking the right moment to say 'Do you want to do some vocals now?' That's when you get good stuff, and that's possibly the thing I'm a bit hesitant about on my own records. They don't always have the greatest performances, because no-one coaxes that out of me. I had some help with vocals on this one, from a guy called Grant Showbiz, but in general I engineer my own stuff and make the decisions. It's a lot easier that way, but I wish I had somebody to get the sound, so I could concentrate on singing.

"Next time, I'll hire a really good engineer for eight weeks and spend eight weeks making the record. This one's taken me 18 months because I've recorded, thrown away, recorded more bits, thrown them away, had a tantrum and erased the disk, and you never quite know when it's finished. The last record was mixed by Simon Emmerson and Mass in London, and I was there mixing it with them, but this time I spent so long and so much money making the album that when it was time to mix, I couldn't afford to get anyone else to do it. But I had the gear, and I thought 'Well, I guess I've got the time to teach myself to do this,' so this whole album's been a bit of a learning curve, engineering-wise. I think I'm quite pleased with the work I've done." SOS

Jim's TLA M4 tube mixer is used in conjunction with mixing and automation in *Logic*. "I use it for mixdown, but also quite a lot for sends. I like to print effects so that you can play them in real time — sending the signal out to loads of delays and turning knobs in real time — and quite often I do rides by sending out the signal, doing a ride of a vocal or whatever through the desk, and then printing it back into the computer. I like the idea of a mix being a performance, but I also like total recall, and that seems the best of both worlds."

Jim Moray is playing at the ICA in London on 28th July.

Free Agent An Introduction To *Pure Data*



Here is a MIDI patch editor constructed within *Pure Data*. It can be easily adapted to control the synths of your choice — right-clicking on any object lets you alter its properties (inset) or access help files relating to it.

Pure Data lets you process audio and MIDI within a powerful modular environment, and lets you combine it easily with video, custom hardware controllers, light shows, and even robotics. But, best of all, it's free!

Adam Armfield

There is a wealth of free audio and MIDI utilities on-line which can be edited and customised easily. They are designed in a free, easy-to-use graphical programming environment called *Pure Data* and can be used to make almost any sound & MIDI tool you can imagine. As well as audio and MIDI, *Pure Data* includes functions (called objects, or externals) for generating 3D graphics and video, and can also control external hardware like stage lighting or robotics. In effect *Pure Data* can use any kind of input (audio, MIDI, video, sensors, and so on) to control any kind of output, with lots of processing in the middle. So if you yearn for an electromagnetic catapult that flings jelly at the audience every time you rattle your tambourine, or a USB incense burner which radiates progressively stronger smells as the temperature in the church hall rises, *Pure Data* can help you achieve this!

Connecting Hardware & Software Using *Pure Data*

Pure Data really shines in combination with other software. For example, I often use it in conjunction with a sequencer, randomising all the settings on my soft synths at the push of a button. I'm also into generative composition, so I like to amuse myself by building random note and rhythm generators, using these to control an external sequencer and VST instruments. Because *Pure Data* can transmit SysEx messages, hardware devices like drum machines can be controlled in unusual ways, for instance by sending messages to restart the pattern playing at arbitrary intervals, allowing construction of complex rhythms.

There are several free programmes which allow you to route MIDI data between applications. For windows there's *MIDIyoke*

Getting Started With *Pure Data*

Pure Data runs on Windows, Linux, and Mac OS X. It also runs on IPods, PDAs, and other exotic platforms such as Silicon Graphics machines. The system requirements are very relaxed, but older systems will be slower — if your machine isn't fast enough for real-time work, you can still use it as a MIDI synth editor and for other tasks which don't depend on accurate timing.

Pure Data is mainly developed on Linux, so this version is usually the best. Readers with older Windows PCs can use a bootable Linux CD like *Dyne:bolic* to run the Linux version of *Pure Data* on their machine without affecting their Windows installation, creating audio files in Linux before transferring them into their sequencer of choice for editing. *Dyne:bolic* loads Linux straight from the CD drive without installing it, leaving the data on the hard drive untouched. (Just thought I'd repeat that!) *Dyne:bolic* will run on Pentium computers with 64MB RAM. If you have any old PCs knocking about, these can be pressed into service as stand-alone synths or noise-generating devices.

When it comes to learning about *Pure Data*, it's worth realising that it's functionally very close to Cycling 74's *Max/MSP*. In fact, the two programs are so similar that tutorials for one can give you a good working knowledge of the other! However, *Pure Data* itself includes many example files (available from the Help menu) which can be edited, butchered, and generally stolen from to help construct your own patches.

When a file is first loaded, *Pure Data* is in Run mode, which means that the patch can be played rather than edited — the sliders and other graphical user interface (GUI) elements work. To edit a patch, you need to select Edit mode from the Edit menu, whereupon the cursor turns into a little hand. Objects and GUI elements are the basic building blocks of a patch, and they are connected together with patch cords. Data flows from the outlets at the bottom of an object, down the patch cords, and into the inlet at the top of the next object. Instead of selecting objects by name from a list, as is the way in *Max*, *Pure Data* prefers that you use the Put menu to place a blank object on screen, then type the name of the specific object you want to create inside it.

(for Windows 2000 and Windows XP) and Hubi's *Loopback* for Windows 98, while Mac OS X has this functionality built into Core MIDI. The Linux equivalent to Core Audio/MIDI is called *Jack*. Open Sound Control (OSC) is another MIDI-like protocol which is used by *Pure Data* and other computer music software like *Max/MSP*, *Csound*, and *Supercollider*, and also by some exotic control hardware (like the Jazz Mutant Lemur multi-input touchscreen).

Eyesweb, a gestural recognition package, also communicates with *Pure Data* via OSC. *Eyesweb* reads the input from a camera (this can be a low-cost webcam) and outputs control signals in response. *Eyesweb* is programmed in a graphical way like *Pure Data*, and is often used for art installations and by dance companies, and in other situations where a mass of stray cables would get in the way. *Eyesweb* is also free software, though it is only available for Windows. Another option is the *Pure Data* graphics library *Gem*, which includes basic visual recognition objects and is very robust on Mac OS X. Stage lighting can be controlled using a protocol called DMX. MIDI-to-DMX converters are available which enable *Pure Data* (or other MIDI software) to control a lighting rig.

There are even *Pure Data* objects which can perform network communication, and this means that *Pure Data* can communicate with any other software with networking capabilities. One example of this is the NetPd project, which enables people to collaborate over a network or via the Internet, each playing the same patch, transmitting control data to the other 'band members' so that each hears the same thing at (roughly) the same time. Pdradio is another interesting application, an on-line radio station which uses *Pure Data* to play audio files — the Pdradio web site (http://pdradio.iem.at) gives instructions on how to build your own Internet radio station on a Linux web server.

Creating Custom Hardware Controllers Using *Pure Data*

Several companies sell pre-built circuitry which can form the core of a custom MIDI controller — often all the user needs to do is add whichever knobs, buttons, and sensors they want to the circuit and mount it in a suitable box. Another approach to building custom devices is to adapt USB game controllers like joysticks, pedals, and dance mats, adding one's own controls to the circuit in place of the existing ones. These devices have become incredibly cheap in recent years. Many artists who used to build hardware from scratch now work with game controllers exclusively, because they are so cheap, and much of the complicated work is already done. A *Pure Data* object called Hid communicates with these gadgets and lets you incorporate them into your music system.

Example: MIDI Patch Editor & Randomiser

To give a better idea of the way *Pure Data* works, and what it can do, I've

Hear What *Pure Data* Can Do

To try to give some idea of the range of sounds available to you through Pure Data's processing and synthesis capabilities, I've created some audio demonstration files which can be downloaded from www.soundonsound.com/sos/july06:

• /audio/Additive.mp3
An eight-voice additive synthesis patch.

• /audio/ChoppedCasio.mp3
Quarter-note segments of a Casio drum loop played in random order.

• /audio/GranularCasio.mp3
Here the same Casio drum loop is subjected to granular alterations using *Pd-extended*.

• /audio/GranularVoices.mp3
A vocal sample altered with a granular synthesis patch from *Pd-extended*. Note the wide variety of sounds that can be achieved with just this patch.

• /audio/WaveshapeSequence.mp3
This is the wavetable synth patch featured in the article. As the sequence plays, the oscillator waveshape is altered by drawing on it.

• /audio/FMPatch.mp3
This is the output from a simple frequency-modulation synthesis patch. The frequency of the modulating oscillator is swept to produce rich timbres.

• /audio/SawtoothMultiplier.wav
Positive and negative sawtooth waves multiplied by each other.

• /audio/SampleAndHold.wav
An analogue-style melodic sample-and-hold patch.

• /audio/303Morph.wav
Randomising the settings on a software Roland TB303 emulation. Another application is being used as a VST host (and to play the sequence), whilst *Pure Data* is used to alter the settings. The patch used here is very similar to the MIDI patch editor featured in the article.

• /audio/WavetableKlingons.wav
Here one wavetable is modulating the pitch of another which provides the oscillator waveform. Both wavetables can be drawn on with the mouse to alter their waveshapes and the pitch of the sequence.

created a couple of *Pure Data* demonstration patches, the first of which is a MIDI synth editor. (You can download this from www.soundonsound.com/sos/july06/patches/SynthEditor.pd.) It can be used to edit VST instruments as well as hardware instruments, and would come in handy for controlling a rackmount synth with a somewhat minimal front panel. Because of *Pure Data*'s totally open-ended nature, you can build a custom editor for your setup. For instance, if you typically patch your VST synth through a number of effects, *Pure Data* allows you to edit all the synth and effects parameters at once — just place some sliders on screen and configure them to send parameters on the MIDI channels/ports of your choice.

More interestingly, you can create relationships between parameters. You might want to limit the level of resonance on a filter (to avoid damaging your speakers), or to only limit the resonance when the cut-off is above a certain level. Or how about setting the rate of one LFO to remain exactly half the rate of another? All this can be done with basic maths in *Pure Data*.

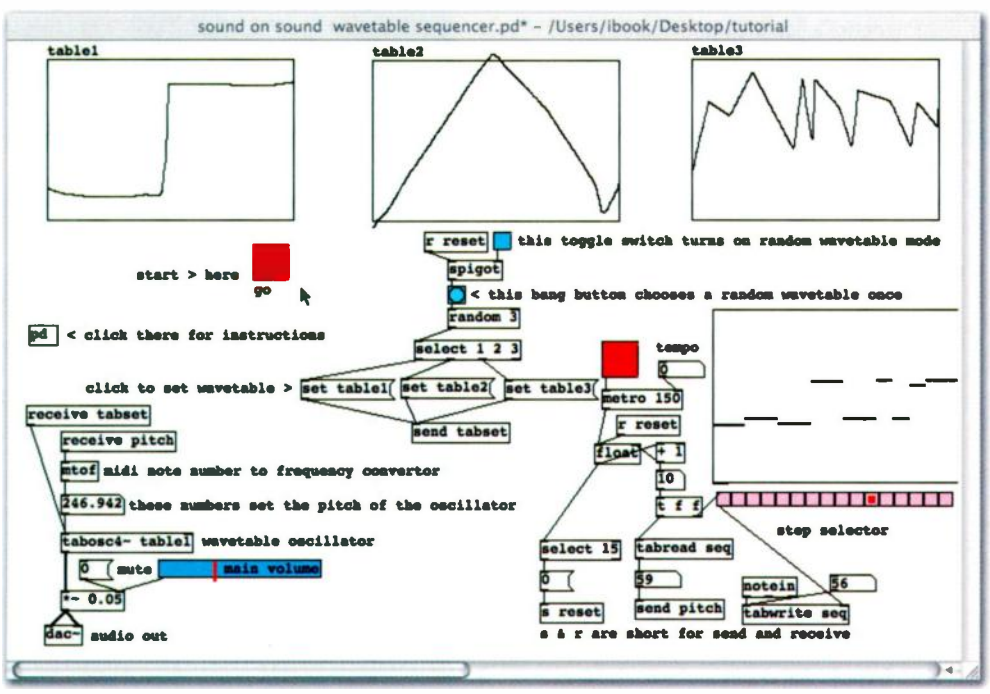
"If you yearn for an electromagnetic catapult that flings jelly at the audience every time you rattle your tambourine, *Pure Data* can help!"

In my example I have set up the editor to control a free VST instrument called *Cheeze Machine*. I've decided that I want the attack and release values to be the same, so I have created one slider that controls both parameters at once. I have also created a button to instantly randomise all parameters, but I have chosen to limit the range of this randomisation on certain parameters in order to control the results a little — totally random settings can sometimes produce disappointing results, for instance when the envelope's attack setting ends up so long that notes don't sound properly.

MIDI changes can easily be triggered from an audio signal, and there's a simple example of this in my patch — a mic input which will trigger parameter randomisation whenever the audio signal reaches above a certain threshold level. However, there's lots more scope for more complex creative control, especially because of *Pure Data*'s nifty pitch-detection object.

Example: Wavetable Synthesiser

The following patch gives an idea of what you can do with some of *Pure Data*'s audio capabilities. (It's available for download at www.soundonsound.com/sos/july06/patches/WavetableSequencer.pd.) This patch is a wavetable synthesizer with a built-in 16-step sequencer. What makes this example interesting is that the oscillator shapes can be altered by clicking and dragging over them with the mouse. The oscillator waveforms are stored in Arrays (the three boxes along the top of the screen



This wavetable synthesizer shows something of what *Pure Data* can do when it comes to audio processing, and it also includes a built-in step sequencer. The sequence can be edited with a MIDI keyboard, while wavetables can be drawn on with a mouse.

Are you Xperienced?

E-MU's new Xboard 61 USB/MIDI Performance Controller delivers an unparalleled hands-on experience. From the superior response of its keys to a host of real-time control knobs and buttons that offer instant control over filter tweaks, patch/program changes and more, the Xboard 61 has been designed for players that demand professional features and performance in a portable package. The Xboard 61 also includes an integrated full version of E-MU's award-winning Proteus X Desktop Sound Module with over 3GB of sounds (featuring a custom bank of E-MU's finest classic keyboard and synth presets), as well as Xboard Control™ software to effortlessly create custom templates that put all of your favorite hardware and software instruments at your fingertips.

Xboard 25

Xboard 49

Get your hands on the Xboard 61 at your local E-MU Dealer and get Xperienced.

Mac

Works with Windows 2000 XP, x64

ProTouch Full-size keys with Aftertouch

Xboard CONTROL Create custom templates

Software Sound Module ProteusX Over 3GB of Sounds Included

CREATIVE PROFESSIONAL

Available at your local E-MU Dealer
www.emu.com

E-MU

Toll-free
0800•9012168

Web Resources

• http://pure-data.info
The main *Pure Data* site. You can download the program from here, and I'd recommend going for the package called *Pd-extended*, because this includes lots of additional processing objects. The Community area offers loads of patch examples to help you get going.
• http://drpichon.free.fr/pmpd
Here you can find the Pmpd collection of objects for *Pure Data*, which offer a variety of physical modelling processes that can be applied to audio processing.
• http://noisybox.net/computers/pd
A selection of useful little *Pure Data* patches from Jason Plumbs.
• www.mortmain.com/pd.html
Useful *Pure Data*-related links page.
• www.loopit.org/jamma_info.htm
The home of the *Jamma* looping sampler, created using *Pure Data*.
• http://ipodlinux.org/PdPod
Pure Data ported to run on Apple's IPod!
• http://dynebolic.org
This CD-based Linux package includes *Pure Data* and will run on old PCs from Pentium models onwards.
• http://cycling74.com
Cycling 74 sell several sensor-to-MIDI systems and other hardware controllers, along with their own graphical programming environment *Max/MSP*.
• http://infusionsystems.com
This is the home of I-Cube X, a pre-assembled sensor-to-MIDI conversion system which is particularly easy to use. However, it is also more expensive than other more DIY options.
• http://eroktronix.com
Here you can get a fairly low-cost sensor-to-MIDI system, but you need to know some electronics to wire things up.
• www.doepfer.de
Doepfer also sell pre-assembled circuits for making custom MIDI controllers.
• www.sparkfun.com
Various sensors, circuit modules, and robotics gear which can be hooked up and used with *Pure Data*.
• www.cnmat.berkeley.edu/OpenSoundControl
Open Sound Control is a newer MIDI-type protocol which allows *Pure Data* to communicate directly with applications such as *Max/MSP*, *Csound*, and *Supercollider*. It also interfaces with the powerful Jazz Mutant Lemur touchscreen hardware controller.
• www.eyesweb.org
This free Windows-only visual recognition software which can communicate with *Pure Data* using Open Sound Control.
• www.netpd.org
The home page of the Netpd project, which encourages musicians to jam on-line in real time using *Pure Data*.
• http://pdradio.iem.at/howto
Instructions for setting up an on-line radio station on a Linux web server running *Pure Data*.

in my example), and samples in a variety of formats can be stored within *Pure Data* in this way.

The sequencer section of my example is at the bottom right-hand side of the picture. A clock object called Metro (short for metronome, though this software was originally written in Paris!) drives a counter, which cycles from one to 16 repeatedly. The counter tells an object called Tabread which step of the sequence it should read. The tabread object outputs a MIDI note number at each step, and this number determines the pitch of the wavetable oscillator. The sequence can be edited using a MIDI keyboard (via the Notein and Tabwrite objects), or by drawing directly on the graph at the bottom right of the screen.

If you wanted to, you could adapt this patch so that the oscillator waveform could be drawn using a knob or slider, or by running your fingers up and down a MIDI keyboard (or just up for more of a sawtooth wave). An input device like a Korg Kaoss Pad or joystick could also be used for this purpose.

The Future Of Pure Data

Pure Data is being actively developed, you're free to suggest changes, and each new revision generally brings notable improvements. However, there's no need to wait for someone else to implement any improvements you're after, because *Pure Data* is totally open source, which means that you are free to adapt it and to get involved in it's development, although you do have to be able to program in the language C to do it.

Hopefully this article has given you a taste of what *Pure Data* can do. It offers a vast range of possibilities for producing sonic and visual arts, and combines media in exciting ways. And given that it's free, there's little reason not to try it out for size! SOS

Big Fish Audio Chill

Multi-format

'Like a deep massage,' the blurb begins, 'these laid-back grooves penetrate your eardrums and inspire your soul'. I know from my own experience writing for this column how difficult it is to write descriptively about samples, and how easily you can end up with mangled metaphors. Even so, a deep massage that penetrates your eardrums really does sound rather painful. Fortunately, this new sample library from Big Fish Audio is a much gentler and more enjoyable experience.

Citing the music of Portishead and Massive Attack as stylistic points of reference, *Chill* promises to draw from jazz, hip-hop, dub, funk, soul, ambient, and pop to 'create a whole new downtempo experience'. Although the words 'trip hop' don't appear anywhere on the cover or in the Readme file, but they do appear in this review...

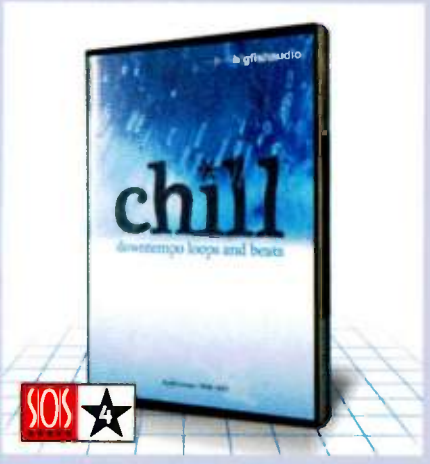
Delivered on a single DVD-ROM, *Chill* duplicates the same basic content in three different formats: WAV, Apple Loops, and REX 2 files. In each case you get 34 different construction kits, and an Extras folder containing a good selection of other bits and pieces (drums, some nice jazzy flutes, guitars, and so on). The samples are recorded at 24-bit, 44.1kHz resolution.

The construction-kit tempos cover 60-110bpm, and are mostly in minor keys. Each kit includes a complete mix of a short composition, together with its various component parts presented in isolation. Some also provide a Hits folder containing one or more one-shot samples (generally drum or percussion hits) suitable for programming extra fills or ornamentation.

The material on offer here is impressive. The construction kits are all quite usable, and the better ones are very convincing indeed. The various producers (five different names are credited) demonstrate not only technical ability, but also creditable musical judgement. The kits have been assembled with taste and restraint, steering well clear of needless pyrotechnics and pointless twiddly bits. The different components sit well together, and the arrangements work.

There's a distinct jazz flavour to many of these samples. Some excellent Rhodes and Wurlitzer keyboard parts are in evidence, often complemented by accomplished, but understated guitar licks. The effect is appropriately 'smooth', but without being over-polished or sterile. There are a few darker kits thrown in as well, but the overall feel is actually quite upbeat and cheerful.

The breaks and beats are all-important for this style of music, and the drum programming and production are handled here with flair. Good use is made of compression and EQ, and



the sounds range from solid, assertive, and upfront, to muffled, degraded, and lo-fi. A couple of the kits feature glitchy fills and flourishes, but for the most part it's kept simple and effective.

Chill would be a valuable sample library for anyone with an interest in down-tempo, beat-driven music, and it could also make itself useful outside of its appointed genre, potentially offering something to pop productions of every kind. *Paul Sellars*

Apple Loops, REX, and WAV DVD-ROM, £55 including VAT.
Time + Space +44(0)1837 55200
www.timespace.com www.bigfishaudio.com

Best Service Chris Hein Horns Volume 1

Kontakt Instrument

Veteran producer and sound designer Christian Hein's sampling work has largely centred round orchestral sounds, but his latest venture features pop horns: alto and tenor saxes, trumpet, trombone, and a virtual trumpet section made up of solo trumpet samples. The saxophones are played by Marc Leymann, Andy Haderer plays trumpet, and the 'golden handbrake' is manipulated by Ludwig Nuss, creator of the near-legendary CD *Horn Players Can't Eat Garlic*. (I kid you not!)

To create an effective sampled pop horn section you have to get the trumpet right, and I'm glad to say Chris Hein has got the trumpet right. Its staccato stabs are short, precise, tight, bright, and punchy, and when played chordally their timing is bang on the money. In a word: bap! Uninhibited jazz mannerisms like 'doits' (a stab followed by a hysterical rising pitch), falls (supplied in five different lengths), and growls abound, and a set of nicely sleazy 1950's big-band-style 'shakes' evoke images of a red-faced, cross-eyed man with bulging cheeks who's about to burst a blood vessel.

The trumpet long notes (all looped) have a good positive attack, 10 dynamic layers, and a nice, subtle vibrato option which can be accessed via the mod wheel — the only caveat is that if you move the wheel while sustaining a note, you're likely to hear a glitch when the two sample sets switch over. On the long sustained no-vibrato samples, pressing the sustain pedal raises the instrument's pitch by 15 cents, the idea being that you can add artificial vibrato by rhythmically tapping the pedal. Bad idea — this eccentric piece of programming not only produces an unconvincing, synthetic vibrato, but also prevents users from elongating notes with the sustain pedal. That apart, this is one of the best sampled pop trumpets I've heard. The virtual trumpet section sounds pretty triumphal too.

Great efforts have been made to keep the instruments' playing styles consistent: the bright-sounding alto and tenor saxes mimic the trumpet's ultra-tight short staccatos and deliver falls, chromatic pickups and run downs, grace notes, swells, and *fp* crescendos with flair and poise. Marc Leymann also whips out a large set of inventive, exuberant jazzy licks and improvised phrases on both saxes. Although stringing them together takes a bit of work, a few of these riffs will give tracks an instant 'live player' feel.

Mr Nuss's trombone sounds big, fat, and commanding, and the instrument's wide tonal variations are nicely reproduced by the multi-dynamic samples. Its pitch slides are played with relish, which one hopes compensates for the prohibited garlic. Taken as a whole, these horns are very versatile, offering all the styles you need for pop, rock, R&B, or jazz big-band arrangements. Although the combination of a dry recording acoustic and mono samples militates against a lush listening experience, the musical content and execution are second to none — add some classy reverb, and these horns will shine! *Dave Stewart*

***Kontakt* Instrument, £232 including VAT.**
Time + Space +44(0)1837 55200
www.timespace.com www.bestservice.de

M-Audio Premium Electric Pianos

Multi-format

There are plenty of multi-instrument libraries on the market, but if you want a sampled instrument that will meet professional standards, it's generally better to buy a specialist title. M-Audio evidently share this view, and their Premium Instruments range of sample DVDs keeps the focus on individual instruments and offers in-depth sampled renditions of some real classics in every sampler format known to man (except *Gigastudio*).

Having been a Fender Rhodes player since the days when Abba ruled the earth (ah ha!), M-Audio's *Premium Electric Pianos* title was the first to catch my eye. The set contains two Rhodes electric pianos: a Stage 73 model from the late '70s, and a Suitcase 73 used in Avatar studios (formerly the Power Station) in America. Both electric pianos were recorded by producer/remixer Chris Griffin, who has worked with Madonna and David Brent's favourite band, The Corrs.

I always loved the bell-like attack of my Stage 73 Rhodes, and on Chris Griffin's sampled version the 'sparkly tines' effect can be heard most clearly in the octave starting on 'C' above middle 'C' — to bring out this quality in other parts of the piano's range, you'd need to EQ it or run it through an amp. The Rhodes' tone is beautifully pure and transparent, but if you want a more processed sound, one program adds a suitably vintage stereo phasing effect reminiscent of the MXR90 effects pedal.

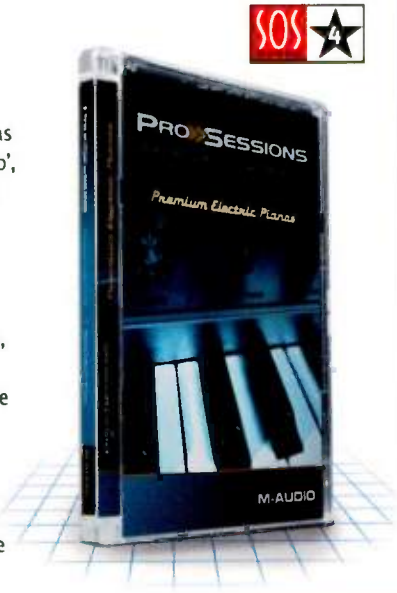
The 'Avatar' Suitcase Rhodes has been souped up by 'Dyno-My-Piano', a set of modifications which made Rhodes pianos sound more poky and aggressive. As a result, the Suitcase 73's attack is somewhat more prominent than that of the Stage 73, but the trademark warm, intimate, and funky sound of the instrument is still very evident. The Avatar Rhodes' samples were recorded in three different ways labelled DI, Amp, and Room. The Amp option adds a nice presence which further accentuates the note attack, while the Room samples introduce an agreeable stereo large-room ambience.

Both Rhodes pianos were sampled at four dynamics, using between 16 and 27 samples in each dynamic to cover their 73-note range. A set of one-dynamic release samples are also provided — these work well in most situations, but if you hold down the sustain pedal and play a lot of notes, they make a sharp, rather obtrusive click when the pedal is lifted. As the trend nowadays is to sample pianos at 12 or more dynamics chromatically or on every white note, I felt it was a shame that a similarly intensive approach wasn't used on these two Fender Rhodes pianos — however, I have to say that they are both supremely playable, highly responsive, and well-balanced across their entire range.

I left the best till last — if you load either piano and push up the mod wheel, a glorious, swirling stereo tremolo effect fills the speakers. I must admit I've no idea how the effect is achieved, but this gorgeous noise is the icing on the cake. *Dave Stewart*

***EXS24*, *Halion*, *Kontakt*, *Mach Five*, and *Reason NNXT* DVD-ROM, £79.99 including VAT.**

M-Audio UK +44 (0)1923 204010
www.maudio.co.uk
www.m-audio.com



Loopmasters Soulful House Sessions

Multi-format

This latest addition to the Loopmasters Origin series not only provide its loops in *Acid*ised WAV and REX 2 formats, but also includes some instrument patches for the *EXS24*, *Halion*, *Kontakt*, and *NNXT* samplers. In total, the CD-ROM contains some 670MB of sample data. The collection is dominated by more than 350 drum loops, organised into four tempo groups spanning 120-130bpm. The majority are full-kit loops, but subfolders of Live, Percussion, and Tops loops are also included. The latter contain the higher-pitched instruments of the drum kit — percussion, hi-hats, high toms, and some crisp snares, for example — and layer well with the full loops. Sonically, this is very much classic House, and the playing and programming seem well done.

The instrument loops include bass, Rhodes piano, guitar, synths, sound effects, and trumpet, as well as a selection of one-shot vocal samples (both male and female). There are not huge numbers of loops in any of these categories (for example 20 Rhodes loops and 30 guitar loops), but what is here is well played and recorded. As suggested by the library title, the mood of these is meant to be at the mellower end of the House spectrum, and this is particularly reflected in the fairly gentle Rhodes loops and the inclusion of the small selection of jazz-influenced trumpet loops.

The vocal samples are a bit of a mixed bag — the male samples are spoken or shouted, but there is nothing here that most producers couldn't easily replicate for themselves. The female samples include some spoken phrases that include a lot of heavy breathing, while the small number of sung phrases are very soulful in tone. These suit the mood of the musical loops well, but it is a shame that these one-shot files have no indication of the original key they were performed in, as this makes mixing them with the other material more difficult. The various instrument patches include some nice basses, a decent Rhodes, and a wide range of individual drum and percussion hits. All of these are available as sampler patches, providing some extra flexibility — live playing can easily be added to a composition based upon a selection of the loops in order to provide a little variety. Although these are very usable, don't expect miracles at this price point, because the patches are based on a few samples and do not include velocity layers.

There are, of course, a large number of House loop libraries on the market, although these do tend to be dominated by more full-on, four-on-the-floor styles. *Soulful House Sessions* doesn't bring anything massively new to the genre, but if you are after something with a somewhat mellower vibe, then at this price it's well worth an audition. *John Walden*

Audio CD and *Acid*ised WAV, *EXS24*, *Halion*, *Kontakt*, *Reason NNXT*, and REX 2 CD-ROM 2-disc set, £39.95 including VAT.

Time + Space +44(0)1837 55200 www.timespace.com
www.loopmasters.com

Forte

A series of exceptional Installation Speakers that combine crystal clear sound with rugged construction.

The Forte range is headed by the three way GS300 series which provides 120, 150 or 200W power options from an 8", 10" and 12" bass driver coupled to a 5" mid range and a 1" soft domed tweeter. Huge magnets provide the power to deliver uncompromising sound quality from substantial MDF bass reflex cabinets.

All speakers and 4000 other products are detailed in the NEW 256 page Studiospares catalogue out now! Available FREE on request.

Power levels between 50 and 100 watts are covered by the beefy BG series with tweeter or horn options. Impossible to beat for construction or audio quality. All models are described in detail in the latest Studiospares catalogue and at www.studiospares.com.

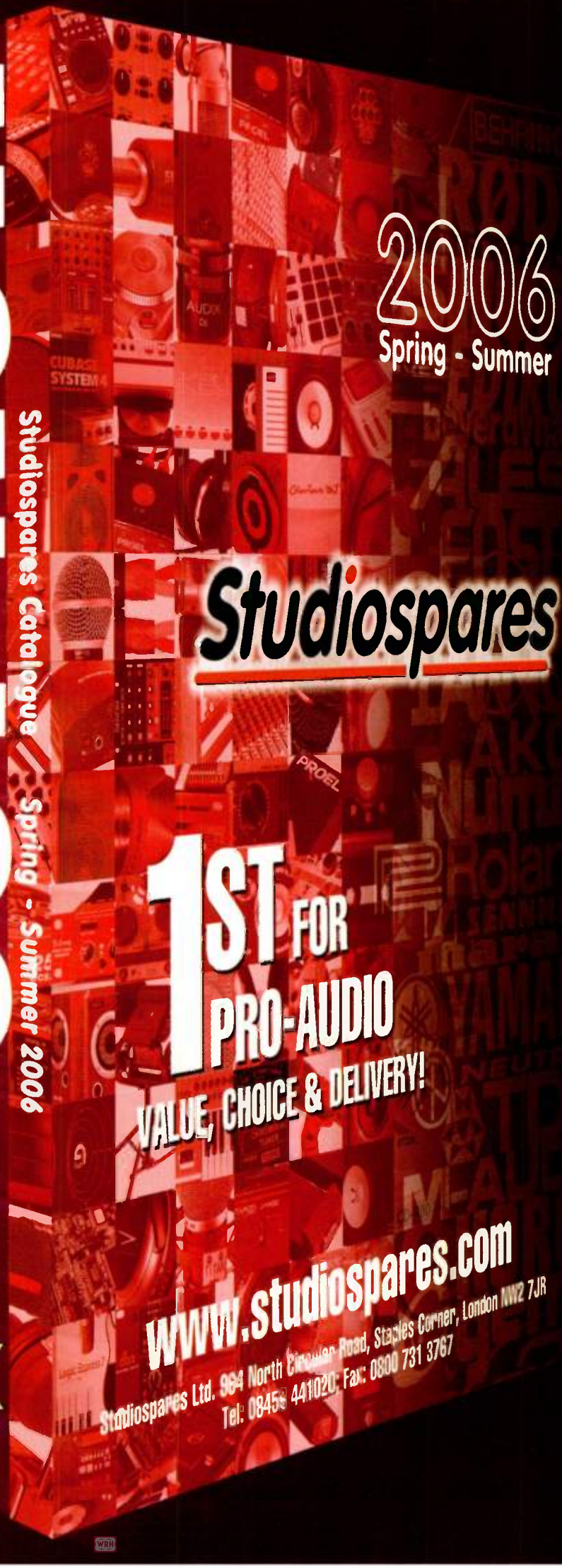
Studiospares
964 North Circular Road,
Staples Corner,
London, NW2 7JR

tel 08456 441020
fax 0800 731 3767
email sales@studiospares.com
web www.studiospares.com

Studiospares

OUT NOW

The new Spring/Summer edition of the Studiospares Catalogue is out now, available FREE on request!



Studiospares

964 North Circular Road,
Staples Corner, London, NW2 7JR
tel 08456 441020 fax 0800 731 3767
email sales@studiospares.com www.studiospares.com

Sonar offers several ways to mix, but is that a benefit or a source of confusion? Learn what the various mix modes do and you can choose the one that's right for your project.

Craig Anderton

The Track view.

Optimum Mixing In *Sonar*

One of *Sonar*'s interesting features is the ability to mix in three different ways: using the Track view, the Console view, or a combination of the two. Beneath the surface they have more similarities than differences, but the user interfaces and workflows are quite different. Consequently, it's important to know each option well enough to decide which one will work best for your application.

The easy answer takes as its premise that there are three basic elements to a project: tracking, editing and mixing. Any view works for tracking, but the Track view is optimised for editing because you can see not only track parameters (volume, pan, effects, and so on) but also the data within the tracks. Due to the track orientation, the 'channel strips' are horizontal, like the tracks.

The Track view's Inspector, which provides a Console-style mixer channel any time you need it.

However, the screen can get very busy, which is why the Console view, which has traditional vertical channel strips, can be a good choice for mixing. At the mix stage it's unlikely you'll be doing things like moving MIDI notes around very often, so you can hide the track data and configure the Console to cover the most important mixing-related parameters. Another consideration is that those raised on hardware mixers may be more comfortable with the Console view, as it resembles a more traditional mixing environment. Those who moved to *Sonar* from programs like *Acid* and *Vegas* might prefer the Track view, as its paradigm is similar to these programs.

In addition to providing these two options, though, *Sonar* also offers a combination of the two, in the form of the Track view Inspector (see screen, left). This shows up towards the left of the Track view and is like a single mixer channel from the Console view embedded in the Track view; it shows whichever track is selected. You can disable the Inspector to reclaim some screen space, but I find it handy to see when working in the Track view. (The ideal setup would be dual monitors, so you could see both views at once.) In any event, there are some significant differences between the views that may affect your preference, so let's investigate further.

Toolbars

The Console view has a toolbar that shows and hides different parts of the mixer (see the top screen on page 188). Each toolbar button is grey when hiding and green when showing. The following buttons are available:

- Wide/Narrow Channel Strip: This affects all the mixer channels. If you want to narrow or widen an individual channel, right-click to the right of the fader and choose Narrow or Wide Strip.

The Console view.

- Meters
- Track icon
- Input section
- EQ frequency-response plot
- EQ controls
- Effects
- Sends
- Mono/Solo/Record buttons
- Pan pot
- Fader
- Output

It's important to have this degree of control, not just to make sure the Console can actually fit on your screen, but to minimise clutter. For example, if you're set up to do a mix, odds are you won't need to see the input and output assignments.

The Sends button has three states: hidden (grey); show two sends (green); and show four sends (blue). If you have more sends inserted than can be viewed, small scroll-arrows appear in the send section's lower and upper right corners to provide scroll down and scroll up, respectively.

The Effects button also has three states: hidden (grey); show effects bin (green); and show effects bin plus four assignable ▶

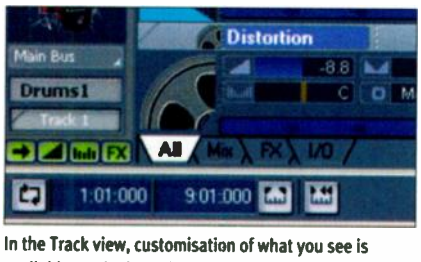
The Console view toolbar, where you can customise the look of the mixer by hiding or showing different elements.

parameter sliders (blue). Similarly to the sends, if you have more than four effects loaded into the effects bin, small scroll-arrows appear. Note that assignable effects sliders are available only in the Console view; there is no equivalent option in the Track view, although the Inspector channel strip can show the faders. No scroll bars appear in the Track view if you have lots of sends or effects inserted. Instead, viewing the available options depends on being able to alter the track height to show these additional controls.

The EQ button's three states are hide EQ (grey); show one band of EQ (green); and show four bands of EQ (blue). Even if only one band is shown, you can select which of the four bands you want to see.

The Track view takes a different approach, offering four tabs (see screen below), for four preset views, as follows:

- All: All parameters.
- Mix: For audio channels, parameters are Volume, Pan, Trim, Phase and Meters (if enabled); MIDI parameters are Volume, Pan, Velocity trim, Key offset, Time Offset, Scale and Root note.
- FX: For audio, shows FX bin, sends and mono/stereo button; MIDI includes the MIDI FX bin, chorus and reverb
- I/O: Shows audio ins and outs. MIDI has in and out but also Bank, Channel, Patch, Scale and Root note.



In the Track view, customisation of what you see is available partly through these four tabs, which provide access to four preset views.

I generally leave 'All' selected, although the 'FX' option is handy if you have a lot of sends happening — it 'de-clutters' the interface considerably.

Control Layout Differences

Tracks and busses seem similar in both views, but look more carefully. In the Track view, track audio goes through the Trim control and is then processed by any effects patched into the track. The audio then passes through the pan and volume faders, and finally to the designated buss or output. The actual look of the Console view reflects this signal flow, but in the Track view you'll notice that the input Trim control defaults to being below the main pan and level faders — the opposite of what you'd expect (conventional input to output flow goes top to bottom). Why is the Track view like this? Because as you reduce the track height, the Trim control disappears first, leaving the more important main pan and level faders still visible. Busses are structured in a similar way. In the Console view, buss input-level and pan controls head the channel strip. In the Track view, they're below the main level and pan faders.

Fader throw between the views is also different. If you're using an external hardware controller to wiggle the faders, the length of on-screen faders doesn't make any difference to you. But if you need to make fine volume changes with a mouse, the Console view and Inspector faders have a longer throw than the Track view faders. Generally, I find the Track view faders useful for setting approximate levels as rapidly as possible, but I switch over to the Console view if I need finer control.

Creating A Friendly Mix Environment

You really don't want to have to think too much when you mix; the aim should be to stay in right-brain mode, be creative and pay attention to the sound. Fortunately, *Sonar* has several tools to help you do just that.

Track icons: *Garage Band* did it first, and it's surprising how rapidly you can 'parse' tracks when they have track icons (see screen below). Channels in either view can include track icons, in a large or small size. I find it helpful to create small graphics of specific soft synths rather than just use a generic 'keyboard' icon. As track icons need to be 96 x 96 pixels, I'll usually grab an easily identifiable, 192 x 192 section of a soft synth and resize it in a paint program.

You need to tweak the bass level — quick! Which track has the bass? Track icons are easier to identify than text when you're in the heat of a mix. Also note the custom *Rapture* soft-synth icon.

Hiding tracks: The Track Manager is extremely valuable in creating a mix-friendly environment. In the Console or Track view, type 'M' to call up this function. Untick the tracks you don't want to see and tick the tracks that should be visible. The right of the window has a Toggle option, where you can select a particular type of track (for example, audio, MIDI, synth, buss). Ticking or unticking one of this type applies the same action to all tracks of that type.

Views On EQ

EQ is handled quite differently in the two main views. Although *Sonar Producer* has six bands of EQ built into each mixer channel, these are visible only in the Console view, or in a track's Inspector view in the Track view. What's more, in either the Console view or Inspector, you can see only four bands of EQ in the channel strip itself. To see all six bands, you need to right-click on the EQ frequency plot and select 'Show EQ Properties'. A pop-up window then appears. If you're working on an EQ-intensive mix (perhaps a typical rock band with a mix of acoustic and electric signal sources), the Console view is a far better choice for mixing, as you can see more clearly what's going on with respect to equalisation. The Track view's limitation of having a 'window' on only one channel's EQ is much more limiting.

rpa

Audio Visual & Pro-Media Specialists

The UK'S Leading Supplier of Disc Duplication, Printing Equipment, CD & DVD Media

CD-R & DVD-R HARDWARE

ESTABLISHED SINCE 1995

THE UK'S N°1 CD & DVD SOLUTIONS EXPERT

TEL: 020 3183 0000/1/2/3/4

RIVER DIAMOND 52x CD-R80

UNBRANDED INKJET PRINTABLE

AMAZING VALUE

PROMO CODE SOS230606 **£0.10p**

Not all discs are created equal and River Media's Diamond CD-R is no exception. The advanced recording dye has been developed to be transparent, creating a disc with a 'true silver' appearance. With high performance and ultra low error and jitter levels, the diamond CDR is the choice of the professional. These fast dry inkjet white or silver varieties create stunning prints on all inkjet disc printers.

Item Ref: 10400 (White)
Item Ref: 10401 (Silver)

RIVER SILVER SILVER 52x CD-R80

UNBRANDED THERMAL PRINTABLE

SUPERB VALUE

PROMO CODE SOS230606 **£0.09p**

The CDR80 Silver/Silver is what we class as River Media's bulk value duplication disc. Using the tried and tested standard phthalocyanine dye and shiny silver thermal surface, the River Silver/Silver is the choice of duplication houses world wide. Our unique dye formula ensures unbeatable reliability and cost effectiveness. A disc you can truly count on time and time again.

Item Ref: 10600

FREE NEXT DAY DELIVERY ON ORDERS OVER £150.00/- ex VAT *

100% ONLINE SECURITY GUARANTEED

EXPERT CUSTOMER ADVICE & TECHNICAL SUPPORT

EXCLUSIVE GREAT DEALS FOR ALL SOS READERS

PRICEMATCH WE WILL BEAT ANY GENUINE QUOTE **

M-TECH AUTOPRINTER 220

Resolution: 4800 dpi
Print Engine: 4 colour (6 colour opt)
Printer: R-Quest Flashjet 4800
Capacity: 220 discs
Operating System: PC
Loading: Fully Automated
Media Size(s): 12cm CD/DVD
Cartridges: HP 56,57 & 59
Software: CD Design Software

With over 10 years manufacturing experience in CD & DVD duplication technology, M-Tech have created a formidable autoprinter. M-Techs strong belief in its quality & reliability is shown by its 2 year warranty* as standard. This coupled with the popular FlashJet printer makes for a winning combination. The high resolution HP engine produces inkjet prints of the highest quality while taking advantage of the ultra low running costs of standard HP cartridges.

£ 2127.58
2309-MTECHAUTOPRINTER

New Low Price

PRIMERA BRAVO PRO PUBLISHER

Resolution: 4800 dpi
Print Engine: 4 colour
Drives: 2 DVD-RW 16x Recorders
Capacity: 100 discs
Operating System: PC / Mac OSX
Loading: Fully Automated
Media Size(s): 12cm CD/DVD
Cartridges: Primera 53335 & 53336
Software: Prassi Primo / Discribe

BravoPro is fast and easy to set up because everything's built-in. Just take it out of the box & plug it in. As the "big brother" to the Bravo II™ Disc Publisher - the world's best-selling CD disc publisher, BravoPro sets new standards for quality, speed and reliability. BravoPro's high-speed colour printing is the fastest you'll find. That's why they call it Afterburner Print Technology (APT).

Bravo Pro CD Publisher	£1695.00
Bravo Pro Autoprinter (Printer Only)	£1225.00
Bravo II CD Publisher	£1075.00
Bravo II DVD Publisher	£1295.00
Bravo II Autoprinter (Printer Only)	£813.00

£ 2075.00
8004-BRAVOPRODVD (PICTURED)

SAVE OVER £1300.00/-
RECOMMENDED SRP £3395.00/-

M-TECH TOWER 10 DVD DUPLICATOR

Disc Capacity: 10 DVD-R or CD-R
Drives: 10 x DVD±RW Writer
Burn Speed: DVD-R 16x / CD-R 48x
Dual Layer: 8x ready
Operation: Multi Mode
Hard Drive: 160Gb
Loading: Manual
Standalone: Yes
Reject Method: Pass/Fail LCD Display

M-Tech have worked hard to build and maintain their unsurpassed reputation for excellence in duplication hardware. Clients include the Met Police, BBC and the cream of the UK production and duplication houses. This 10 drive 16x DVD & CD tower duplicator is their most advanced yet.

SAVE OVER £115.00/-
RECOMMENDED SRP £909.39/-

£ 792.41
3202-MTECHDVD1210

RIVER CD & DVD ACCESSORIES

UK'S LOWEST PRICES

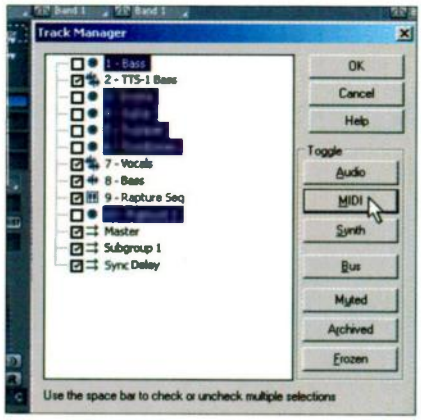
RIVER CD & DVD COLOUR PROTECTION SPRAY **£4.70p** each

RIVER CD & DVD 5" CLEAR POLY WALLET **£0.04p** each

RIVER CD JEWEL CASE WITH BLACK TRAY **£0.10p** each

more deals online at
www.riverproaudio.co.uk

River Pro Audio | Unit 6A | Juno Way Industrial Estate | London | SE14 5RW
Telephone : 020 3183 0000/1/2/3/4 | Fax : 020 3183 0006
Email : sales@riverproaudio.co.uk | Web : www.riverproaudio.co.uk
Registered in England 32055993 | Vat No. GB702147770



All MIDI tracks have been deselected, as the *TTS1* output has already been tweaked. The mix will now show only the audio tracks and busses.

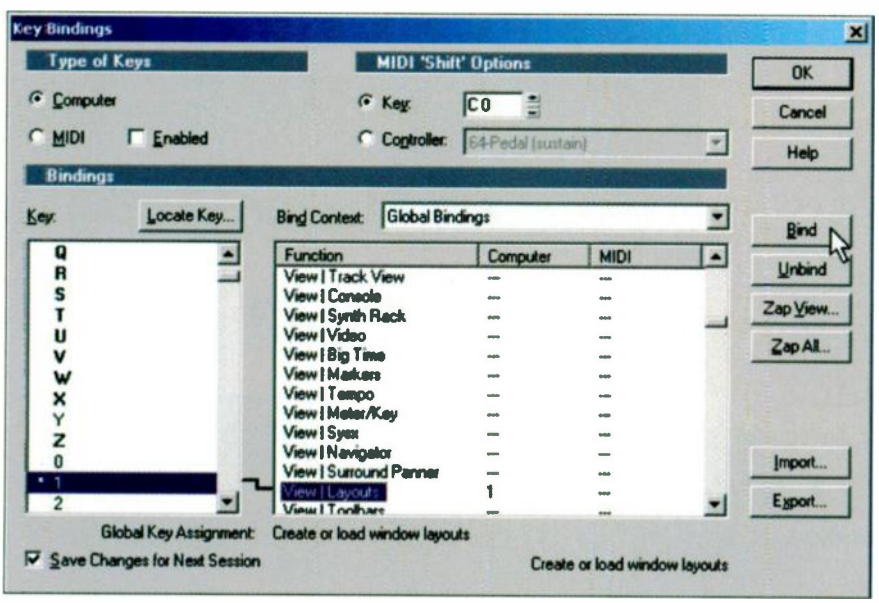
Note the orange button in the master buss track header; this indicates that Waveform Preview is on. Peak Markers have been enabled, and this one shows that the buss signal level reached +0.1dB.

▶ That's useful, but it's a little-known fact that the Console and Track views have independent Track Manager settings. For example, suppose you have a soft synth with multiple outputs. You can show only the audio outs in the Console view for clutter-free mixing, but in the Track view you can show the MIDI tracks driving the audio outs, should you need to edit them.

Track Layouts: Speaking of switching back and forth between Console and Track views, it's very helpful to be able to do this if you still have some residual editing to do as you mix — and that's where two mix-specific track layouts can come in very handy. Here's how to set them up.

1. Type Alt-3 to select the Console view.
2. Resize it to fill all the screen, except the program's toolbars at the top and status bar on the bottom.
3. Go View / Layouts, and then click on the 'Add' button.
4. Name it '!Just Console' in the dialogue box that appears. Using the exclamation mark moves the entry to the top of the list of layouts. Click on 'OK'.
5. Type Alt-0 to select the Track view.
6. Resize it to fill the screen, as in step two.
7. Go View / Layouts and click on the 'Add' button once more.
8. Name it '!Just Tracks' in the dialogue box. It will end up below '!Just Console' in the list of layouts. Click on 'OK'.

Now it's easy to select either view; go View / Layouts and '!Just Console' will be at the top of the list. Hit return, and you'll see the Console view. To go to the Track view, hit the down arrow once to select the next entry ('!Just Tracks') and hit return. But let's make it even easier...



The Layouts window is being bound to the '1' key for easy recall.

1. Go Options / Key Bindings.
2. Under 'Type of Keys', make sure 'Computer' is selected.
3. In the 'Key' field to the left, scroll down until you see '1', then click on it.
4. In the 'Bind Context' field to the right, scroll down until you see 'View|Layouts', and click on it.
5. Click on the 'Bind' button and then on 'OK' (see screen, left) Now you can call up the layouts window simply by typing '1' on your keyboard. At that point, the Console view is only a return key away; the down arrow key plus Return will get you to the Track view.

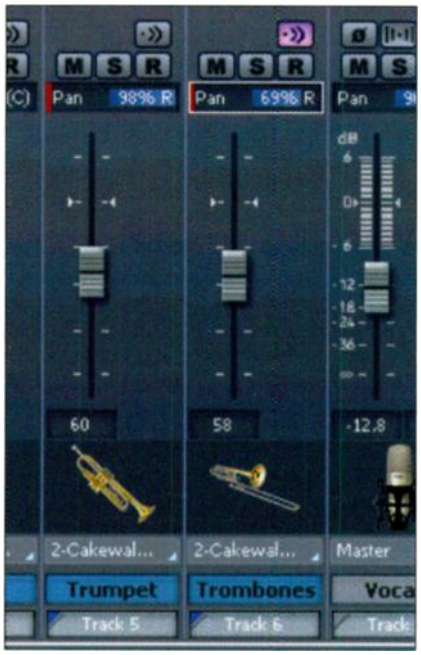
Waveform Previews: Enabling Waveform Preview on your master buss (which draws the mixed waveform in real time as you mix) can save a lot of time, by letting you catch when and where any distortion occurs, so you can stop, fix the problem immediately, then do another mix. This feature is unique to the Track view, and in some cases it's important enough to cause you to mix with the Track view instead of the Console view.

Buss and soft-synth tracks include a Waveform Preview button, located to the right of the solo button. (Note that this appears on soft-synth tracks only when the synth is inserted using the standard Insert command, which creates both a soft-synth audio track and accompanying MIDI track. If you create an audio track, insert the synth

into an effects bin, then create a MIDI track to drive it, *Sonar* assumes that these are separate audio and MIDI tracks, not an integrated instrument track.) Click on this button to enable Waveform Preview.

As the audio plays through the buss, *Sonar* will draw a waveform in the buss track. Should distortion occur, you'll see a red line there. This makes it easy to rewind and see why there's distortion — for example, a loud bass note and a kick drum might hit at exactly the same time. Do the fix, rewind to just before the distortion occurred, play back, and see if the distortion continues. If not, proceed with your mix. It's also possible to enable a Peak Marker at the buss' highest level that shows the level attained by the peak. To do this, click on the meter options and select 'Show Buss Peak Markers' (see screen on left).

Quick Groups: This feature applies to Track or Console view. We've already covered the more 'formal' version of grouping in a previous article (in short form, you right-click on a parameter, and assign it to a group). However, Quick Groups are, as the name implies, a quick way to group and ungroup faders without having to go through the effort of assigning them to a specific group. What's more, when you Quick Group channels, almost all parameters are grouped. For example, assume you have two guitar tracks and you want to shift them both slightly to the right in the stereo field. Assign their associated channels to a Quick Group, tweak the panning on one track, and the other will move correspondingly. Any number of tracks can form a Quick Group. To create one:



The trumpet and trombone tracks have been 'Quick Grouped', as evidenced by the small blue triangles to the upper left of the Track 5 and Track 6 labels. Pan is being altered on track 6; track 5 follows along, and both parameters have a small red rectangle to indicate they've been grouped.

1. Click on the small triangle in the upper-left corner of the track name (in either the Console or Track view). It turns blue.
2. Control-click on the small triangles of any other tracks you want associated with the group. They turn blue as well.
3. Edit the parameter in one of the tracks. The parameter will be outlined, and small red rectangles will indicate parameters that are grouped with the parameter you're moving (see screen, left). Grouped tracks will follow your edits.
4. To cancel the Quick Group, click on one of the blue triangles, or click on the triangle in a different track. SOS

All About Quantise

In *Cubase SX*

Mark Wherry

When sequencers first allowed you to record musical notes in real time, one of the features that became ubiquitous in all systems was quantisation. This month we take a look at how quantise has evolved and is used in *Cubase SX*.

In its broadest scientific sense, quantisation means taking a continuous value and making it fit against a scale of pre-determined values. In MIDI sequencers like *Cubase*, the timing of notes is quantised against the internal resolution of the sequencer, which is measured in 'Parts Per Quarter Note' (or PPQN), as you perform them. The higher the PPQN, the more accurately the timing of your performance is rendered. However, while the issue of PPQN was really important maybe 15 years ago, modern sequencers offer such a high internal resolution that nobody really talks about the resolution of their sequencer any more.

Although modern sequencers do a pretty good job of accurately representing the timing of your playing, even the best player will deviate from the metronome click when recording. So it can be useful to further quantise your playing against a much larger or smaller resolution (depending how you look at it), such as a quaver (or eighth note), enabling you to effectively take an out-of-time recording and make it completely accurate against the sequencer's precise musical time grid. There are both functional and creative uses to *Cubase*'s Quantise features, which is what we'll be looking at in this month's technique article.

Quantising In *Cubase*

Let's start with the basics. Say you've played a MIDI part into *Cubase* (badly, as shown in Figure 1) and you want to simply tighten up the timing by quantising the start positions of the notes. In the Project window, you would choose the quantise resolution from the Quantise pop-up menu on the toolbar, select the MIDI part (or parts) you want to quantise, and select 'MIDI / Over Quantise', or press 'Q'.

You can also do this from the Key Editor, which features a Quantise pop-up menu on the toolbar that's linked to the one on the Project window (choosing a resolution in one window automatically updates the other with the same selection, for example). However, if there aren't any notes selected in the Key Editor, all the notes displayed in the editor will be quantised, unless the 'e!' Edit Active Part Only button is active, in which case only the notes in the currently active part will be quantised. You'll notice the resolution specified in the Quantise pop-up menu also determines how the grid is drawn in the Key Editor, which can be quite useful when you get into more complex rhythmic quantise patterns.

Once you've quantised your part, all the notes will move to the nearest beat specified by the resolution you chose from the Quantise pop-up menu. If you look at Figure 2, you'll notice that I quantised my original recording with a resolution of 1/16 (a semiquaver) — in order to preserve the correct timing of the music, it's important to make sure that you choose the shortest note used for the rhythms in your performance as the quantise resolution. The shortest notes used rhythmically in my performance were semiquavers, which is why I chose 1/16 as my resolution — if I had chosen 1/8 (as in Figure 3), you will notice that the basic rhythm of my original performance would have been destroyed.

Selecting the 'wrong' resolution isn't always a bad thing, though, as you could end up with a creative variation on your performance by over-quantising. Alternatively, if the shortest note is a semiquaver, you might want to tighten the timing a little by quantising with a shorter resolution, such as a 64th note, although there are better ways of achieving a more human feel with quantisation, as we shall see shortly. As a footnote, it isn't just straight note values that are available: if your rhythms are based on dotted notes (such as two dotted crotchets, or quarter notes, per bar of 6/8, for example) or triplets, there are suitable options also available in the Quantise pop-up menu.

On the subject of changing resolutions, remember there's nothing to prevent you from making selections of different notes in the Key editor and quantising them with different resolutions. This can be handy if you play a piano part with a slow-moving bass line underneath a fast arpeggiation for example.

Once you've quantised notes, you might want to change your mind and return to an unquantised version and try something else. *Cubase*'s multiple undo feature can come in handy here, but what if quantising wasn't the last thing you did, or what if you quantised the notes in question last week and many different versions of the Project ago? Fortunately, *Cubase* offers a dedicated Undo Quantise command (which you can find in the MIDI / Advanced Quantise menu, although I would usually assign it to the 'U' keyboard shortcut, as it was in 'classic' *Cubase*) to return notes to their original positions, which are always stored in the Project.

This is pretty handy, but it's important to note that Undo Quantise always returns notes to the positions they occupied when they were first input into *Cubase*. This matters when you use Quantise multiple times, because if you play in some notes, quantise them, and then further quantise them again, Undo Quantise will undo to the original positions, not the intermediary quantised positions. If you want to reset the currently quantised notes as the original positions for a set of notes, you can do this with the Freeze Quantise command.

Advanced Quantise

So far we've looked at a fairly functional use of quantisation — moving notes to rigid metronomic positions — but it's also possible to use other rhythmic templates and a variety of other options in *Cubase*'s Quantise Setup window. To open this window, select either 'MIDI / Quantise Setup' or choose Setup from the bottom of the Quantise pop-up menu. The Quantise Setup window is where you set up the quantise templates we've already been using from the Quantise pop-up menu, and once this window is open, any settings you make automatically become the currently selected quantise template.

The Grid and Type menus at the top of the window allow you to define the type of quantise templates we've looked at already: Grid allows you to pick a note length (for

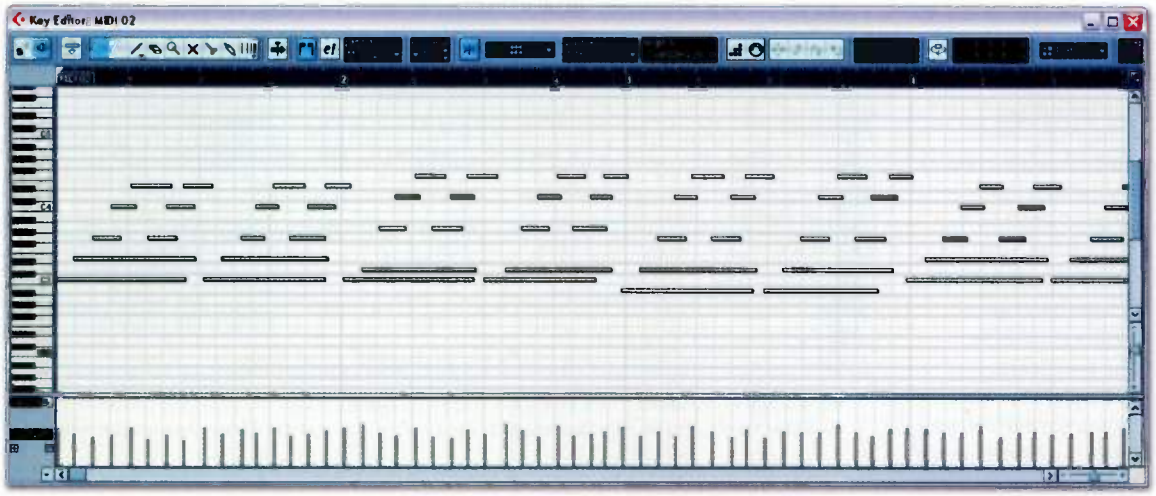


Figure 1: Here you can see a badly played MIDI part that's not quite in time, before being quantised.

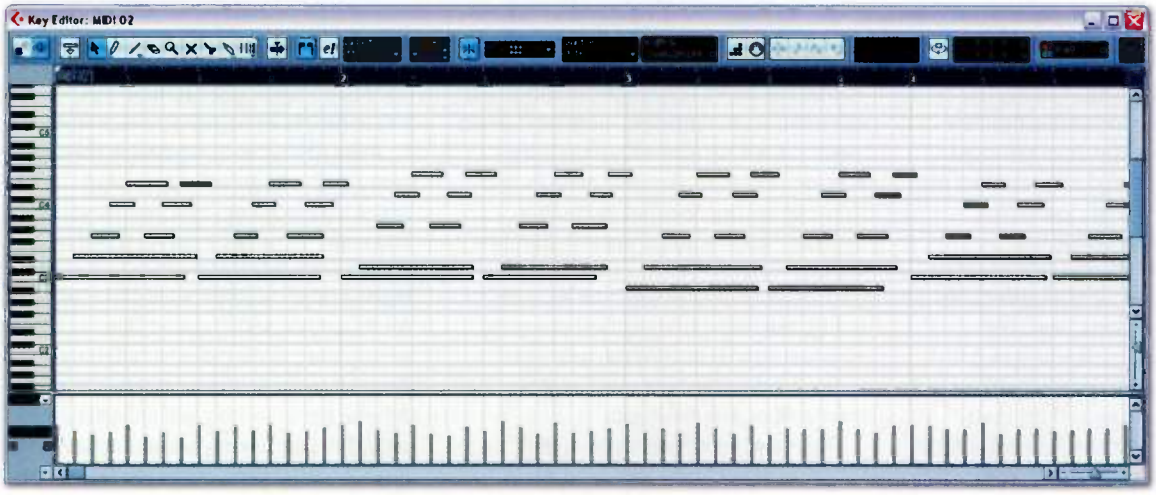


Figure 2: After quantising the notes using a 16th-note resolution, everything is precisely in time.

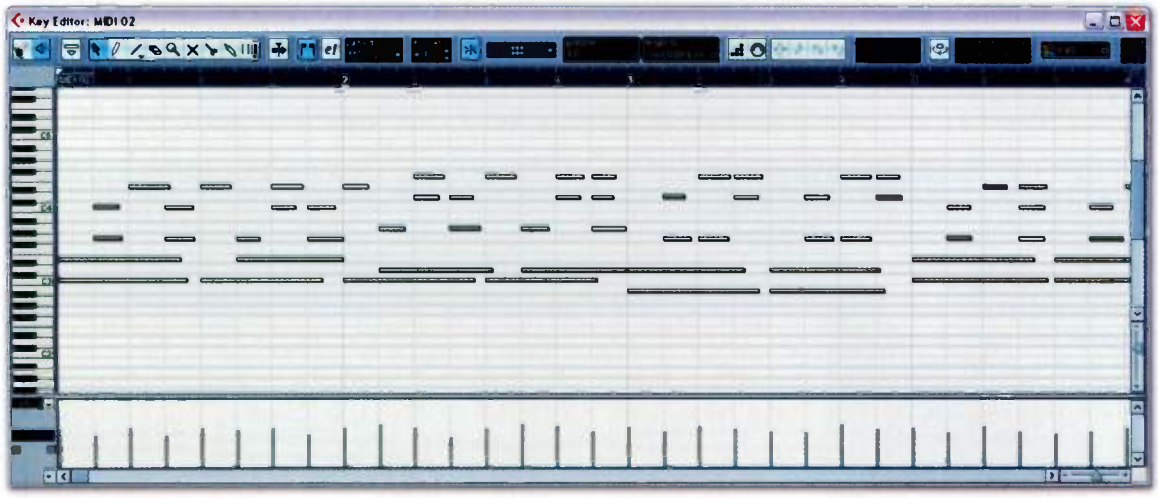


Figure 3: By over-quantising the original performance with quavers (eighth notes), the piece becomes substantially different.

example: 1/4, 1/8, and so on) to use for the resolution, while Type sets a modifier for the note length: Straight, Triplet or Dotted. Speaking of triplets, there are times when you might need to quantise against other divisions of a beat other than three (ie. triplets), and *Cubase* allows you to set up such patterns with the Tuplet setting, which is set to 'Off' by default.

Take the third Opus-90 *Impromptu* by Schubert, for example, where you have crotchet sextuplets (six notes divided into the space of a quarter note). To set up a suitable quantise template in *Cubase* you would set Grid to 1/4, Type to Straight and Tuplet to six.

There's also a Swing slider where you can specify how much of a swing feel (as a percentage) to apply to the quantise pattern. For example, say you have a straight-16th performance (much as in the example figures) and you want to inject a swing feel: set Grid to 1/16, Type to Straight, and add as much swing as you can handle. To carry out the quantise operation you can either use the normal commands or click 'Apply Quantise' in the Quantise Setup window — see Figure 4 on the next page.

As you were dragging the Swing slider, you might have noticed that the grid in the Key editor is automatically redrawn to show where the beats fall in the new quantise template. You can also make *Cubase* automatically apply the actual quantising to the notes while you experiment with settings in the Quantise Setup window by enabling the Auto checkbox just below the Apply Quantise

▶ button.

In addition to the Key Editor showing the quantise pattern in its grid, the Quantise Setup window also offers a graphical display of where the beats will be in the current quantise pattern, represented by dark blue lines. This is particularly useful for another Quantise Setup feature called Magnetic Area. Normally when you use Quantise, notes are moved to the nearest beats as specified by the quantise template; however, the Magnetic Area allows you to specify a distance from the beats in the quantise template (as a percentage) and only if a note is within the specified distance from a target beat will it be moved. The Magnetic Area is indicated on the Quantise Setup window's grid by a light blue region drawn around the darker blue 'quantise beat' lines.

The opposite to Magnetic Area is the Non-Quantise setting, which enables you to specify a distance in ticks (where there are 120 ticks in a semiquaver) from the target beat where notes within this distance will not be quantised. This is pretty useful for rendering a more natural-sounding performance, as you quantise only the notes that are badly out of time, leaving the ones that are almost on the beat alone. The Non Quantise distance is indicated on the Quantise Setup window's grid by light red regions that are drawn around the beats.

Another feature that can be useful to make your performances sound more in time without being completely mechanical is Random Quantise. Here you specify a value in ticks so that when a note is quantised to the nearest beat specified by the other parameters in the quantise template, it is offset by a random amount from zero to the value specified by the Random Quantise setting. This is similar to the final Quantise parameter, Iterative Strength, which is used in conjunction with a separate Iterative Quantise command in the MIDI menu. Iterative Quantise works similarly to normal Quantise except notes are moved closer to the quantise grid by the percentage specified by Iterative Strength. Repeated use of Iterative Quantise will keep moving the notes closer to the grid, so you can experiment with how relaxed you want to the timing to sound, and eventually Iterative Quantise will end up completely quantising the notes as if you'd used the standard Quantise command.

There are two other features in the Quantise Setup window that can come in useful. Firstly, the Move Controller option specifies whether controller data within a note being quantised should be moved along with the quantised note, and secondly, the 5.x Import button allows you to import a quantise pattern from an earlier version of *Cubase* that was saved as an old *Cubase VST* 5.x Part (.prt) file.

Once you have a quantise template you want to use again, you can store it as a Quantise Preset so that it appears in the Quantise pop-up menu by clicking the Store button in the Quantise Setup window. The Preset is automatically created (and selected) and is given a name by *Cubase* based on the current settings in the Quantise Setup window. If you want to change the name, with the Preset selected, double click the Presets pop-up menu in the Quantise Setup window, enter a new name in the aptly-titled 'Type In A Preset Name' window and press Return or click 'OK'. The currently selected Preset in the Quantise Setup menu can be deleted by clicking the Remove button.

That's all for this month's *Cubase* Technique column but keep your eyes open for a future edition where we'll be taking an in-depth look at Groove Quantise. SOS

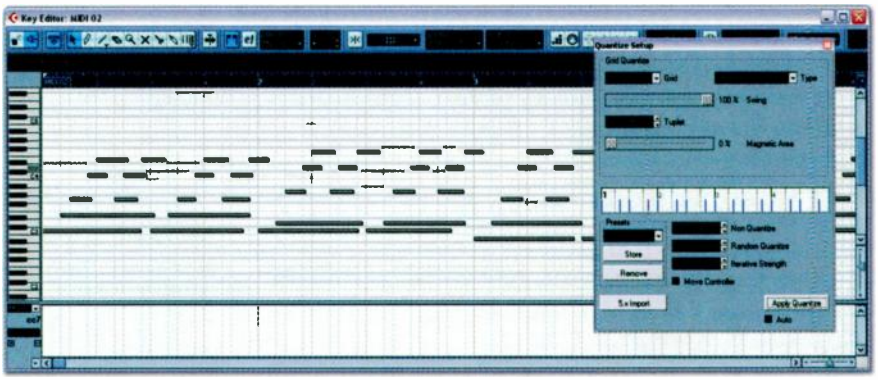


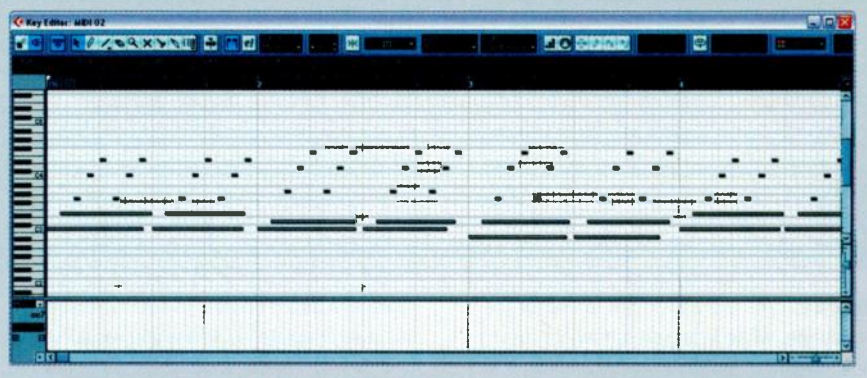
Figure 4: Here you can see a 16th-note swing pattern setup in the Quantise Setup window and applied to the notes in the Key Editor. The grid in the Key Editor also reflects the grid described by the Quantise Preset — Bach would be proud.

It's Not Just About The Start

With the exception of Groove Quantise, where the Velocity and Length of a source recording are included in the Quantise preset, the quantise commands mentioned in the main text (Quantise and Iterative Quantise) usually only operate on the start positions of notes. However, there are two further Quantise commands that let you process either the end position of the note or the length of the note — the appropriately named Quantise Ends and Quantise Lengths commands — which are available from the MIDI / Advanced Quantise sub-menu.

Quantise Ends does exactly what you would expect, and moves the end of the note (leaving the start position where it was) to the nearest grid position specified by the current Quantise Preset. Quantise Lengths quantises the length of the note to the nearest value specified by a separate Length Quantise pop-up menu on the Key Editor's toolbar. By default, Length Quantise is set to Quantise Link so that the value is taken from whatever the Quantise menu is set to, although you can set this to an independent value if you wish.

It's important to remember that Quantise Lengths doesn't make all the notes the length set by the Length Quantise menu. This is possible with a command called Fixed Lengths, which uses the resolution specified in the Quantise pop-up menu and makes the specified notes the length set by that resolution. For example, if you want to make a set of notes precisely demisemiquavers in length, select 1/32 in the Quantise pop-up menu, select the appropriate notes if you're in the Key Editor (or none at all to process all the notes in the Key Editor) or the appropriate part in the Project window and choose MIDI / Functions / Fixed Lengths.



By selecting the upper notes, setting Quantise to 1/32 and using the Fixed Length command I was able to get an instant Glenn Gould-like feel for my performance.

PORTABLE VOCAL BOOTH £229

"if you own a microphone you need one of these"
Steve Levine

The Reflexion Filter

Patent Pending

Exclusive Worldwide Representative for sE Electronics
01582 470260
www.sonic-distribution.com
SONIC DISTRIBUTION

sE
Sound Engineering
www.seelectronics.com

Spider Sense

Using *Reason*'s *Spider Audio* & *Spider CV* Devices

Derek Johnson

The grey *Spider* devices look mild-mannered and unassuming — but they have super-powers that can unlock the hidden reserves of *Reason*'s sound-making devices, as well as solving a host of more mundane problems.

Reason's *Spider* merger/splitters must be the least processor-heavy devices in the software's rack, but although recent *Reason* developments may have overshadowed their potential, used creatively they can help you redefine what's possible with the other devices.

Both varieties of half-rack device are uncomplicated in design, with *Spider Audio* being probably the most simple *Reason* device of all. It provides two basic circuits: a 4:1 mono or stereo merger designed to merge four signals into one, and a 1:4 stereo splitter designed to split a single signal into four so that it can be sent to four different destinations. Front-panel 'LEDs' indicate the presence of audio activity, and that's it for external features (see pictures above). There aren't even any level controls for the inputs in the merge circuit. As for automatic routing issues, there aren't any of those either, since the user makes all the connection choices.

Spider CV has a little more to it. Again, it offers a single 4:1 merge circuit, for gates and/or CVs, and each input is equipped with a sensitivity knob so that input signals can be balanced. Two four-way splitter circuits are also offered. This is quite logical and allows the Gate and CV outputs of the *Matrix* sequencer to be split to multiple devices from one *Spider*, with each device playing the same *Matrix* sequence.

Spider devices, both audio and CV variants, can be infinitely chained with no processing delays, CPU compromises or other artifacts. To create a big merge, just link the merged output of one *Spider* to a merge input of another, alongside any other audio or gates/CVs that are being merged. Similarly, big splits are created by routing a split output to another splitter-circuit input.

Spider CV does have some automatic routing logic. For example, one output from each of the two splitting circuits will connect to the first available pair of Gate and CV ins in the rack, and a *Spider CV* created while a *Matrix* is highlighted creates an automatic link between the *Matrix*'s Gate and CV outs and the inputs to the two *Spider* splitter circuits. This might not always be convenient, of course. The solution is to hold down the Shift key while creating a *Spider CV*; doing so disables automatic connections. (Actually, as most of you are probably aware, this operation disables automatic connection with any device.)

Simple Uses Of *Spider Audio*

While there are no really esoteric uses for the *Spider Audio*, the facilities it offers quickly become indispensable when you're linking large numbers of effects. The simple mixing offered by the merge circuit is also a tidy little problem-solver.

The classic *Spider Audio* merging application is to mix audio from several related devices. For example, a number of *Dr:Rex* devices loaded with REX loops of the different sections of a sampled song would ordinarily require a *Remix* mixer input each. Using *Spider Audio*, up to four can be mixed, with a single stereo out requiring just one *Remix* channel; all EQ and effects processing on the channel would thus be the same for all the related REX files, which is generally what you'd want (see top screen, opposite).

The operation is simple. First, create a *Dr:Rex* for each related REX loop (two sections of a long verse, a chorus and a break, say). Then create a *Spider Audio*. Disconnect the *Dr:Rex* devices from the *Remix* mixer and reconnect them to the *Spider*'s merge circuit. Name the *Spider* something like '*Dr:Rex* submix' and route the merge circuit's output to a *Remix* input. The naming step is a good habit to get into, since the name is reflected in the *Remix* input-channel scribble strip and will help you keep track of the elements of a really busy rack.

As an aside, remember that grouping this bunch of related *Dr:Rex* devices into a *Combinator* patch will make for easy recall later in a different song. Highlight the *Dr:Rex*es and the *Spider Audio*, then select the 'Combine' command from the Edit (or contextual) menu, and finally save the result as a Combi patch. Again, if you choose this step, give the *Combinator* a meaningful name so that you can track what's happening from its *Remix* input channel.

Speaking of the *Combinator*, the *Spider Audio* could have a similar use within a complex Combi. Let's say you've created a Combi that consists of several *Subtractor*s or *Malström*s that are set to different velocity

and/or key ranges, for a dynamic velocity-split effect. They'll need to be mixed in some way. You can use a *Remix* mixer or a *Micromix* line-mixer in the Combi if you need their facilities (such as panning and effect sends), but if simply summing the layered synths for common processing (as described in the *Dr:Rex* example above) will suffice, use one or more *Spider Audios*. Of course, it's perfectly OK to merge several *different* devices, if all you need is processor-efficient submixing and are happy to set levels on the devices themselves. The merged result could even then be processed in the same way as a stereo submix on a real-world mixing desk — add compression, EQ, or even a chain of *MClass* mastering processors to the group. Merged audio is also good *BV512 Vocoder* input fodder — either as modulator or carrier.

Spider Audio & Effects

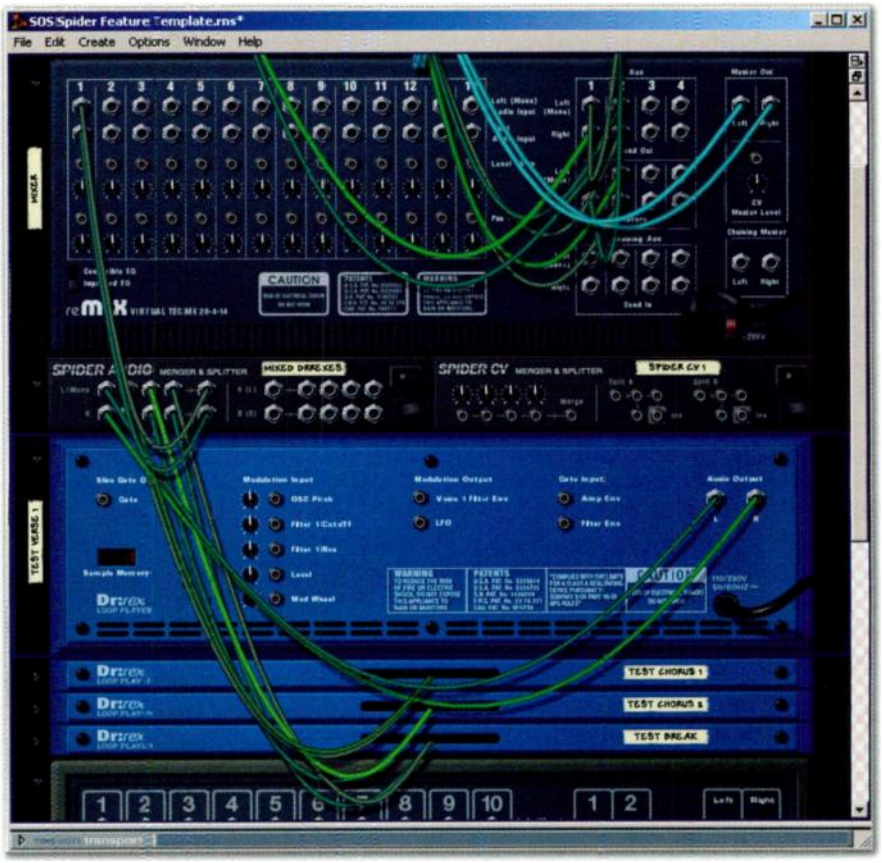
The *Spider Audio*'s splitter circuit has yet more uses. Routing a *Remix* aux send output to the splitter input lets you create a four-way parallel effects chain (just take each split of the signal to an effect of your choice). Using effects in parallel produces a different sound to chained effects, allowing each processor to cleanly add its own treatment to the mix, unaffected by the others. The effect outputs will probably need to be mixed somewhere, so why not use the merging circuit on the same *Spider Audio*, if nothing fancy is required? Just take the last stereo output from each of the four chains and patch them to the inputs of the merge circuit. The merged output can then be routed to the original *Remix* send's aux return (see screen, right).

The parallel processing idea can also be used when creating complex insert effects — where sound-making devices in the rack are patched directly to effects devices. Again, use the splitter circuit and merge the results all within one *Spider Audio* device.

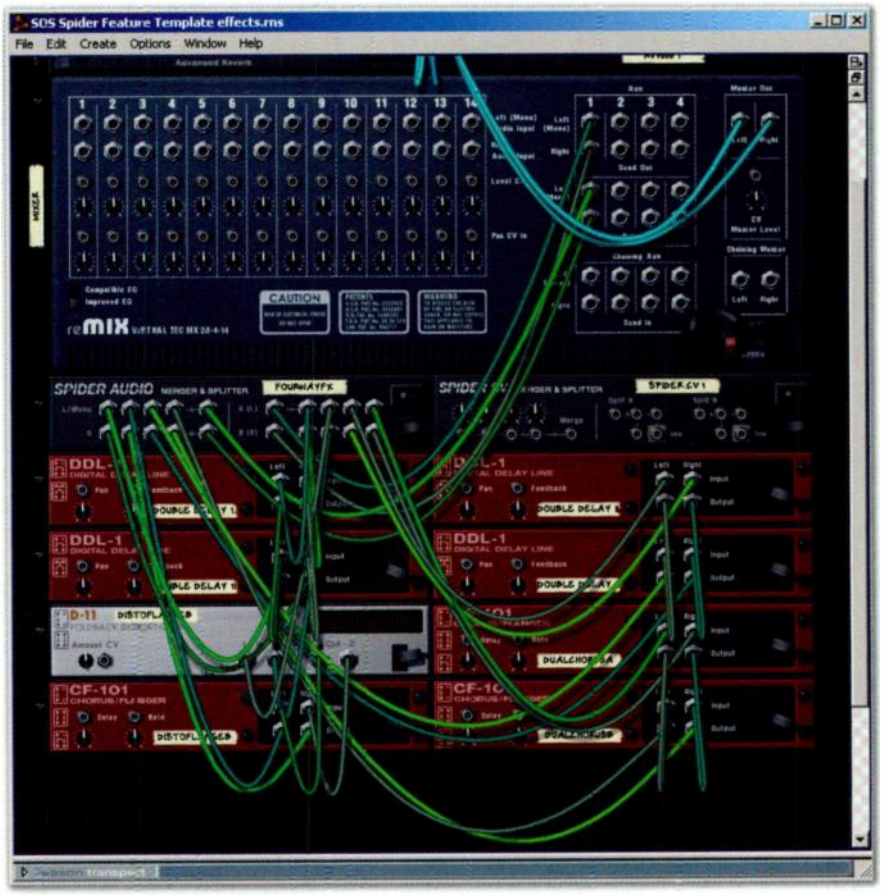
Spider CV Splitting

The *Spider CV* device provides real creative potential. Again, there are simple uses, the most obvious being to let a single *Matrix* sequencer play several devices in parallel.

This technique exploits the *Spider*'s two splitter circuits. Create a *Matrix* and a *Spider CV* and simply connect the *Matrix*'s gate and CV outs to a splitter input each (you'll only be able to use three of the inputs, as the fourth is fixed to 'inverted' — more on this below). Now route the three pairs of split outs to three devices to be played in parallel: you could create three *Subtractors*, each loaded with a different patch. Ensure that the split CV and Gate signals are routed to the relevant inputs on the target synths. Instant layered parts are easy, and you can keep your result tidy, as ▶



Four *Dr:Rex* devices in a simple REX loop setup — they only use one input of the mixer. There could be lots more, either chained through multiple *Spider Audios* or in parallel via several *Spiders*.



A simple four-way parallel multi-effect setup. One aux send feeds four simple chains — a couple of dual delays, a dual chorus and a distorted flange — the outputs of which are mixed by the same *Spider Audio* that does the splitting.

always, by loading the linked devices into a Combi (and perhaps merging their audio outs with a *Spider Audio*).

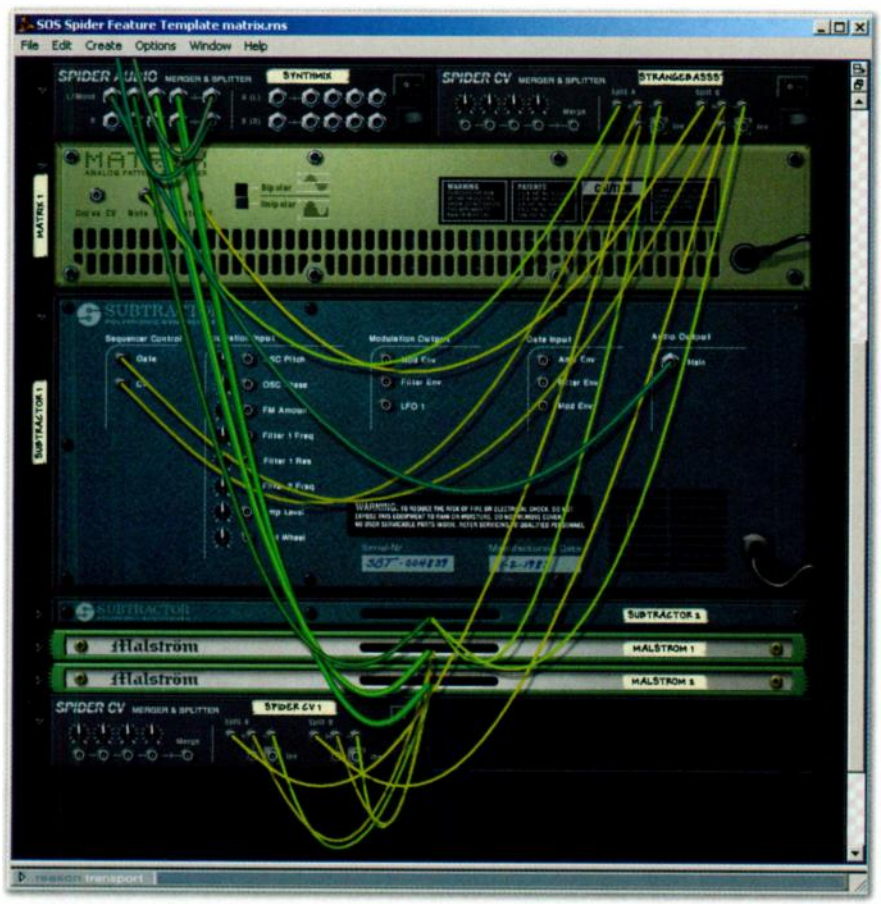
As mentioned above, one of the *Spider CV*'s outputs is inverted, and this can be useful for special effects. For example, interesting rhythmic patterns can be produced: route a straight *Matrix* gate to, say, *Subtractor*'s normal Gate input and the inverted version to its FM CV input (adjust the sensitivity control and make sure that both oscillators are active, so that FM can take place). Although the part will play normally, the FM effect will hit on the off-beat. Normally, the inverted output would be used with a modulation CV that's being split to, say, the delay CV inputs of a pair of *CF101* flanger effect devices; the normal and inverted modulation CVs would move in opposite directions, creating a more complex flange or phase that can be enhanced by panning the outputs of the two devices to opposite sides of the stereo field.

Of course, splitting isn't restricted to playing layered devices. Any gate or CV on a *Reason* device's rear panel can be split and sent to multiple destinations. Doing so is ideal if you'd like to route, say, one of a *Malström*'s complex modulator waveforms to a *Subtractor* or *NN19* while still having it routed to one of the Malström's own parameters. Not only would the target parameters — filter frequency, oscillator phase or whatever — be modulated at the same tempo, they'd be treated with the same modulation pattern, which adds a nice homogenous feel to a mix. The actual parts being played don't necessarily have to be related — the targets could be bass lines, pads and leads — but the rhythmic fluctuations can work on an almost subliminal level to add a feeling of unity to the mix.

Spider CV Merging

Merging gates and CVs enters a slightly esoteric area, but if you have *Reason* running as you read, and you try some of the examples, you'll hear what I'm driving at.

One basic use of the merging option is to create complex modulation waveforms from other devices' LFOs (or Malström's already complex modulators). In these circumstances, you might even create devices just to independently access their modulators. The merge circuit allows each contributing modulation source to be freely mixed, with sensitivity knobs allowing excellent fine-tuning of the result. An example is easy enough to describe.



The *Matrix* plays two *Subtractor*s and two *Malström*s, via *Spider CV*. Just for fun, the audio from the four synths is mixed by a *Spider Audio* that happened to be on-screen. An extra *Spider CV* is required, due to the first *Spider CV*'s fourth CV output being fixed to 'inverted'. However, chained *Spider*s introduce no delays.

Reason News

Peter Tools, they of the unexpected *Reason* add-ons, are now marketing the full release version of their *Hammer* Rewire-based audio-input tool. It's PC-only and allows audio input via the *Rebirth Input Machine* device. In addition, the *Live Set* collection of real-time MIDI performance modification tools has just had an upgrade, to v1.5.

Check out the Peter Tools web site (www.petertools.com) for details of a neat bundle of both packages for 109 euros (or US$138). In addition, registered *Live Set* users can buy *Hammer* for 10 euros (US$12). Or you can just buy *Hammer* by itself for 30 euros (US$38).

- Create a *Subtractor* and a *Malström*, plus a *Spider CV*.
- Flip to the back of the rack (hit the Tab key), and route the *Subtractor*'s own LFO1 modulation output to a *Spider CV* merge-circuit input.
- Then route *Malström*'s Mod A modulation output to another input of the same circuit.
- Connect the merged output to any target. The illustrated example (see top screen overleaf) routes the merged CV to the *Subtractor*'s FM Amount modulation input.
- Play a note or pattern on the synth and tweak the merging input sensitivity knobs to hear the result. I tend to start with the mod input sensitivity controls fully left and tweak each one slowly to hear what they're adding to the overall sound. Remember that the LFO or modulator has its own suite of parameters to play with on the front panel.

Of course, this merged modulator could be routed to several CV inputs. Just route the merged output to the input of the CV-splitting circuit of the *Spider CV* and connect the outputs to the desired destinations.

Spider CV Gate Merging

Gates can also be merged, to unexpected effect. For example, merging the gate outputs of up to four *Redrum* voices can create a user-definable trigger source that offers lots of new rhythmic possibilities — as long as the relevant voices have been programmed to play as part of a *Redrum* pattern! (However, the voice doesn't need a sample loaded into it in order for its channel to be programmed — the gate will be transmitted in any case.) One of the merged gates could also be the main trigger being generated by the *Matrix*.

In the next example, we'll create a *Subtractor* with attached *Matrix* sequencer, a *Redrum* and a *Spider CV*.

- Create the devices as listed above.
- Program a drum pattern — the one in the illustration (see bottom screen overleaf)

ANDERTONS Music

MUSICIANS NO.1 SUPERSTORE

est. 1964 — Anderton's Music Company — Established 1964

ORDER HOTLINE: 01483 456733

www.andertons.co.uk

Next Day Delivey on all 'in-stock' items.

STAMP OF APPROVAL

DEALS OF THE MONTH!

Native Instruments 10th Anniversary Special

To Celebrate 10 years of fantastic great product....

Andertons have FAB deals on selected Native Instrument products. Visit our website at www.andertons.co.uk for the full picture. Below are just some examples!

LIMITED STOCKS!

NATIVE INSTRUMENTS — SOFTWARE SYNTHESIS

ABSYNTH 3

£99.99

SAVE £90 WAS £189.99!

TRAKTOR 3.0 DJ STUDIO

£79.99

SAVE £120 WAS £189.99!

FM7

£79.99

SAVE £130 WAS £199.99!

Guitar Combos

£49.99

SAVE £70 WAS £119.99!

Alesis HD24

THE industry standard stand alone hard disc recorder!

Features full 24-bit/48kHz recording, dual recording bays, bullet-proof reliability, and more!

WAS £989.99 - SAVE £120!

NEW LOW PRICE!

£869.99

Yamaha AW2400

24 Track System. Built In CD-RW and USB.

The new flagship of the AW line with 24 simultaneous tracks of playback, motorized 100mm faders.

WAS £1499.99 - SAVE £200!

NEW LOW PRICE!

SAVE

£1299.99

Hypersonic 2 Bundle

Includes 3 Wizoo Hypermodules:

GP-2 Grand Piano

NB-3 Hammond B-3

US-1 Ultra Synth - Leads Pads etc

WAS £299.99 - SAVE £50!

LIMITED STOCKS!

£249.99

Tascam US122

USB Powered Audio Interface with MIDI -

Where size matters, the new, ultra compact, US-122 brings portable, easy to use, professional recording capabilities.

WAS £149.99 - SAVE £40!

LIMITED STOCKS!

£109.99

Novation X Station 25

All in one - With USB

Combines a synthesiser, audio solution, MIDI remote controller, MIDI interface and a multi-effects processor.

WAS £369.99 - SAVE £70!

NEW LOW PRICE!

SAVE

£299.99

Emu 0404

All this for less than £70!

- ✔ EMU 0404
- ✔ 2GB PROTEUS SOUND LIBRARY
- ✔ EMU VINTAGE X PRO BUNDLE VOLS. 1, 2 AND 3

SAVE OVER £100!!

TURBO BUNDLE!

£149.99

ALL THIS AND MORE!

YAMAHA · ALESIS · digidesign · MACKIE · M-AUDIO · BEHRINGER · RØDE · EDIROL · Focusrite · TASCAM · Roland · AKG · MOTU · Soundcraft · audio-technica · KORG · steinberg

SYNTHS, GEAR, TOYS ETC.

SYNTHS AND WORKSTATIONS

Alesis Fusion 8HD 88 key workstation	999.99
Korg Triton Extreme 61 Key Workstation	1329.99
Korg Triton Extreme 88 Key Workstation	1859.99
Roland SH201 Analogue-modelling synth	499.99
Yamaha MO6 61 note Synthesizer	949.99

MULTITRACK RECORDERS

Edirol R09 24bit Wav/MP3 recorder	299.99
Korg D3200 32 Track Hard disc Recorder	849.99
Roland VS2000CD 20 Track Multitrack Recorder	1259.99
Tascam 2688 24 Track Multitrack Recorder	729.99
Yamaha AW1600 16 track hard disc recorder	649.99
Zoom MRS8 8 Track Digital Recorder with free 128mb card and Power Supply	219.99

STUDIO MONITORS

Adam P33 Studio Monitor - EACH	724.50
Alesis M1 active 520 monitors (pair)	199.99
Audioblob Media Speaker System black	149.99
Audioblob Media Speaker System white	149.99
Emu PM5 monitors (each)	149.99
Event ALP5 Speakers (Pair)	279.99
Event SP8 Studio Precision Monitor Speakers (pair)	849.99
Genelec 8030A Two Way Active Studio Monitor (Each)	299.99
Genelec 8050A Two Way Active Studio Monitor (Each)	829.99
Mackie HR626 Active Studio Monitor (Each)	349.00
Mackie HR824 Active Studio Monitor (Each)	449.99
Prodipe Pro5 5 inch Active Studio Monitors	249.00
Yamaha HS50M Powered monitor	124.99
Yamaha HS80M Powered monitor	194.99

CONDENSER MICROPHONES

AKG C2000B Condenser Microphone with Cradle	99.99
AKG C3000B Condenser Microphone	139.99
Audio Technica AT3035 Condenser microphone	120
Rode NT1000 Cardiod Condenser Microphone	169.99
Rode NT1A Large Diaphragm Condenser Mic with Shockmount	119.99
Rode NT2000 Multi Pattern Studio Condenser	269.99
Rode NT3 Condenser Microphone	99.99
Rode NT6 Condenser microphone	214.99
Rode NTK Valve Condenser Mic	269.99
SE Electronics SE1a Condenser Microphone	69.99
SE Electronics SE2200 Microphone	139.99
SE Ribbon Ribbon mic wih 2 micron diaphragm	649.99
Sigma Ribbon microphone	429.99

COMPUTER SOFTWARE

Apple Logic Pro 7.2 (Mac only)	639.99
Digidesign Music Production Toolkit	339.99
Emu Emulator X 2	219.99
IK Multimedia AmpliTube 2	268.99
IK Multimedia CSR Classik Studio Reverb	199.99
IK Multimedia Miroslav Philharmonik	289.99
IK Multimedia SampleTank 2.1 XL	334.99
IK Multimedia T-RackS plug-in	259.99
Steinberg Cubase SX3 Trade in version (proof of ownership required)	299.99
Studiocase 2 inc SE3, Virtual Bassist SE	169.99

AUDIO & MIDI INTERFACES

Digidesign MBOX2 Factory Edition	389.99
Digidesign MBOX2 with Pro Tools LE Software	299.99
Edirol FA66 Firewire Interface	199.99
Edirol Firewire FA101 Audio/Midi Interface	299.99
Emu 0202 USB2 Audio Interface	97.00
Emu 0404 USB2 Audio Interface	149.99
Emu Proteus X 2	149.99
Focusrite Saffire Firewire Interface	299.99
Focusrite Saffire Pro 26 I/O	469.99
Mackie SPIKE Powered USB Recording System	149.99
MOTU 828mk2 Audio / Midi Interface	541.99
MOTU HD192 Core System	1299.99
MOTU Traveler Firewire Interface	699.99
MOTU Ultralite Firewire Audio Interface	469.99
Novation X Station 49	349.99
Presonus Firebox Firewire Audio Interface	249.99
Presonus Inspire Audio Interface	139.99
Tascam FW1804 Firewire Audio Midi Interface	339.99
Tascam FW1884 Firewire Audio Interface and Controller	799.99
Yamaha 01X Audio Interface and Digital Mixer	779.99

KEYBOARD CONTROLLERS

CME UF8 88 note Master Keyboard with Aftertouch	349.99
Edirol PCR M30 32 note USB keyboard Controller	114.99
E-mu Xboard 25 USB midi keyboard controller	84.99
E-mu Xboard 49 USB midi keyboard controller	99.99
Korg Pad Kontrol	134.99
Novation Remote 61SL	399.99
Novation Remote Zero SL	228.99
Novation Remote37 SL	349.99
Novation X Station 61	399.99

SAVE

Yamaha MO8 88

88-Key Music Production Synthesizer

was £1349.99

£1239.99

Alesis Fusion 6HD

61 Note Semi-Weighted Keyboard Workstation

was £999.99

£799.99

Korg Radius Rack

Analogue Modelling Synth and Vocoder

was £899.99

£799.99

Alesis Multimix 12FX

Compact 8-Channel mixer with 100 28-bit digital effects.

great deal!

£299.99

Edirol MA7 7

Stereo Micro Monitors. Commanding performance!

PAIR was £74.99

£64.99

SAVE

Adam P11

Nearfield Powered Studio Monitor

in stock now

£474.99

Mackie HR824

Active Monitor. The stereo sound field is wide, deep, and incredibly detailed.

PAIR

£899.99

NEW

Focusrite Liquid MIX

32 channels which provide EQ and Compressor emulations.

in stock now

£498.99

AKG C414B XLS

Large diaphragm condenser microphone.

was £599.99

£529.99

NEW

Clavia Nord Stage 76

Featuring the best of Clavia's technologies.

in stock now

£1999.99

Emu 1820M

24-Bit/192kHz Digital Audio Interface

while stocks last

£229.99

BUNDLE

Waves SSL

Includes three meticulously modeled plug-ins.

while stocks last

£649.99

IK Ampeg SVX

The first bass amp plug-in with a real signature sound

in stock now

£268.99

NEW

Focusrite Saffire LE

6in/8Out Ultra-low Latency Firewire interface.

in stock now

£219.99

MOTU 828mk2

USB2 Audio and midi interface. Total of 22 inputs and 20 outputs!

save save save

£479.99

SAVE

Edirol PCR M1

Compact, powerful USB MIDI Keyboard Controller!

was £94.99

£89.99

www.andertons.co.uk ONLINE STORE

TEL: 01483 456733 MAIL ORDER HOTLINE

GUILDFORD STORE OPEN 7 DAYS A WEEK!

58 & 59 Woodbridge Road. Guildford. Surrey GU1 4RF

Tel: 01483 456777. Education Dept: 01483 456888. Fax: 01483 456722

VISA · MasterCard · Solo · Delta

Most credit cards accepted

All prices include VAT. E&OE.

WRH

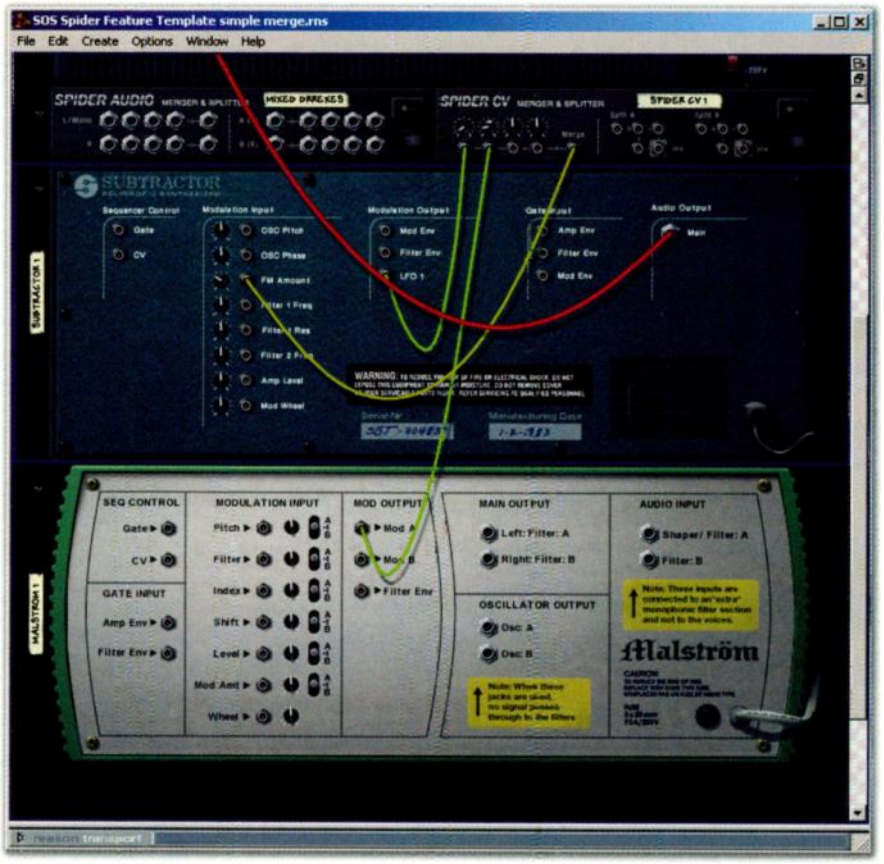
- uses just a kick, snare and hi-hat — and create a synth part with the *Matrix* (say a bass line), to play on the *Subtractor*.
- Now take a patch lead from each of those three *Redrum*-voice gate outputs on the back panel and connect them to the *Spider* merge circuit.
- Grab the gate patch-lead from the *Matrix* and link it to the remaining empty merge input jack. Connect the merged output to the *Subtractor* gate input.
- Start playback with the merge input-sensitivity knobs at zero and gradually turn them to the right. The result of changing the 'level' of each merged gate input is to create playback patterns that are very different from the basic Matrix pattern.

You'll find, with patches that have a longer release setting on the amplitude EG, that some notes will be re-triggered, adding even more rhythmic interest and creating parts that you may never otherwise have thought of. The pattern will be closely related to the *Redrum* pattern but careful use of the sensitivity controls will stop the result from sounding obvious. Keeping the sensitivities low creates a subtle effect that's more texture than trigger. The other side of the coin is that any merged gate input with its sensitivity whacked fully to the right produces a hard, velocity-like hit each time it fires. I find this particular merging technique works great on octave bass-lines with chuggy patches that have moderate amplitude EG release values.

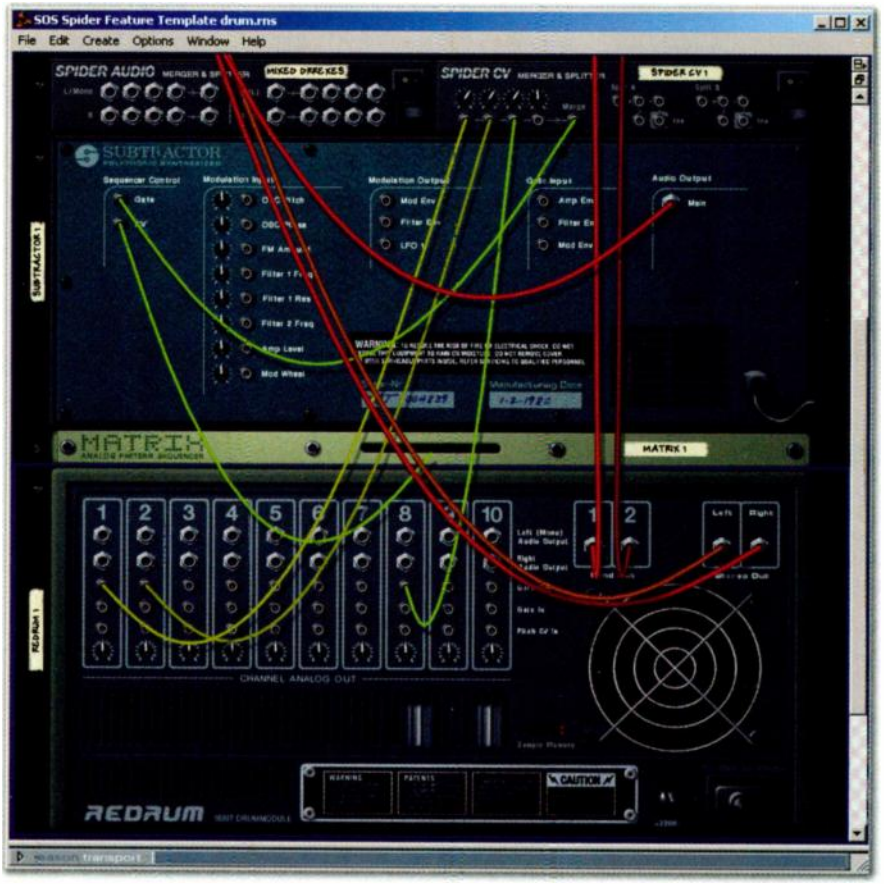
The only problem is that the sensitivity controls at the rear of *Reason* devices can't be automated! If you like several of the different variations of patterns that are produced, you could try to recreate them within the *Matrix*, or bounce them to disk as short audio files for importing into a *Reason* sample playing device. You could also create a *Combinator* patch of the *Redrum*/*Matrix*/*Subtractor* (or whatever) set, and add multiple *Combinator*s to the rack. Load the new Combi into each, set the *Spider CV* merge-sensitivity knobs in each *Combinator* and automate *Remix* mutes to enable and disable the variations you want.

Finally

Your adventures in merging can become even more esoteric when you mix Gates *and* CVs in one *Spider CV* merge circuit. The output could be routed to gate or CV inputs of target devices. You may not have though of doing so before, but it's worth knowing that LFO and modulator CVs can, in many cases, actually trigger other devices, although pitch information still needs to come from a *Matrix* or linear sequencer track. The result can be hit and miss, but the hits are worth working for. SOS



Here, the *Malström* is providing an extra modulator for *Subtractor*, but it's being mixed with the *Sub*'s own LFO1 output, creating something more complex with a simple cabling setup.



The gate outs of three *Redrum* voices, which are happily playing their own parts, are merged to provide the triggers for a *Subtractor*, which is deriving pitch information from a *Matrix*. The *Matrix* is folded because my screen wasn't big enough to show the whole example!

UNIVERSAL AUDIO

Tube & Solid State Class A Mic Pres · Channel Strips · Classic Compressors · 192K A/D & D/A · DSP & Plug-Ins

NEW! UAD-1e PCIe DSP Card *Express PAK & Expert PAK*

- *Cross-platform (Mac/PC) PCI Express x1 card*
- *UAD-1e & UAD-1 in one system, up to four cards*
- *Customizable plug-in bundles, vouchers included*

UAD$100 voucher
when you expand your UAD-1 system
details: uaudio.com/promo
expires Aug. 31

- *Exclusively Neve licensed/UA modeled*
- *Includes "DSP Lite" high-instance 1073SE*
- *Most widely used EQ in popular music*

NEW! NEVE 1073 & 1073SE EQ *Exclusively for UAD-1e & UAD-1*

SOLO Series *Microphone Preamplifiers & DI Boxes*

- *SOLO/610: Award winning 610 mic pre circuit*
- *SOLO/110: flagship Precision Class A circuit*
- *UA sound & build quality at a project studio price*

UAD$100 voucher
for each SOLO you register
details: uaudio.com/promo
expires Aug. 31

analog ears | digital minds

www.uaudio.com

WRH

DP's New Instruments

The Bundled Synths In *DP5*

Last month, we passed briefly over the flashier additions to the latest version of *DP*, in favour of some less obvious but rather useful new features. Now it's time to return for a closer look at those shiny new bundled instruments...

Robin Bigwood

The inclusion of a set of software instruments in *DP5* seems to have been a smart move on MOTU's part: *DP* no longer looks like an 'incomplete' package, especially for those who are just starting out in sequencing. Of course, MOTU have been in the virtual instrument game for a while, and it's fair to say that the *DP5* synths don't really challenge *MX4*, *Mach Five* and *Symphonic Instrument* in terms of features and breadth of usability. But the six newcomers are all good in their own way, and they pull off the trick of being accessible and easy to use while being surprisingly flexible and 'deep'. Here's a round-up of the individual instruments' facilities and quirks, followed by a look at how sample and patch management works, and the way in which the instruments can take advantage of existing *DP* features to expand their capabilities.

Bassline

In terms of its appearance, *Bassline* can't seem to decide whether it's trying to be a Minimoog or a Sequential Pro One. Either way, this is a very simple synth: just one oscillator, a resonant filter with a decay envelope, and an amplifier stage, again with a simple decay envelope. There's no LFO, and there's no other response to using the mod wheel on your controller either. However, the filter and amplifier both have adjustable velocity sensitivity (where the zero setting equals no sensitivity) and there's adjustable overdrive, for those TB303 moments!

There are some nice touches, though. The Waveform knob adjusts smoothly between sawtooth and square waves, but there's a pronounced octave difference between the two (with the square wave lower), allowing for a rather greater range of harmonic variation than you might first think. At a 50:50 setting you'd be forgiven for thinking there were two oscillators, tuned an octave apart. There's also a detune knob, which, again, mimics the sound of two oscillators playing slightly out of tune, producing a fatter sound with a touch of stereo width.

The new *Bassline* synth: simple, but surprisingly pleasing.

Polysynth

Polysynth looks a lot like a Roland Juno 106 and again, in the grand scheme of things, it's a simple synth. Just one oscillator is apparently on offer, but it offers a mix of waveform types: triangle, sawtooth, rectangle and two square-ish sub-oscillators, sounding one and two octaves below the playing pitch. Just one envelope generator modulates the amplifier section and optionally the filter, while a single sine-wave LFO can modulate oscillator pulse width (of the rectangle waveform), pitch or 'wah' (filter frequency). An effects section offers distortion and/or chorus, applied after the amplifier section.

Polysynth will probably not be your weapon of choice for serious sound design, but it's one of those synths that sounds really good despite the simple architecture. The sub-oscillators can lend some serious welly, and the combination of pulse-width modulation and chorus help it to pull off some good swishy string-machine-type patches. You need to watch processor use though: all those different waveform-type sliders are the outward appearance of separate oscillators. If you were to push up Tri, Saw, Rect, Sub1, Sub2 and Noise, and then play one note, you'd be playing six oscillators! Play a four-note chord and all of a sudden it's 24 oscillators — and a chunk of your available processor power will ▶

Polysynth, a single-oscillator polyphonic instrument offering a choice of waveforms, and pan control that allows you to continuously vary its output from full stereo right down to mono.

inevitably have disappeared. Also, *Polysynth*'s stereo oscillator detune is very pronounced. A noticeable 'click' occurs as you push its slider up from the zero 'unison' position and the stereo image becomes very wide. If this is too much, try using a *Trim* plug-in in a plug-in slot following *Polysynth*, and adjust the stereo width there. This is a useful technique for any synth or other signal that is getting too 'wide'.

Proton

Compared to *Polysynth*, *Proton* steals very little processor power. It's a strange little synth, using the Frequency Modulation (FM) approach that lay behind Yamaha's legendary 1980s DX synths, but implemented differently. If you like warm, rich analogue synths you'll probably hate *Proton*'s rather plastic-sounding and often edgy digital character. However, FM sounds do have a certain charm, and Proton offers many opportunities for creating strange digital bleeps, clicks and hisses that, when sampled and treated further, are just the thing for ultra-rarefied glitchy electronica.

Without getting bogged down in FM theory, you can think of *Proton* as effectively having just one audible oscillator — called the Carrier — that, left to its own devices, produces a sine wave. The basic pitch is controlled by the Carrier section's Harmonic and Fine knobs. To create more complex sounds, you modulate the Carrier with another oscillator, called the Modulator. This also has variable pitch (courtesy of its own Harmonic and Fine knobs), and has access to a range of waveforms controlled by the Wave knob. You never hear the Modulator directly — only its effects on the Carrier. The amount of influence the modulator has over the Carrier is controlled by the central FM (amount) knob.

There are, then, three main ways to create more complex timbres. The first is to simply run the carrier and modulator at different pitches, by adjusting their Harmonic knobs. The results caused by different combinations are rather hard to predict, but the Spectra display can give you a lot of detail about the resulting harmonic content. Second, the FM knob intensifies any harmonics produced by the frequency modulation of the Carrier by the Modulator, generally making the sound louder and brighter. Finally, changing the Modulator Wave adds yet more harmonic complexity. Don't overlook the Fine knobs: quite different from their normally subtle equivalents on a subtractive synth, these can have a profound effect on the sound, often leading to inharmonic and non-musical results. Selecting the Fixed option for Carrier pitch can also be surprising, causing every key on your keyboard to produce a different sound, often reminiscent of results achieved with a ring modulator.

All other parts of *Proton* are synth-like modulators — an LFO each for the Carrier and Modulator oscillators, plus envelope generators for modulating pitch, FM amount and the amplifier. There's no filter, but there's nothing to stop you using an audio plug-in such as *Multimode Filter* to treat *Proton*'s output.

Modulo

Up against the minimalistic *Bassline*, *Polysynth* and *Proton*, *Modulo* seems an absolute beast. In fact, it's a kind of cut-down version of MOTU's *MX4* and still retains some of its best features — flexible, wavetable-inspired oscillators, a multi-mode filter and a modulation

Proton is about as friendly as FM synthesis ever gets. As you tweak the Modulator and Carrier parameters the central 'Spectra' display indicates the resulting harmonic content of your sound, with blue lines indicating 'in tune' harmonics, and orange lines the more edgy inharmonic components of the timbre.

Digital Performer News

As I write, there's still no sign of an Intel-compatible Universal Binary version of *DP5*. However, MOTU have released a version 5.01 update which promises to clear up some MIDI playback problems and provide some 'optimisations' (whatever that means) for the *DP5* instruments. It's available as a free download from www.motu.com. Some users have reported strange behaviour after installing the update. This behaviour stopped after they restarted the Mac (rather than just logging out and then in again), so restarting is probably a good thing to try if you encounter any problems.

Universal Binaries *are* available for MOTU's *Symphonic Instrument* and the newly released *Ethno* sound library/instrument. Both are available for download from the MOTU website (as above).

The professional palm-size recorder.

MicroTrack 24/96

record | transfer | edit | share

palm-size

The MicroTrack 24/96 is the new palm-sized 2-channel recorder that lets you record professionally on your terms–anywhere, anytime. This rugged device delivers all the sound quality of our critically acclaimed audio interfaces and preamps–all the way up to 24-bit/96kHz. There's even phantom power for your favorite condenser microphones. Use CompactFlash cards to record song ideas, practice sessions, jams, gigs, field samples and board mixes. Then drag and drop your files to your music app or post them right to the Web. The MicroTrack 24/96 is smaller, lighter and records longer than any 2-channel recorder on the market.

- 2-channel WAV and MP3 recording up to 24-bit/96kHz
- records to convenient CompactFlash or microdrives
- immediate USB 2.0 file transfer to PC/Mac
- balanced ¼" line/mic ins with phantom power
- S/PDIF digital in; RCA and 1/8" headphone monitor outs
- record up to 5 hours per battery charge

GET M-POWERED

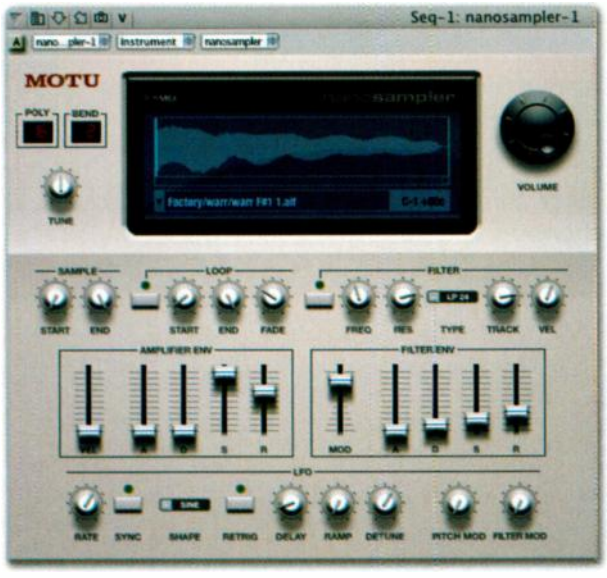
M-AUDIO

www.maudio.co.uk

M-Audio, 6th Floor, Gresham House, Clarendon Road, Watford, Herts, WD17 1LA tel: 01923 204010 email: info@maudio.co.uk

Modulo is a cut-down version of MOTU's stand-alone *MX4* synth and is the most complex and flexible of the new bundled instruments. Note the modulation matrix at the lower right of its window, here showing the modulation destinations for LFO1.

Modulo has a more sophisticated patch-management system than the others. Shown here is its File pane.



Nanosampler, a conventional and very simple sample-player that can nevertheless be a real problem solver.

▶ matrix. The basic architecture is nothing special, comprising two oscillators and a noise source, which pass into a mixer before hitting the filter and amplifier, but the modulation is flexible: the two LFOs can modulate multiple oscillator parameters, as well as filter frequency, and there are three envelope generators, of which one can control oscillator mix, pitch, phase and symmetry, just like the LFOs. The modulations are set up in a dedicated matrix section, so the five modulation sources that aren't 'hard wired' — LFO 1 and 2, the modulation envelope, the modulation wheel and key velocity — can be routed to multiple destinations simultaneously, and in differing amounts. This means there could be well over 30 individual modulations occurring at any one time in the most complex of patches, which is rather impressive. Even given this potential level of complexity, the modulation matrix is easy to work with. Just click on one of the five 'Source' buttons and then dial in modulation amounts by dragging on the red 'bar graphs' for each destination. The source buttons glow yellow to indicate that they are currently applying modulation to a source, and red when they're the source currently being displayed and edited.

Nanosampler

In these days of software-based mega-samplers with highly sophisticated features, who wants a sampler that can only load one sound?

That's all *Nanosampler* does, though it does have some basic sample-manipulation and looping options, and a simple synth architecture. It's all extremely simplistic — and there, in my opinion, lies its strength. *Nanosampler* doesn't try to be *Mach Five* or *Kontakt* and so remains very immediate and useful, both as a conventional sampler and as a little problem-solver.

Clearly, *Nanosampler* has some use as a musical instrument. It's good for electronic sounds such as synth basses or pads, because these don't tend to suffer badly from the changes in duration and transposition of harmonics associated with changing playback speed. But even for voices and acoustic instruments it can work. It's capable of convincing looped backing-vocal 'aahs', for example, and some acoustic instruments can take a surprising amount of transposition before they start to become implausible. Think outside the box, though, and *Nanosampler* offers a lot more.

To start with, you can view *Nanosampler* as a sort of MIDI-triggerable soundbite player. For example, you might choose to use it for drum replacement, or repeatedly playing a loop or riff without having to laboriously duplicate and place it in the Sequence Editor. Loading a soundbite is trivially easy. You just drag and drop a soundbite from the Sequence Editor or Soundbites window to *Nanosampler*'s waveform display and immediately it becomes playable.

Also not to be overlooked are *Nanosampler*'s varispeed capabilities. *DP* has been criticised in the past for offering pitch and time manipulation for soundbites ▶

A R T
APPLIED RESEARCH AND TECHNOLOGY

TOOLS OF THE TRADE

ARTcessories™
Creative audio solutions in cool little boxes.

DPSII
Digital Tube Preamp

Features two channels of award-winning ART tube preamplification. The DPS II adds ART's critically acclaimed 24 Bit – 96kHz A/D converter – for warm, vibrant and clean digital audio. The DPS II features ART's V3™, which delivers a wide range of professional sounding presets.

£299.00 incVAT

SP 4x4
Metered Power Distribution System

Features a backlit linear meter to monitor input voltage, it has eight total power outlets to choose from and a surge and spike protection system. The unit also has RFI and EMI filtering, a 15 amp breaker, and a total power capacity of 1800 watts.

£99.00 incVAT

TCS
Twin Compressor System

The TCS is a two-channel high performance compression system which features two types of compression per channel utilizing ART's Dual Topology technology.

£299.00 incVAT

HQ231
Pro Dual 31-Band EQ

Features ART's FDC™ Feedback Detection Circuitry, individual high and low pass filters on each channel. Applications include live sound systems/PA, permanent or fixed installations, club and practice PA systems, DJ sound systems, home and project recording systems, and monitor systems.

£295.00 incVAT

HeadAmp

The HeadAmp provides amplification and individual volume control for up to 4 headsets. It can be driven by either line-level or headphone amplifier signals (mono or stereo).

£49.99 incVAT

USB Phono Plus

Acts as an interface between your computer and a wide variety of analog and digital sources. The analog inputs are switchable between RIAA Phono and Line level signals and a low cut filter removes rumble and wind noise while leaving the audio pristine.

£79.99 incVAT

CoolSwitch

The CoolSwitch provides full A/B-Y switching functionality, plus parallel common in/out jacks. It lets you switch a common source signal between two amplifiers or amp channels. Or conversely, it lets you switch two source signals into the same amplifier.

£29.99 incVAT

Tube MP Studio V3™

Features V3™ Technology (Variable Valve Voicing), which provides optimized reference points to begin the recording process. V3™ delivers a wide variety of settings designed for various microphones and DI inputs, including guitars, basses, keyboards, synths, electronic drums and much more. Don't settle for a copy-cat look-alike. Go with the real thing - the Tube MP by ART!

£109.00 incVAT

SLA2
200W Power Amplifier

A studio linear power amp that delivers 200W a side of clean, clear power and will reproduce your mixes with near pinpoint accuracy inside a single rack space.

£319.00 incVAT

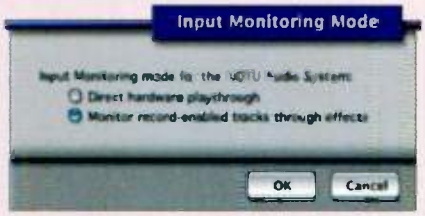
...JUST ADD INSPIRATION

SONIC 8
www.SONIC8.com
08701 657 456
exclusively distribute

ACOUSTI PRO
creamware™
A R T
APPLIED RESEARCH AND TECHNOLOGY
fxpansion
algorithmix
ADVANCED DSP TECHNOLOGIES
DATEQ
audio technologies
YELLOWTOOLS

DP5 Feature Spotlight: Input Monitoring

Digital Performer **has always offered two distinct approaches to input monitoring, configurable via Setup menu / Configure Audio System / Input Monitoring Mode.**



If you monitor your input signals in *DP* you have two basic choices: use effects but suffer latency, or forgo them and achieve much lower latency. This dialogue box is where you make that choice.

'Monitor record-enabled tracks through effects' allows the input signal to be processed by MAS and Audio Unit plug-ins before being monitored, so if you were recording vocals you could give them some reverb in the headphone mix. The problem with this approach is that the monitor signal is afflicted by the latency that results from having to be routed through the MOTU Audio System. Exactly how much latency is dependent on the sample rate you're using and the Buffer Size setting in Setup menu / Configure Audio System / Configure Hardware Driver. Divide buffer size by sample rate to get a rough idea, in seconds. You need a fast computer to run much of a mix and still be able to monitor through effects with low latency.

This is why the other option, 'Direct hardware playthrough', exists. With this selected, the input signal to be monitored takes a shorter journey through your computer, picking up much less latency, independent of Buffer Size. However, it can't have any *DP*-hosted effects applied to it.

Other monitoring options exist, of course, including using an external mixer, and taking advantage of the zero-latency monitoring of interfaces, such as MOTU's own Cuemix facility, found on its PCI and Firewire audio hardware. However you choose to monitor, there are a number of things to watch out for in *DP*, and some new things that were introduced in *DP5*.

MOTU call routing an Audio Track's input directly to its output, for monitoring purposes, 'Audio Patch Thru'. In earlier versions of *DP*, this was enabled and disabled in the Studio menu, or by clicking the little 'headphones' button that appeared in the title bar of the Consolidated Window (and Audio Monitor window, when it was popped out of the Consolidated Window). This worked fine, but only for tracks that were record-enabled, and even then only 'globally' — you couldn't configure Audio Patch Thru status for individual tracks. To cap it all, there was a further option in the Setup menu's Input Monitoring Mode dialogue box that determined how pre-existing track audio was incorporated with inputs being 'patched thru'. (See *Performer* Notes from December 2005, which is available online at http://www.soundonsound.com/sos/dec05/articles/performernotes.htm.) It could be made to work, but I'm sure I'm not the only one who found the whole system cumbersome.

In *DP5*, input monitoring and Audio Patch Thru have, fortunately, been much improved. The old trusty headphone button has gone, because Audio Patch Thru is now handled not globally, but on a track-by-track basis. To turn on input monitoring for a track, do one of the following:

- **Click in the new 'Mon' column in the Tracks window, to illuminate a blue 'speaker' icon for the track.**
- **Click the new 'Input' button on the track's channel strip in the Mixing Board.**
- **Click the small 'speaker' button next to the track name in the Sequence Editor.**

Any of these methods will enable Audio Patch Thru independently of the track's record-enable status. This ability really comes into its own when you're working with *Reason* Instruments routed to *DP* via Rewire, or monitoring external hardware synths or effects processors in *DP*. With this sort of setup, you need audio channels that stay 'open', to allow you to monitor the synths and effects while you slowly develop your project. You could use Aux tracks to do this, but since you can't record on them you'd have to create audio tracks to finally record the synths and effects when your project neared completion, and spend time configuring them before dismantling the old Aux tracks. With the *DP5* approach, you can create audio tracks from the word go. You then just enable input monitoring for the tracks while you work on your project, and finally record-enable them when you need to record your synths and effects.

In *DP5*, Audio Patch Thru is selected on an individual, track-by-track basis. There are three ways you can turn on input monitoring for a track: click in the 'MON' column in the Tracks window; engage the Input Monitor button in the Mixing Board; or click the 'speaker' box by the track name in the Sequence Editor.

DP5 **provides four modes that control how Audio Patch Thru of inputs is integrated with audio that already exists on the audio track during playback or an Auto Record (punch-in/out) session. They're chosen from Studio menu / Audio Patch Thru, and described on page 213 of the *DP5* manual, but I like to think of them according to typical use:**

- **If you want track inputs to always take precedence over existing audio, choose Input Only.**
- **If you want to hear track audio during playback, but your input during recording, choose Auto.**
- **If you want to hear track audio and your input together (a good idea in the lead up to an Auto-Record punch in, for example) choose Blend. In these three cases you'll only hear your input when you actually record.**
- **The remaining Audio Patch Thru mode (Off) kills Audio Patch Thru completely, which is essential when you choose to monitor your inputs externally (ie. via a mixing desk or using Cuemix).**

▶ (via the Spectral Effects window) but not true varispeed. Now soundbites can be loaded into *Nanosampler* and the 'Tune' knob used to provide a true varispeed effect over a four-octave range — great for drum loops, of course. Also, if you play a sample and simultaneously use pitch-bend, the much sought-after 'tape stop' (or indeed 'tape start') effect can be created, especially when pitch-bend is set to its maximum value of 12 semitones.

Model 12

Model 12 is an easy to use drum-sound module with clever pitch-shifting that doesn't affect duration, and time-stretching that doesn't affect pitch. It can load 12 drum sounds, each of which can be individually tuned, stretched, filtered and, as I described last month, sent to separate audio outputs. *Model 12*'s basic operation is pretty well covered in the manual, but what's not made so obvious is the modulation options available for various parameters.

The central, grey-coloured pane in *Model 12*'s window contains controls that relate to the individual drum parts beneath. If you click

The *Model 12* drum-sound module. As well as being able to load 12 drum sounds and tune, stretch and filter them individually, *Model 12* offers modulation facilities that make interesting drum parts easier to produce.

and drag the Tune knob, you can tune an individual drum over a two-octave range, but you might also notice that some further options appear in the display section above as you do so. These are modulation sources for the same Tune parameter, and they provide a way for key velocity, a random value generator and a decay envelope to control pitch in real time. You simply click and drag in their bar graphs to dial in the desired amount of modulation or make other settings. Other parameters that offer these modulation options are the sample Start parameter, Volume (which inevitably is modulated by key velocity), Pan, Filter Cutoff, Resonance and Drive. You might try introducing a little randomness into the sample Start and Tune parameters for a programmed hi-hat part, say, to make it sound a little less clinical, or use the decay envelope in the filter to create electronic-sounding versions of otherwise straight acoustic drum sounds.

Presets & Patches

All *DP5*'s synths, except for *Modulo*, use their window's title-bar mini-menu to load and save patches. *Modulo*, on the other hand, uses its own bank and patch management system, accessible both from the usual front panel (for loading patches) and from the File button at the bottom left of its window (for managing them). *Nanosampler* and *Model 12* also offer extensive facilities for managing the samples that they use, and offer the user an opportunity to build their own personal sample library. More on this, and on how all the synths integrate with *DP*'s wider mixing environment, next month. SOS

Getting Started With Surround Sound

In *Logic*

The facilities for surround sound in *Logic Pro* v7 are often overlooked, so here we take a peek at how to get going.

Russ Hepworth-Sawyer

By choosing the Surround 5.1 template, you cause *Logic* to create a pretty comprehensive set of audio and instrument tracks (all with surround sound panners) specifically for laptop or larger screens.

Whether it is producing music for film or television, or simply creating an SACD mix of your latest stereo creation, we will all at some point consider trying surround sound. Provided that you have at least four audio outputs on your interface, and as many speakers, *Logic Pro* is great at handling most surround-sound work straight out of the box. But first we need to make all the necessary connections...

Configuring *Logic* For Surround Sound

Before we can go ahead and mix, we need to configure the outputs to connect with the appropriate speakers. By clicking on the Audio menu in the title bar and choosing Surround, the Surround Preferences appear, saving you having to go Logic Pro / Preferences / Audio / Surround. Immediately you are drawn to the routing matrix for up to nine speakers for more detailed surround-sound mixing — more of this in a moment. Working from the top of the pane downwards, the first drop-down menu labelled Show As restricts the configurable outputs on the routing matrix to the system you are working with, for example LCR (Left, Centre, and Right), Quadrophonic, 5.1 EX, Prologic, or other permutations. For the purposes of surround-sound music mixing we would commonly choose 5.1.

Below this drop-down menu, there are three button choices for automatically routing *Logic*'s outputs to the respective speakers via your audio interface. These are set to three standards: the first, labelled

Logic News

One of the surprise features introduced by Apple with *Logic* v7.2 was direct support for Serato's highly regarded *Pitch 'n Time* plug-in. This program has long been in use in the Pro Tools TDM world for high-quality manipulation of the pitch and tempo of audio, and its relatively high cost has been justified by the speed and quality of the results obtainable. The announcement that a version was being made available for *Logic* was a pretty mouth-watering one; a dribble that turned to a torrent when rumours surfaced that the proposed price would be just $99 as opposed to TDM's hefty $799. As it turns out, when the product was finally released, its price was set at a more logical $399. This is still something of a bargain, though it does come with the caveat that you'll need an iLok security dongle to use the program. If you have an iLok already, there's a demo version of the software on Serato's web site, along with audio-only examples for the dongle-less. There's been a modicum of confusion as to the program's implementation under *Logic*. Although it's technically an Audio Unit plug-in, it actually appears as an extra algorithm in the Arrange page's Audio menu and in the Time Machine, so you use it in exactly the same way as you would *Logic*'s own pitch- and tempo-manipulation features.

Interestingly, it's not just Serato that Apple have allowed to access *Logic*'s pitch and time core; Izotope, makers of the excellent *Ozone* mastering suite, have announced *Radius*, a plug-in which integrates into *Logic* in exactly the same way as *Pitch 'n Time*. It's slightly cheaper than Serato's offering (at least for the first few months), doesn't require an iLok, and you can try the demo out at their web site. The opening of *Logic*'s core to third-party software houses is an interesting move, and we can hope that other manufacturers take advantage. Of course, many of us are wondering why we have to pay extra for high-quality pitch- and time-manipulation anyway, when other programs, notably Ableton's Live, already have it as an integral part of the software. However, the thought of Celemony's *Melodyne* pitch-processing algorithms possibly becoming available for use directly in *Logic*'s Time Machine is certainly an exciting prospect.

There's one downside though. The way *Logic* currently handles pitch- and time-manipulation is clunky and counter-intuitive. While you can tempo-change audio files directly in the Arrange page, how it's actually performed isn't very obvious — and the Time Machine is so obtuse that I know many a seasoned *Logic* user who has never ventured into that particular backwater. What *Logic* really needs is some way of tempo-stretching audio sequences directly in the Arrange page with the mouse. For pitch changes, you should be able to select a sequence and, using the desired algorithm, transpose it using the Parameters box Transpose feature in a similar fashion to the way *Logic* handles MIDI and virtual-instrument data. *Stephen Bennett*

W www.izotope.com
W www.serato.com

Surround Monitoring Control

Monitoring control systems for surround sound are expensive devices, and are usually provided separately to that of your normal audio interface or mixing console — one example is the SPL Model 2380 we reviewed back in *SOS* August 2003. Fortunately, the recent addition of the Master Volume fader in *Logic* offers a solution to this, and will act as an overall volume control for your surround-sound system without you having to invest in any other hardware.

Default, is *Logic*'s own preference (1 Left, 2 Right, 3 Left Surround, 4 Right Surround, 5 Centre, and 6 Subwoofer). Other choices include the ITU (International Telecommunications Union) standard used by many professionals, and the WG4 standard set by the DVD Forum for DVD-Audio. These three selections will route your 5.1 audio out of your audio interface in slightly different configurations and will therefore determine the connections you make to your amps and speakers. However, it is quite possible for you to alter the setup to suit whatever connections you have made to your speaker system by assigning each output from the individual drop-down menus. There are also options to customise the file extensions added to each of the audio files created when you bounce down your surround mix.

Getting Surrounded

There are two ways to get started. The first assumes you wish to start from scratch with a 5.1 or 7.1 session by selecting one of the Song templates offered to you as you start a new project. Choices include 5.1 and 7.1 setups for both 15-inch Powerbook and 20-inch Cinema displays.

By choosing a template, the routing and surround panners are already provided for you on each channel, ready to go. You will notice that the group outputs remain as stereo pairs (1+2, 3+4, 5+6, and so forth), which is fine for *Logic*'s Default routing as 1+2 are Front Left and Right, and 3+4 are Surround Left and Right. However this then groups the output for the Centre and LFE (Low Frequency Effect — the subwoofer to you and me) channels, which may limit the way in which you handle the discrete surround-sound outputs. For example, you may wish to filter out any high frequencies ▶

Have Your Say!

If you want to suggest changes or improvements to *Logic*, then here's your chance! The Apple development team are inviting *SOS* readers to send in their suggestions of what they'd most like added or changed in *Logic*. Email your top five suggestions (in order of preference) to logicnotes@soundonsound.com, and we'll forward your lists on to the *Logic* team. We'll be asking them for feedback on which changes users deem most important and how these might be addressed.

Current Versions

Mac OS X: Apple *Logic Pro* v7.2
Mac OS 9: Emagic *Logic Pro* v6.4.2
PC: Emagic *Logic Audio Platinum* v5.5.1

Logic handles surround-sound automation using Angle and Diversion parameters and Centre and LFE channel level controls.

▶ emanating from the LFE buss, which would not be possible as part of a stereo group without affecting the other channel. Should you want to do this then it is easy to change this by deleting these stereo groups and following the procedure outlined below.

The second way to get surrounded is to alter an existing stereo mix to output to your surround system. Presuming that your mixer and routing within *Logic* are quite standard, you are likely to have an output buss labelled Out 1-2. In order to talk to the surround-sound speaker system, we need to delete this buss and create some further Audio objects to represent each of the discrete outputs. To do this within the Environment page choose Audio Object from the page's New menu option. Repeat this five more times and label each one (Left, Right, Left Surround, Right Surround, Centre, LFE). Select each Audio object in turn and assign it to outputs one to six (this follows *Logic*'s own Default connections for surround as described earlier) or use your preferred configuration. Before expanding each channel to reveal the traditional channel strip, it is likely that you will not require all the parameters available to you. As such, select the six objects and untick the Show boxes in the Parameters. Next double-click the Audio objects to reveal their full strip. Now, should you need to tailor each of the outputs, you can add EQ and inserts as required by ticking the appropriate Show box.

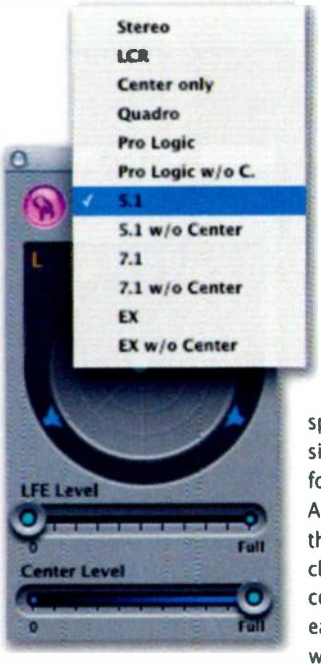
Panning Around

To make your stereo mixer channels output to the surround mix busses, there are changes that need to be made. On the channel strip in *Logic* go to the output list, usually listed as Out 1-2, click this, and choose Surround. (This can be done somewhat faster for the whole mix by selecting all the channels you wish to change first and then pressing Alt when clicking to Surround). Automatically a surround-sound panner replaces the conventional stereo pan control, showing the space between the five full-range speakers. Within this sound stage a small circle denotes the intended destination of the sound, and the panning is managed accordingly for you. Double-clicking any of these surround panners brings up an enlarged version in its own window, within which there are two additional sliders for LFE and Centre levels. Pressing Apple when using the mouse in the surround sound panner allows you to lock your sound to a plane, for example front to back, or front left to rear right in a straight line. Pressing Control locks the distance from the centre and allows you to rotate around the centre on a fixed circumference.

You might well ask why there's any need for a separate Centre level control. Well, in many applications of surround sound, such as film, the Centre speaker is often used for dialogue applications, therefore keeping the viewers' focus on the cinema screen and not on the left or right speakers depending on where you sit in the cinema. Music stems might be kept away from the Centre in this instance. By adding this control, you can manage whether the panner is controlling its

LFE Management

The Low Frequency Effect channel is the '.1' part of 5.1, and offers you the ability to add rumble to your film soundtrack, or deep bass to a music mix. The use of the subwoofer might be dictated by your monitoring system, and should not be confused with the LFE channel. It is intended that in a true 5.1 system all five speakers should be pretty much full range, while the subwoofer acts only as a sub-100Hz driver used for special effect — such as when a jumbo jet flies past in the film. In this instance the '.1' channel is intended to receive only information on this channel. However, in monitoring systems for music the subwoofer is often used to improve the bass extension of smaller nearfield monitors, and in some surround monitoring systems bass from the five main channels may also be redirected to the LFE as well for this reason. Be sure that you understand how your monitoring system is handling the bass end, and use the LFE channel for the extra bass only when necessary.



Double-clicking any of the channels' surround panners opens up a larger screen for more accurate control, and also allows you access to the LFE and Centre level faders. A drop-down menu at the top of the panner window lets you choose different output configurations.

centre image through the standard left and right speakers, or whether it feeds signal to the centre speaker for true 5.1 immersion. Another way of managing the Centre content is to choose the output configuration from the top of each of the enlarged panner windows. This can be restricted to '5.1 w/o Centre', which will omit the Centre feed and spread the signal at the front between left and right speakers, as in a traditional stereo system.

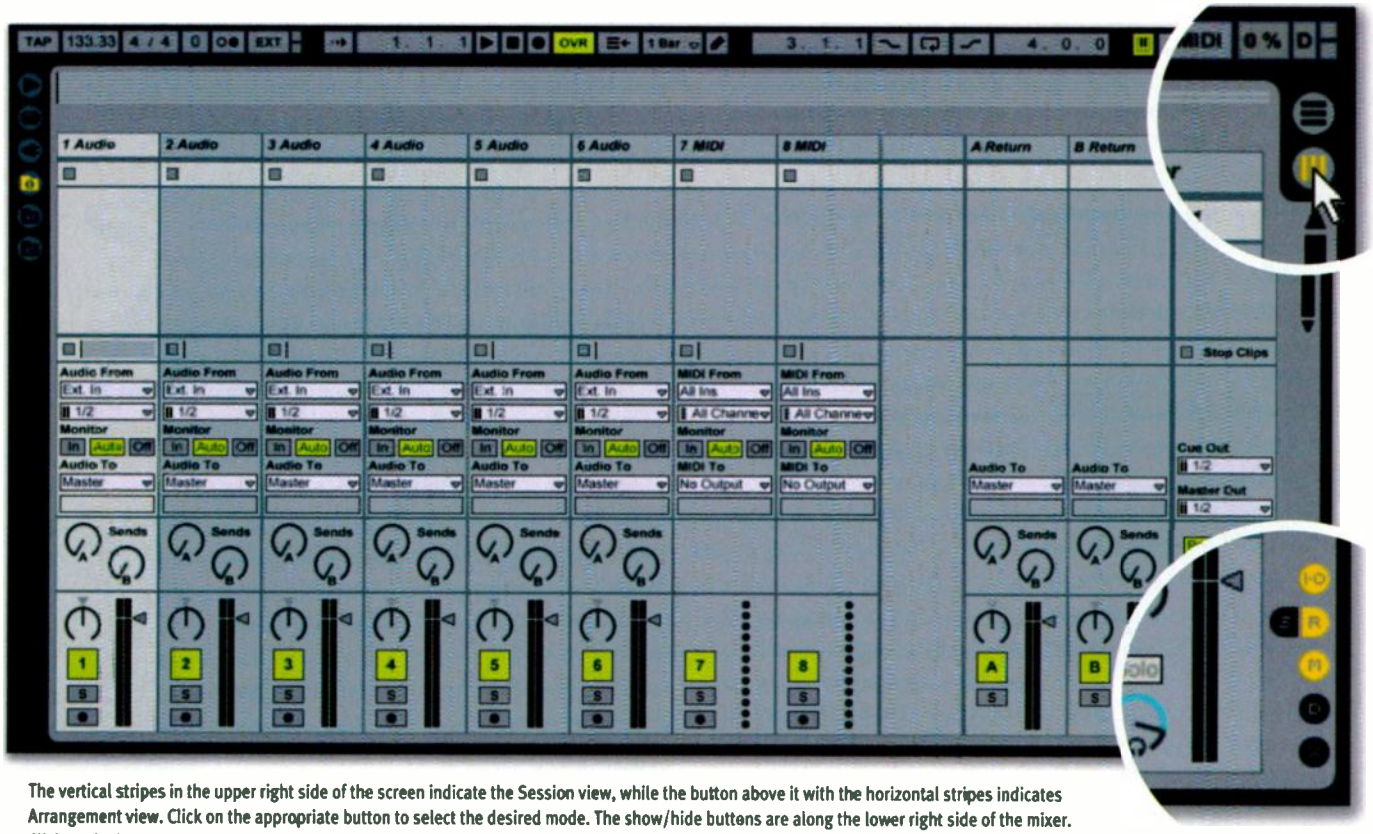
Automation Of Surround Sound

Surround-sound automation is handled in a similar way to the automation of any other control in *Logic*. By changing the status of the automation mode on the channel strip to Write, Touch, or Latch, *Logic* will respond to any panner movements you make as the track plays. Again it is better to open the larger panner window for more accurate placement. These moves can be edited later using the Display Track Automation menu option in the Arrange page, which shows the surround-sound options for close scrutiny. *Logic* offers four automatable parameters, Angle, Diversion, and LFE and Centre levels. The Angle parameter positions the sound around the five speakers, whilst Diversion dictates how far from the centre 'sweet spot' the sound is drawn away. The LFE fader is provided so that you may manage the amount of content sent to the subwoofer independently. For example, in a band recording you may only wish for the bass drum and the bass guitar to be fed in part to the LFE, keeping the bottom end clear of things like synth parts which contain frequencies that sometimes go that low and cloud the mix.

Bouncing Down

The next question is how you finalise your surround-sound mixes for mastering. Bouncing for surround sound is just as easy as it is for stereo. Just select the Bnce button from one of the newly created output-buss Audio objects or select Bounce from the File menu. The same dialogue box will appear as when doing stereo bounces, so you will need to change the 'surround bounce' drop-down menu to the variety of surround you want, in this case 5.1. Notice that the file-size information at the bottom of the page changes according to the format chosen. Your six files can then be burnt to a CD or DVD for mastering to AC3 or another format.

Straight out of the box, *Logic* provides fantastic facilities for producing polished surround mixes for a wide range of applications. Surround is clearly here to stay, so perhaps it's time to have a go for yourself. SOS



The vertical stripes in the upper right side of the screen indicate the Session view, while the button above it with the horizontal stripes indicates Arrangement view. Click on the appropriate button to select the desired mode. The show/hide buttons are along the lower right side of the mixer. Click on the buttons shown in orange to make sure everything you need to see is visible.

Complete Production

In Ableton *Live*

Craig Anderton

Ableton *Live* isn't just a DJ tool, but a very capable audio workstation. This month we begin a look at how to approach *Live* if you want to use it as a fully fledged audio sequencer.

Sit 10 people down in front of *Live*, and you'll see 10 different ways of using the program. A DJ might load mini-songs into Clips and 'warp' the tempos so that they match up, then crossfade between them to create sets. An avant-garde composer may choose to use it mostly for live, on-the-fly signal processing that gets captured and resampled; I use it as a 'remix machine' for doing live arranging with overdubs — sort of a cross between mixing and performing. Others see it as a standard digital audio workstation with some unusual extras thrown in... and that's what we'll address in this column.

The idea for this article came from *SOS* Editor Paul White, who proposed a series of 'defector's guide' articles for those who were switching from one sequencer to another — for example, those who used *Logic* for Windows before Apple dropped support, and felt on unfamiliar ground when trying to climb the *Sonar*, *Cubase* or *Acid* learning curve. They were to cover the fundamentals so those using the new platforms could quickly figure out navigation, where to insert plug-ins, and so on.

As it turned out, I would just about get an article written when a new revision would be introduced, and it seemed I could never catch up. In this case, though, *Live*'s DAW paradigm (like the program itself) has remained internally consistent. Those who are intrigued by *Live* but are most comfortable in coming in via the DAW 'back door' can do so, but it takes a bit of effort to understand the '*Live*' way of doing things. Once traditionalists are comfortable using *Live* as a DAW, they may find it a lot easier to get into *Live*'s many additional features.

Let's Make A Mixer

Live has two main views; Session view and Arrangement view. Session view is *Live*'s unique take on doing on-the-fly looping, composing, and remixing. The Arrangement view follows the 'linear timeline' recording model, much like conventional DAWs. You can go into record mode, then record into the arrangement as you would any DAW track.

However, there's the added bonus that *Live* can also act like *Acid* or *Sonar*, where you can drag loops directly into the Arrangement view. These will be converted into 'warped' Clips that loop, and stretch with any changes in tempo. You can also work in this manner with portions of a recorded track, or even an entire track.

So is the Session view irrelevant? No, because this is where you put together ▶

NOW YOU CAN.

KORE

THE UNIVERSAL SOUND PLATFORM

www.native-instruments.com

NATIVE INSTRUMENTS
SOFTWARE SYNTHESIS

GENERATE THE FUTURE OF SOUND

The track called 'Guitar' is selected and the Track view is visible, which shows the plug-ins inserted into this track — the *Auto Filter* that comes with *Live* and the AudioDamage *FuzzPlus 2* free VST plug-in. Note that the *FuzzPlus 2*'s GUI is available for editing, because the cursor has clicked on the plug-in's 'Edit Panel' button (the one that looks like a wrench).

a mixer. For DAW fans, you can consider the Session view the console or mixer view, while the Arrangement view is the edit view. Here's how you go about creating a mixer in the Session view.

By default a new *Live* Set will include an Audio track, a MIDI track, and two Return tracks (although they may be hidden). Each track has 20 scenes. As we won't be working with scenes if we're using *Live* as a traditional DAW, they aren't needed and just take up space; you can remove them by Selecting All from the Edit menu (Ctrl/Apple + A) and then selecting Cut Scenes, also from the Edit menu. One scene will be left, because all tracks must have one scene.

Live's Mixer has several show/hide options on the bottom right-hand side of the main Session screen. To most closely mimic a traditional mixer view, click on 'IO' to show the I/O section, 'S' to show sends, 'R' to show return tracks, and 'M' to (of course!) show the mixing controls (fader, pan and so on).

Each track in Session view corresponds to a track in the Arrangement view. To create a track, right- or Control-click on any blank space within the Clip/Device drop area, and select the type of track you want to insert (Audio, MIDI or Return). For now, let's create a mixer with six audio tracks, two MIDI tracks, and two Return tracks (which should already exist if you started with a default new *Live* Set).

To delete a track, click on the track name to select it, then hit the Delete key or select Delete from the Edit menu. To move a track, for example to have all the audio or MIDI tracks together, click on the track name and drag it to the desired new position.

Of course, you can add tracks at any time. But for now, you should have a mixer that looks somewhat like the picture on the previous page (this assumes you've hidden the browser to make more space).

The *Live* 'Audio Channel Strip'

The *Live* channel strip works a bit differently to those in other sequencers, but there are also a great deal of similarities. Let's start at the top and work our way downward.

At the top is the track title bar, which contains the track name. To rename, right/Control click on it and type in a new name.

Now comes our first major point of departure: where are the insert slots for plug-ins? *Live* has a separate Track view pane (which is keyed to the selected track) where you can drag effects in from *Live*'s browser. If this isn't visible, the easiest option is to just double-click on the track name; this opens up the Track view and displays any effects that are present for that track. If the Track view is already open, a single click on the track name will reveal the Track view for that track.

Signal flow is from left to right, with audio hitting the input, and the output going to the pan/fader/send section. Interestingly, *Live* has a feature I wish all sequencers would adopt: in the Track view, there's a level meter at the input and output of the chain, as well as between each effect. This makes it very easy to do gain staging with multiple effects.

Unlike conventional sequencers, when one of its a built-in plug-ins is inserted, *Live* automatically shows that plug-in's GUI in the Track view. VST effects are handled differently; there are four interface editing options, depending on which button you click on and select to the immediate left of the plug-in name.

If neither button is on, you'll see an X-Y controller where you can choose the X and Y parameters from drop-down menus. This allows for fast tweaking of important parameters.

If the right arrow button is selected, the effect 'unfolds' toward the right to reveal the various parameters offered by the effect, but presented as linear faders using graphics provided by *Live*. Once selected, this button turns into a left arrow button; click on it to fold the parameters back up.

If the 'wrench' (Edit Panel) button is on, *Live* shows the effect's native graphic interface. This can be enabled with either the parameters unfolded or not, give you two editing variations.

Ins, Outs & Monitoring

The 'Audio From' field selects the signal source. The option that will be most familiar to conventional DAW users is 'Ext. In'. The field below this chooses from whatever input sources are enabled, as chosen under Options / Preferences / Audio / Settings / Input Configuration. For example, if you have an audio interface with four pairs of stereo inputs, and they're all enabled, you'll be able to choose any of the pairs, and most likely either side of the pair (mono) as the

TASCAM's DM-3200.

The digital centre of your universe.

* 48 mixing channels (32 channel inputs, 16 aux returns all @96k/24-bit * 16 busses or two 6.1 surround busses * 8 aux sends * 16 analog mic/line inputs with inserts * Powerful automation with 100 mm touch-sensitive motorized faders * Dedicated DAW control surface layer for Pro Tools®, SONAR®, DP™, Logic, Cubase® and Nuendo® * EQ and dynamics on each channel * 2 built-in effects engines with TC® reverb * Per-channel LED ring encoders for pan, aux sends, EQ and more * 24 channels of TDIF and 8 channels of ADAT* built in * 2 expansion slots for FireWire*, Surround Monitoring, ADAT*, AES/EBU, TDIF and analogue cards * TASCAM Mixer Companion software (Win® XP/OSX) with driverless USB connection to computer for back-up/restore of mixer data * Get the whole DM-3200 story on-line @www.tascam.co.uk or visit your TASCAM dealer today.

TEAC UK LTD Units 19 & 20 The Courtyards, Hatton Lane,
The Croxley Centre, Watford, Herts, WD18 8TE

Brochure hotline..01923 438888 e mail..info@tascam.co.uk

TASCAM
www.tascam.co.uk

The 'Audio From' drop-down menu lists a variety of possible signal inputs to the selected channel.

▶ input source. There's also a quick shortcut to the configuration option: In the Master track's Master Out field, select Configure and you'll jump right to the audio preferences.

Note that *Live* also offers some creative options not necessarily found in typical DAWs, as shown in the above picture.

The Audio From field will show any Rewire devices, should you want to Rewire an output into a *Live* mixer channel. Be aware that selecting a Rewire source will not automatically launch the Rewire-compatible application — you need to do that manually.

Resampling routes the master output back into the channel input. This is like a 'track bounce' function, as you can mute tracks, change levels, panning and so on, to create a particular mix at the master out, then bounce it to the 'resampling' track. It's a great way to build up massive sounds without having to burn up lots of tracks.

The track numbers show that you can route the output of a particular track into the channel, including the Return tracks. This is particularly handy when used with virtual instruments, as described in last month's *Live* Techniques article.

Monitoring choices are 'In' (sends whatever signal is feeding the channel to the output), 'Off' (the input signal doesn't go to the output, although any previously recorded signal does), and 'Auto'. The latter is most like the record monitor function found on multitrack tape decks: when record-enabled, Monitor is set to In and monitors the incoming audio. During playback, Monitor is off, and you hear whatever was recorded. This is probably the option you'll use most often, although once more referencing my previous article, you might do things differently with soft synths.

The Audio To field determines where the track's audio will be sent. External Out sends the signal directly to your audio interface. If selected, the field below it chooses from whatever output sources are enabled, as specified under Options / Preferences / Audio / Settings / Output Configuration. This process works similarly to how inputs are selected, except that you're choosing outputs.

Master is a more conventional option, as it sends the channel to the Master buss. This, in turn, is assigned to a particular audio interface output. You can also send the output to the input of any track except the one you're sending from (well you don't want howling feedback, do you?).

If you have all the I/O set the way you want and don't need to do any further editing (say you're mixing down previously recorded tracks, for example), use the 'IO' show/hide button to hide the I/O section.

Solo, Mute & Record

Live departs a bit from the standard sequencer protocols with send and mute. First of all, there is no mute function, but rather, a 'track activate' button (the big button with the track number). When on, the track is enabled; when off, the track is muted.

Tracks one, two, and three are activated, four is not. Tracks two and three are soloed, so track one's activate button is greyed out. Note that its meter is also grey.

There are Solo buttons, and these are 'additive' in that you can Ctrl or Apple-click on Solo buttons to solo multiple tracks simultaneously. When any solo is active, tracks that are muted but normally activated change colour from yellow to light grey, whereas tracks that are muted but normally deactivated are coloured dark grey. Meters for tracks that are muted are also greyed out, making it very easy to see which tracks are soloed.

The record enable button works exactly as you would expect it too — it arms a track for recording. As with the solo buttons, if you Ctrl or Apple-click on several record buttons, you can record into multiple tracks simultaneously.

Level, Pan & Metering

The Pan control is right above the Track Activate button, and works as these things normally do. The fader works as expected, but has no calibrations; level-setting is something where you really do have to 'play it by ear'.

You can shrink a channel's width to show more channels by grabbing the right side of a track's title bar (the cursor turns into a bracket) and dragging leftward. However, this greatly shortens the fader throw, as it stacks the fader on top of the track activator/solo/record buttons. I wouldn't recommend this unless you're using a control surface and don't really care about what the fader looks like on-screen.

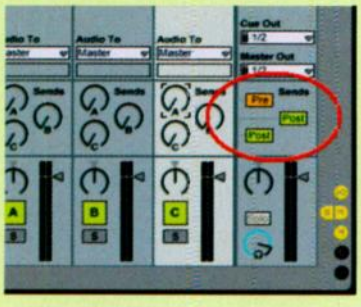
Finally, metering in *Live* has no calibrations, so you don't really know by how much a signal is over or under the maximum allowable headroom. However, the meters do have the helpful characteristic of turning red in the event of distortion, making it easy to see if the track has clipped.

Well that's enough for now; in part two we'll by cover the MIDI channel strip, and then work our way over to the Arrangement view to further explore *Live* as a DAW. SOS

Sends & Returns

Whenever you create a Return Track, *Live* obligingly adds another Send control to the Send section, located between the I/O assignment and pan/fader sections. 'Aux buss' effects are added similarly to other effects, but with the Return tracks. In other words, you select the Return track, then drag the desired effects into the Return track's Track view.

However, when it comes to Sends, *Live* is not as flexible as the average DAW because there is no separate pre-fader/post-fader option for each send. Instead, sends are paired with like-lettered Return tracks — the 'A' send controls feed the 'A' Return track, the 'B' send controls feed the 'B' Return track, and so on. Any letter group can be made pre- or post-fader, with all send controls in that group being either pre or post.



All Sends that belong to a lettered group can be either Pre or Post (the buttons are circled in red). In this screenshot there are three groups: the 'A' sends are pre-fader, while the 'B' and 'C' sends are post-fader. Note that you can send a Return track back to itself if you want to create feedback.

TUBE WARMTH AT 24-bit 96kHz

In a cold hard digital world, benefit from real tube performance at 24-bit 96kHz.

Hundreds of talented artists and producers are already benefiting from the ability to maintain the great sound of analogue recordings while maintaining the flexibility that digital audio provides.

By adding tube based Ivory 2 Series pre-amps and processors into their signal path, fitted with integrated high quality 96kHz digital converters, the otherwise elusive classic sound of the analogue era remains constantly at their finger tips.

Check out who, how and why at www.tlaudio.co.uk
or call Sound Technology on 01462 480000

TL Audio®
PASSIONATE ABOUT TUBES

sound TECHNOLOGY PLC

SIMPLICITY

C01U USB Studio Condenser microphone and NEW Q1U USB Handheld Dynamic microphone with direct USB connection to your Mac or PC.

Now there's a simple way to record vocals and acoustic instruments to your computer in the studio or on location. The C01U and new Q1U combine a seamless plug-and-play USB interface with high quality performance.

For more information please visit www.soundtech.co.uk or call Sound Technology on 01462 480000 or better still, try them out at your local Samson retailer.

SAMSON AUDIO

sound TECHNOLOGY PLC

WRH

Choosing A Limiter

Pro Tools Mastering Limiter Shootout: Part 2

Mike Thornton

We follow up our comprehensive round-up of mastering limiter plug-ins for Pro Tools with a look at hot new products from three further manufacturers.

Since we published our shootout of mastering limiters for Pro Tools in the May 2006 issue, PT users have seen two new mastering limiter plug-ins released from TC Electronic and Massey Labs. We've also had several letters pointing out that the original roundup didn't include Wave Arts' popular *Final Plug* or *Multi Dynamics*, so in this month's Pro Tools workshop, we're going to run a 'take two' shootout comparing all these limiters. In the interests of fair play all round, I'll be using the same comparison procedure and the same test source material; for more details see May's issue, or read the article on-line at www.soundonsound.com/sos/may06/articles/ptworkshop_0506.htm.

Single-band Contenders

TC's *MD3 Stereo Mastering* package includes single-band and multi-band limiters, while Wave Arts' *Final Plug* and Massey Labs' *L2007* are single-band designs, so there are three new entrants in this category.

▪ **Wave Arts *Final Plug***

Final Plug is a look-ahead peak limiter with bit-depth management and noise-shaping dithering. It looks and feels very simple to the Waves *L1/L2/L3* range except that it has a graph section added into the window. This shows a very comprehensive dither and noise-shaping section with a good selection of noise-shaping curves to suit the different sample rates and bit depths. It has a feature where you can turn off the dither noise with a Mute button, allowing you to compare the undithered sound with the dithered output. *Final Plug* also includes a range of presets ready for CD and DVD Audio mastering. Unlike the other products on test here, Wave Arts' plug-ins are not available in TDM format, but they are the only ones of the three that include Audiosuite off-line versions.

▪ **TC Electronic *Brickwall Limiter***

TC's *Brickwall Limiter* and *MD3* were originally options for the System 6000 hardware processor, and software versions have been around for quite a while as part of the Powercore mastering suite, but TC have only just got around to releasing Pro Tools versions of them. *Brickwall Limiter* operates at 48-bit and uses a five-times oversampling algorithm for accuracy. The meters always upsample to make sure inter-sample peaks are detected and measured — the meter scale goes beyond 0dBFS to +3dBFS so you know you are going right over. TC Electronic boast that *Brickwall Limiter* is bit-transparent below its threshold, and there is a green light next to the Threshold control which lights to show when it is operating in bit-transparent mode. However, *Brickwall Limiter* only has basic bit-depth management and dither, with no noise-shaping options available.

▪ **Massey *L2007***

This new limiter comes from Stephen Massey, who used to work for Trillium Lane Labs and has now set himself up producing very cost-effective plug-ins that are creating a lot of interest. Stephen's *L2007* mastering limiter looks excellent value — you get both TDM and RTAS versions for $89 — and is again similar in look to the Waves L-series, except that he has chosen to go for knobs to adjust the threshold and output level. In addition there are two rotary switches, the first of which is called Mode and alters the behaviour of the envelope detector, packaging a complex range of adjustments into a simple four-position control. The Release rotary switch is obvious in its function and settings.

Multi-band Contenders

In the multi-band category we have TC Electronic's *MD3* and the Wave Arts *Multi Dynamics* plug-in.

▪ **TC Electronic *MD3***

The *MD3* plug-in is much more than a multi-band limiter: it has an EQ section with four-band parametric EQ, a normaliser section, a three-band dynamics section with compressor, limiter and expander options, an M&S encode and decode option and a brickwall limiter on the end. Other interesting features include automatic gain make-up and a lookahead delay to help with peak handling. All of this makes *MD3* a very comprehensive mastering tool in one package.

The main window has six pages, which relate to *MD3*'s various sections. It starts with the main page, where you set input levels, M&S encoder and decoder options, and the crossover points between the three dynamic bands. Next we have a page for the four-band EQ section, followed by a Normalise page to help adjust the levels going into the dynamics section. This takes up the next two pages, one for an expander page and the next for a compressor. Finally, the output page includes the brickwall limiter.

- **Wave Arts *Multi Dynamics***

The *Multi Dynamics* plug-in offers up to six bands of dynamics with both compression and expansion, and has a comprehensive graphical interface to display everything that is going on in just one plug-in window.

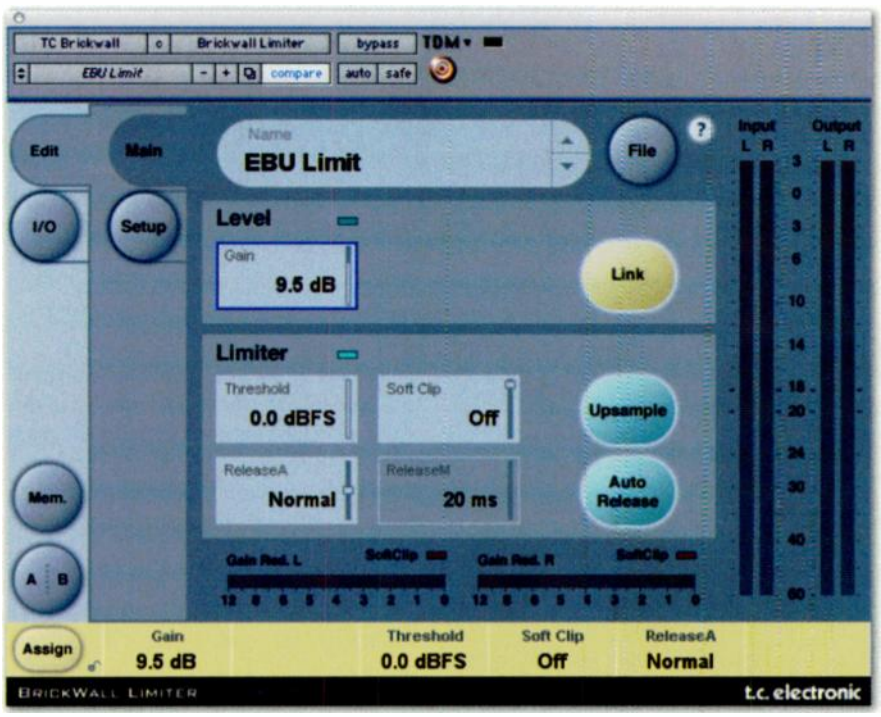
Unlike conventional dynamics processors that work around threshold, ratio and gain make-up controls, *Multi Dynamics* still has threshold and ratio controls but the gain make-up is replaced by two separate controls. Simply put, the Lo Gain control determines the amount of gain applied to the signal when it is below the threshold, and the Hi Gain control determines the amount of gain applied above the threshold. The steepness of the transition from Lo Gain to Hi Gain, as the signal passes through the threshold, is set by the ratio control. So unlike a conventional compressor that continues to apply more gain the further past the threshold the signal goes, *Multi Dynamics* limits the maximum amount of gain with the Hi Gain control. Similarly, a conventional expander continues to apply more attenuation the lower the signal goes below the threshold, but in *Multi Dynamics* the maximum attenuation is set by the Lo Gain control. To configure *Multi Dynamics* as a compressor, therefore, you set the Lo Gain higher than the Hi Gain, while if it is an expander you need, you set the Lo Gain below the Hi Gain.

Loudness Improvements: Single-band Limiters

Loudness Leq	Drum & bass	Rock opera	Jazz quartet	Classical	Vocal ballad
L1	6.1	7.9	7.0	4.3	4.6
L2	7.4	9.6	7.6	4.9	6.9
L3 Ultramaximiser	8.5	9.4	8.5	6.6	7.0
Maxim	6.2	9.4	6.2	3.7	5.5
Oxford Limiter	6.3	8.1	8.0	5.8	6.7
Ozone Loudness Maximiser	6.7	9.9	10	5.7	7.8
Brickwall Limiter	6.8	8.4	8.3	6.3	6.7
Final Plug	6.5	8.3	9.1	6.4	5.2
L2007	8.3	8.5	8.8	6.5	7.0

Sound Quality: Single-band Limiters

	Drum & bass	Rock opera	Jazz quartet	Classical	Vocal ballad
L1	6 (out of 10)	7	5	6	7
L2	8	7.5	6	6.5	7.5
L3 Ultramaximiser	9	8	7.5	8	9
Maxim	6	7.5	7	8	8
Oxford Limiter	8	8.5	8	9	8.5
Ozone Loudness Maximiser	8	8.5	9	8.5	8.5
Brickwall Limiter	8	8.5	8.5	8	8
Final Plug	8	8	7	8.5	7.5
L2007	8.5	8.5	8	9	8.5



The Tests

Just to recap from May's article, my tests used five clips covering a wide range of different programme material: a drum & bass track, a rock opera with solo vocal and choir, a jazz quartet piece, a solo vocal ballad and a classical orchestra piece. First I measured the loudness (Leq A-weighted) of the unmastered version of each track to get a reference. Then I used each limiter to increase the loudness as far as I could without experiencing unacceptable side-effects. Finally, I measured the loudness of the processed version and came up with a loudness improvement score for each track processed by each limiter. The loudness increase is a definite measurement, and although I acknowledge that the point where unacceptable side-effects start is subjective, the fact I did all the tests during the same session, then reviewed the results and redid any that I wasn't happy with, should give as fair a test as I could achieve. See the test results table for more information. On this take 2 shootout I used the best two in each category to get a sensible benchmark to compare sound quality.

Single-band Limiters

- **Drum & bass**

The three new limiters were all quite similar to each other, but *L2007* edges it for me, maintaining some space whilst producing very high gain improvements.

- **Rock opera**

Again, all three were very similar in sound, although *Final Plug* appeared to produce some strange intermodulation-type effects which only became obvious while I was doing the final comparisons between each version of this track. However, they all managed to retain the vocal cut I commented on in the original article.

- **Jazz quartet**

The problem with the piano under the vibes distorting showed up again with all three contenders. *Final Plug* suffered the most with TC's *Brickwall Limiter* performing the best.

▪ **Classical**

All three handled this remarkably well. *Brickwall Limiter* produced a less smooth, rounded sound and yet again *L2007* handled itself very well, especially with the Mode switch on Mellow.

▪ **Vocal ballad**

This track proved just as hard for the three new contenders as for the products in the first round-up. I had to reduce the amount of limiting significantly on *Final Plug* to lose the problem of hearing it work too hard. *Brickwall Limiter* did remarkably well, but the star of the show here is *L2007*. With the Mode switch on Smooth, it produced an excellent sound with a very good loudness improvement.

Loudness Improvements: Multi-band Limiters

Loudness Leq	Drum & bass	Rock opera	Jazz quartet	Classical	Vocal ballad
Ozone Multi-band Dynamics	1.1*	4.3	4.5	2.0	3.4
T-Racks Multi-band Master Limiter	8.4	9.8	7.3	5.2	5.7
L3 Multimaximiser	8.6	9.5	8.7	6.7	7.4
MD3	9.5	10.3	10.1	6.7	9.5
Multi Dynamics & Final Plug	8.1	8.3	9.8	6.5	7.6

*Note that *Ozone*'s Multi-band Dynamics module is not primarily intended for loudness maximisation.

Sound Quality: Multi-band Limiters

	Drum & bass	Rock opera	Jazz quartet	Classical	Vocal ballad
Ozone Multi-band Dynamics	5 (out of 10)	6	6.5	6	6.5
T-Racks Multi-band Master Limiter	9.5	9	8.5	8	9
L3 Multimaximiser	9	9.5	9.5	8.5	9.5
MD3	10	10	10	9.5	10
Multi Dynamics & Final Plug	8	8.5	8	8	8

Conclusions: Single-band Limiters

▪ **Wave Arts *Final Plug***

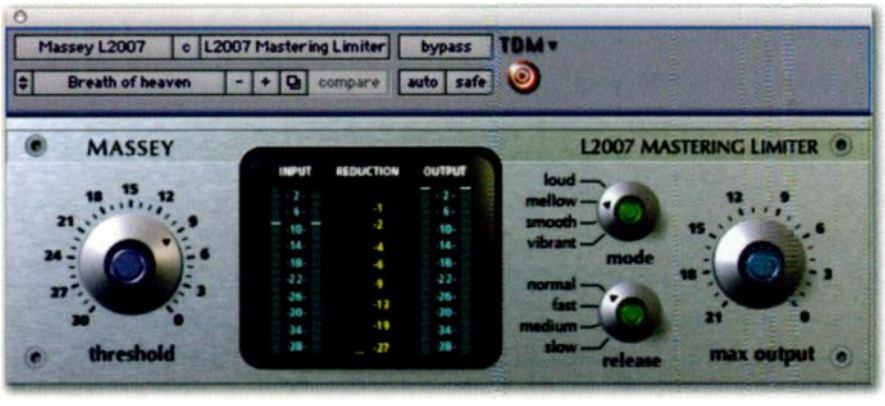
The interface is very intuitive, and I like the way the meters work too. You can see exactly what is going on and the sound from it is good too. It is a shame that the noise-shaping profile name is truncated; you need to click on it to get the pull-down menu list to see the name of the profile you are using. The manual gives a very useful guide as to what each profile does, comparing where relevant to equivalent Waves *L1* and *L2* profiles. This is the only plug-in of the three which has an Audiosuite version, but I did come across a bug in regard to this: when copying the settings from the RTAS version to the Audiosuite one, the Dither profiles didn't copy across. All the other settings came across fine, but the Dither profiles didn't get transferred, whether I copied and pasted the settings or Saved them in one and opened them in the other.

▪ **TC Electronic *Brickwall Limiter***

Perhaps because it's the only one that doesn't look like a Waves plug-in, I found *Brickwall Limiter*'s interface somewhat confusing! I would have been less confused if the Threshold control had been called Output and the Gain control Threshold, and it took me a little while to work out what they did in practice. The bit-depth management and dither are hidden away on another page in the plug-in window, and there are no noise-shaping options, which surprises me for a plug-in in this price bracket.

▪ **Massey *L2007***

This plug-in has got to be the star of the show; when you consider its price, it is truly amazing. The Mode switch is simple and effective, though I would have preferred to have been able to go back one position instead of having to keep clicking to get it back round again. I also took a little while to find the best way of moving the mouse to get the rotary controls to respond accurately — as I have said before, I struggle with rotary controls when using a mouse. That said, let's hope that Stephen Massey continues producing plug-ins of this quality at an unbeatable price and that he doesn't get swallowed up by above manufacturer who then immediately puts the prices up!



Conclusions: Multi-band Limiters

▪ **TC Electronic *MD3***

As you will see from the results tables, there is no contests when it comes to multi-band limiters: TC Electronic's *MD3* walked away with it every time. At first I didn't get on with the multi-page format of this plug-in, but very soon I found I was very easily switching between the different pages to make the various adjustments, though I do still feel they could more attempts to make this plug-in's interface more graphical in style.

Once you start to work with *MD3* you find that it is perfectly designed for mastering. I especially liked the M&S feature, and was able to use it to improve the stereo presentation of all of the test tracks. I also

attention. to detail.

proaudio.com

a division of DIGITAL VILLAGE

DV Pro Audio has a long and proven record in supplying Digidesign Pro Tools systems at every level. From simple portable setups through to full ICON™ HD systems, depend on us for expert advice, reliable service and keen pricing.

Why buy from anyone else?

PRO TOOLS 7
audio [r]evolution

Birmingham
Tel. 0121 687 4777
birmingham@dvproaudio.com

North London
Tel. 020 8440 3440
northlondon@dvproaudio.com

East London
Tel. 020 8510 1500
eastlondon@dvproaudio.com

South London
Tel. 020 8407 8444
southlondon@dvproaudio.com

West London
Tel. 020 8992 5592
westlondon@dvproaudio.com

DV Installation
Tel. 01708 771906
installation@digitalvillage.co.uk

Educational Sales
Tel. 01708 771900
sales@mveducation.com

Southampton
Tel. 023 8023 3444
southampton@dvproaudio.com

Cambridge
Tel. 01223 316091
cambridge@dvproaudio.com

Bristol
Tel. 0117 946 7700
bristol@dvproaudio.com

really appreciated the summary section of the window across the bottom, which changes to show settings for the features you select, with individual bands and global controls all available at your fingertips. One of the other carefully thought-out features is the provision of indicators for soft clipping from the Normaliser section displayed in the Compressor section, so you are immediately aware that the preceding section is clipping without having to even select it to check. In general I found it very easy to get a good sound with this plug-in. TC Electronic have got the balance of the number of controls available to tweak just right and the pedigree from the System 6000 comes shining through in this plug-in.

• **Wave Arts *Multi Dynamics***

Once you appreciate what this plug-in is designed to do and the function of the Lo and Hi Gain controls, its interface makes much more sense. Wave Arts have managed to get a lot of information into the one window, and designed an intuitive graphical interface. I like the 'roller' controls that

control a parameter across all bands, and the way the cursor changes to an up/down indicator when you adjust a control to show you which way to move the mouse. It is features like this that, for me, make such a big difference in the ease of use of any plug-in or application.

In the first shootout, we had the Izotope *Ozone* multi-dynamics section, which turned out to be much more of a dynamic EQ plug-in rather than a limiter *per se*, and the same should be said for Wave Arts *Multi Dynamics* plug-in. It is very clear in the manual that it is designed to perform as a dynamic EQ processor. This doesn't rule it out it as a mastering tool — far from it, it is very useful at handling all sorts of problems that neither EQ nor dynamics plug-ins will handle by themselves. As *MD3* has a limiter in its final stage, it seemed a fairer comparison to put Wave Arts *Final Plug* after *Multi Dynamics* and set them up together, so *Multi Dynamics*' scores in the multi-band section are for both plug-ins together.

Even with the two plug-ins in tandem, though, I still found it tricky to get a really good sound in the time available. The low-level content came across as gritty, which is why it got marked down on sound quality. It may well be possible with more time and experience with this plug-in to get a better result, but at the end of the day I didn't get on with it particularly well as a mastering tool.

What Is Dither?

Pro Tools HD systems process all audio internally at 24-bit, and *Pro Tools LE* processes audio at 32-bit floating. Assuming we are mastering to CD, we need to have the end result as a 16-bit file, but simply truncating 24-bit audio to 16 bits produces what is called quantisation noise. On its own, this doesn't sound too pleasant, albeit at low level. The solution to this problem might seem strange: we add small and controlled amounts of noise, in a process called dithering. Amazingly, adding a small amount of noise actually increases the low-level detail in a digital audio signal.

When adding dither into your final mix, you need to make sure you only do it once, and it has to be the last stage in the mastering process. Consequently, you need to know whether your host application is already adding dither, and if not, you need to add a plug-in in the final stage to do it for you. Remember that the inserts in the Pro Tools master fader channel are after the master fader itself so that last insert is the last point in the mixer. The standard Pro Tools mixers do not add dither, so you must use a plug-in to do this for you, although Digidesign do have versions of their HD mixers with dither built in — one for stereo and one for surround — and you can swap to these if you wish.

The term 'noise shaping' refers to the fact that dither algorithms can be designed to concentrate the noise energy in areas where the ear is less sensitive, thus making it less perceptible, although it will still work its magic by allowing the low level detail to come through. For example, if your final output file is at 96kHz, you can use a dither shape which concentrates the noise above 20kHz where we can't hear it.

Overall Comments

In the single-band section the overall winner has to be the Massey *L2007* mastering limiter. It performed very well on both loudness improvement and sound quality, and at the price, it is just amazing. In the multi-band section, TC's *MD3* is top of the list, not just among these three products but of all the multi-band limiters I've tried. The sound quality was always just that bit smoother and rounder than the competition. It isn't the cheapest plug-in, but if you are serious about mastering it is well worth the money. SOS

Contact Details

Wave Arts

£ *Final Plug* and *Multi Dynamics* $199.95 each.
T Wave Arts +1 781 646 3794.
F +1 781 646 7190.
W www.wavearts.com

TC Electronic

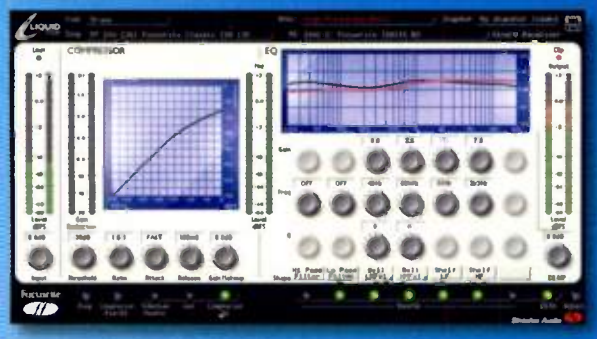
£ *MD3 Stereo Mastering* (includes *MD3* and *Brickwall Limiter*) £1116.25 including VAT.
T TC Electronic +44 (0)800 917 6510.
F +44 (0)800 917 6510.
E infouk@tcelectronic.com
W www.tcelectronic.com

Massey Labs

£ *L2007* $89.
W www.masseyplugins.com

LIQUID MIX

32 channels of classic EQ and Compression simultaneously... on-board DSP means you get it all.



Only Focusrite's Dynamic Convolution gives you genuine emulations of the most revered signal processors used on every hit throughout recording history.

40 Compressors and 20 EQ's at your fingertips.

32 channels running within your recording application,* powered by Liquid Mix's on-board DSP, means your DAW becomes a powerhouse at mixdown.

The extensive Liquid Assets library means virtually any classic processor you've ever dreamed of is yours to use. You can even build your own hybrid EQ band by band!

Want to learn more? Check out the video at **www.focusrite.com/liquidmix**

You'll wonder how you ever lived without it.

*VST/AU/RTAS compliant. (RTAS requires FXpansion wrapper)

audio heritage | sonic integrity | professional solutions

t: +44(0)1494 462246 e: sales@focusrite.com

Focusrite
www.focusrite.com

apple notes

It's been an interesting four years for Mac-based audio engineers and musicians since Apple acquired Emagic, so this month we take the opportunity to acknowledge the anniversary and consider what's changed over this time and what's still to come.

Mark Wherry

Four years ago this month, on July 1st 2002, Apple announced that they had purchased Emagic, which was a pretty surprising announcement for most in the audio and music worlds. We covered the story in detail in September 2002's *SOS* (www.soundonsound.com/sos/Sep02/articles/emagic.asp) and concluded that "with Apple behind them, Emagic are potentially better placed than ever to deliver products for musicians which are optimised for a specific platform. Only time will tell, however, whether this potential will be realised." In the four years that have passed since we wrote that comment, has this potential been realised? How have the other companies fared that Apple acquired during the same period? And do their fortunes have any implications for Mac-based musicians?

Apple's *Shake* Down

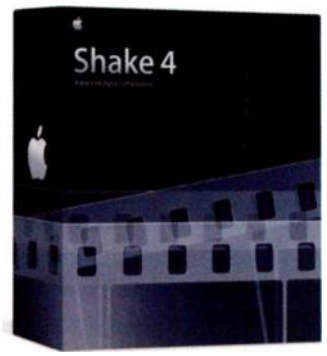
Emagic wasn't the only company acquired in 2002; earlier in the year Apple had also bought three other companies involved in video compositing and other video-related effects: Nothing Real, Silicon Grail and Prismo Graphics. Nothing Real gave Apple *Shake*, an acclaimed compositing application used in high-profile motion pictures such as *Lord Of The Rings*. It's interesting to note that *Shake* was only available for Windows, Linux and IRIX (for Silicon Graphics workstations) at the time of Apple's acquisition. A Mac OS X version came a year later, but the Windows version was discontinued (as with Emagic's *Logic*), along with the IRIX version.

Apple maintained the Linux version because, while the company might stand a chance of persuading some Windows *Logic* users to buy a Mac to run their former sequencer of choice, film studios tend to run *Shake* on hundreds of systems and would be unlikely to replace this level of investment with Macs. To capitalise on this situation, when Apple added the ability for *Shake* to distribute rendering tasks across multiple computers on a network, Mac users got the ability to run the 'node' software on as many Macs as they wanted for free, while Linux users had to pay for a per-node license to do the same. This was an obvious example of Apple's strategy of using the 'Pro Applications' (of which *Shake* was one) to drive sales of Mac hardware.

Silicon Grail was a notable purchase because the company was a competitor to Nothing Real in compositing software, but also offered effects and other layering technology. Prismo Graphics also owned some video effects technology, but their main product was one for creating motion graphics, such as titles. With the release of *Final Cut Pro 4*, in 2003, Apple bundled three new applications, including *Live Type*, a video-titling tool that probably used some of the technology acquired from Prismo Graphics, and *Soundtrack*, an audio application for adding loop-based soundtracks to video productions. (The latter was later made available as a product that could be purchased separately.) *Final Cut Pro 4* also bundled a package called *Cinema Tools* (the result of a previous acquisition) that Apple had originally sold at $999 as a companion to *Final Cut Pro*.

In 2005, however, Apple took bundling to a whole new level, putting together *Final Cut Pro*, *DVD Studio Pro*, *Motion* (a motion graphics application introduced a year earlier) and a new version of *Soundtrack* (now known as *Soundtrack Pro*), to create the *Final Cut Studio* bundle. This sold for the bargain price of $1299, which was significantly cheaper than buying the products separately. Apple would later stop selling the individual products.

In the last couple of weeks, Apple have made a potentially alarming move with the release of



Apple acquired Nothing Real and their *Shake* compositing software in 2002. Four years later, it looks as though *Shake* 4.1 will be the last release of the software.

the Universal Binary version of *Shake* 4.1. What's alarming is that, following Apple's decision to reduce the price of the Mac OS X version of *Shake* from $2999 to $499, rumours started to circulate that 4.1 would be the last version ever released. (Note that the price for the Linux version is still $4999.)

Although the reason Apple might have for killing one of its flagship professional applications is still a little unclear, a next-generation compositing application is apparently in development. However, one rumour is that this won't be introduced until 2008, so Apple are leaving a fairly big gap between *Shake* and whatever comes next. It's worth remembering that, in addition to selling the *Shake* software since 2002, Apple also continued to sell maintenance support for *Shake*, unlike any other professional application in the product line, which is one reason why it was worthwhile to keep all the Linux users happy until now. On the surface, it seems as though the company want to draw a line in the sand and start afresh with a completely new system.

In order to reduce the impact of *Shake*'s demise for high-end users, such as film production studios, there's apparently a source-code license available for a not-insignificant price tag. *Shake* also has a fairly flexible SDK (Software Developer Kit) to allow third parties to extend the functionality of the application. So it seems that while Apple wanted to end their responsibilities with *Shake*, they've been careful to make sure that users are left in a good position until the 'next big thing' arrives.

The *Logic*al Outcome

In our original article looking at what was going to be in store for the Apple and Emagic relationship, we based our predictions for Emagic's future on what Apple had done with their video-orientated acquisitions. *Final Cut Pro* and *DVD Studio Pro* led to *iMovie* and *iDVD* in the consumer space, so it seemed reasonable to expect an 'iLogic'-type application, which of course was later introduced as *Garage Band*.

Since its release two and a half years ago, *Garage Band* has made a huge impact on the consumer MI (musical instrument) industry, both in terms of the explosion of low-cost MIDI and audio accessories from companies such as M-Audio (who have now produced a *Garage Band* clone called *Session*, to try to achieve similar growth in the Windows world), and in terms of the content that has been produced or repackaged for *Garage Band* in Apple Loops format.

Final Cut Pro is also available in a cut-down form (both in terms of features and price) as *Final Cut Express*, so the simplification of the *Logic* product line from *Micrologic AV*, *Silver*, *Gold* and *Platinum* versions to just *Logic Pro* and *Logic Express* was also quite predictable. However, what ▶

audioagency

Home | News | Products | Dealers | Reviews | Purchase | FAQ/Links | Company Info | Contact

Direct Sound

EXTREME ISOLATION Headphones

PLANE quiet

MoreMe Studio Phones

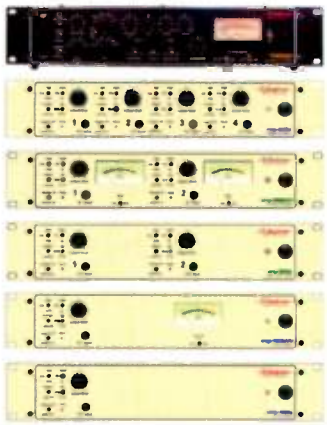
Peluso

Awesome valve microphones hand made in the USA and inspired by and styled on the classic German Neumann-U47 and Telefunken-251 models of the 50's and 60's

~Sebatron~

Innovative Australian Audio

Unique hand made organic valve and class A microphone pre-amps and channel strips.



AcoustiLock

HEAR YOURSELF THINK

Aeron Studio Chairs

AxeTrak

The AxeTrak® is a valuable production tool allowing the engineer or player to quietly capture the unique characteristics of their individual guitar sound. AxeTrak is capable of producing natural sounding rhythms and screaming lead guitar tones when feed through the output of your guitar amp and plugged into a mixing board, soundcard, or recording console.

The ultimate recording device for rock guitarists. Guitar and bass versions available.

FMR Audio

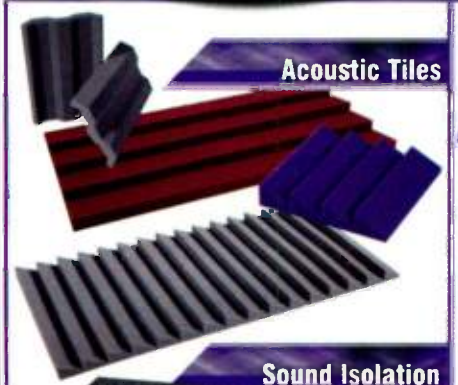
Home of the Really Nice Compressor

Auralex acoustics

Rarely has one manufacturer produced such a powerful range of acoustic solutions capable of transforming an untreated room into a professional mastering recording studio. For over 20 years Auralex engineers have pioneered the art of sonic audio performance delivering the highest level of acoustic control at realistic and affordable prices. Whatever your budget, discover how Auralex can redefine your sound and truly fulfil your audio expectations. Auralex Acoustics – Total Sound Control for the real world. Available through a network of UK Pro-dealers.

TOTAL SOUND CONTROL

Acoustic Tiles



MAX-Wall Systems

Bass Traps

Sound Isolation

Roominator Kits

Sound Construction

www.audioagencyeurope.com

Tel: 01908 510123

Email: info@audioagencyeurope.com

wasn't quite so predictable was Apple discontinuing Emagic's line of plug-in instruments and effects and bundling all of them with *Logic Pro 6*, and some of them with *Logic Express 6*. (It probably wasn't worth Apple's while to manage so many small products and the associated dongle authorisations for copy protection.) In view of *Logic Pro*'s $999 price tag, this was a huge saving, and users upgrading from previous versions could get all the plug-ins for a couple of hundred dollars. Great — except if you'd bought all the *Logic* plug-ins separately before this happened!

Of course, this bundling is a direct parallel to what happened with both *Final Cut Pro* and *Final Cut Studio*, and while Apple obviously make money selling software, they make a great deal more money by selling new Macs to *Logic* users who are looking for more power now that they have more plug-ins to run! This action reminded us again that Apple are a hardware company, and it certainly puts pressure on third-party developers to follow suit; except how can they? If Steinberg bundled all of their optional plug-ins with *Cubase*, the company would lose a chunk of their income. Steinberg are a software company, after all.

Would MOTU have included a new set of instruments as a major selling point for the recent *Digital Performer 5* upgrade if Apple hadn't thrown down the gauntlet? And would Digidesign have bundled *Xpand!* (the first instrument created by the company's new Advanced Instrument Research division) if no other music-making application bundled significant instruments? Whatever the reason, the trend of increasing the value of these products (without increasing the price) has definitely created a win-win situation for the users — assuming that the software-only companies can stay in business, or justify creating Mac-compatible versions.

With the introduction of the first major new version of *Logic* under Apple's ownership, *Logic 7*, the Emagic web page disappeared and Emagic's hardware products were quietly discontinued. Fortunately for Mac users, though, while the Windows drivers have languished, Apple continue to update the drivers for Emagic's old hardware devices under OS X, and it's a great relief to still be able to use the Unitor 8 and AMT8 with Intel-based Macs.

Logic Pro 7 also introduced the concept of *Logic Node*, where you could network a group of Macs together and offload certain processing from your main *Logic* system to the other Macs on the network. This concept seemed similar to Apple's video applications, where compression or rendering tasks can be offloaded to other Macs on a network, and you can almost imagine the directive given to *Logic*'s programmers to implement a similar technology for audio that would encourage the sales of more Mac hardware.

Making Plans For *Logic*

In terms of Mac technology, a great deal has happened in the last four years: the introduction of the G5 in 2003 and the availability of Macs with Intel processors at the start of this year, to name just two milestones. In both of these cases, *Logic* has, of course, been in a position to be the first application of its kind to take advantage of such new technologies, and for *Logic* users this is obviously a great thing. So Emagic was indeed "better placed than ever to deliver products for musicians which are optimised for a specific platform."

However, there's one area of improvement where *Logic* still hasn't quite caught up: full support for the multiple cores available in the high-end Power Mac Quad. Although there are some benefits to running *Logic* on a Power Mac Quad, *Logic*'s audio engine only runs on two of the four cores (which has nothing to do with the fact that *Logic*'s multiprocessor handling is derived from the OS 9 days, as some have suggested). This means that it can't handle quite as much DSP processing as the Mac version of *Cubase*, for example, which launches an audio engine on all four cores.

From what I understand, the lack of Quad support so far is a conscious choice that was made so that more resources could be given to optimising *Logic* for Intel-based Macs, as we've discussed in previous columns; but assuming there will be a dual-core, dual-processor, Intel-based Power Mac replacement this year, hopefully *Logic* will be poised to take advantage of this extra power. But does the decision to optimise first for the new portable and consumer Intel-based products mean that Apple doesn't regard *Logic* users requiring high-end Mac hardware as a priority? And if so, is this necessarily incorrect? How many musicians are running Power Mac Quads? We'd need a survey to know for sure.

That Apple would 'retire' *Shake* raises at least a concern for *Logic* users, but there's an important difference between the two apps, in that there are more unique users of *Logic* than *Shake*: in the case of the latter, there are single users who buy a large number of licenses. However, that Apple would consider creating a next generation of compositing software does raise the question of whether there will ever be a post-*Logic* audio and music application from Apple. Or will they simply continue to improve *Logic*, as they did for version 7? Perhaps *Logic Pro 8* will go some way towards answering this question.

In the meantime, I think *Logic* is safe with Apple. Even if *Garage Band* represents the future of mass-market music-making, once a musician outgrows it they need somewhere else to go — and the only way Apple can guarantee that they stay with the Mac is by making sure the *Garage Band* user buys *Logic*. There might be other Mac music software today, but Apple controls the destiny of *Logic* to ensure there is always a high-end music and audio package on the Mac.

This theory actually plays out, even at the moment, when you consider that every update to *Logic* includes features to make it compatible with songs created in the latest version of *Garage Band*. And some of these features have actually benefited the high-end user, such as Apple Loops, in a rare example of the low end driving the high end, as opposed to the other way around.

Outside the professional applications, there's been another important development since Apple acquired Emagic: the iTunes Music Store, which has fuelled the growth of Apple's consumer music products. To this end, Apple can now control almost every step of a music and audio workflow, from creation to distribution, and for the most part has continued to endear itself to professional musicians and consumers alike. At the end of the day, along with *Garage Band*, these might have been the most valuable assets Emagic contributed to Apple. SOS

Logic Pro 7 was the first major new release of the application since Emagic's acquisition by Applw.

TRILOGY
TOTAL BASS MODULE

INSPIRATION
POWER
SIMPLICITY

Trilogy
The pro's first choice solution for every type of Bass sound
- Acoustic,Electric and Synth.

WWW.SPECTRASONICS.NET

SPECTRASONICS
VIRTUAL·INSTRUMENTS

Exclusively distributed by Time+Space Distribution Ltd.
www.timespace.com sales@timespace.com +44 (0)1837 55200

TIME+SPACE

pc notes

Buss-powering USB and Firewire peripherals from a laptop seems like a good idea, cutting down on annoying power supplies and keeping everything portable — but it's not quite as straightforward as you might imagine.

Martin Walker

Many musicians buy Firewire audio interfaces with a view to buss-powering them from their PC. This is certainly convenient, saves plugging in yet another wall-wart, and ensures that the interface powers up and down safely with your PC. However, while the 6-pin Firewire ports found on desktop PCs do offer buss power (as do the 6-pin ports on Firewire-to-PCI and PCI Express adaptor cards), unfortunately the 4-pin Firewire ports found on the vast majority of PC laptops don't provide it (one of the few that does is Samsung's X50, which offers a single 6-pin powered port).

This is indeed frustrating, but is fairly easily explained. Typically, 6-pin Firewire ports in desktop PCs supply 12V DC at up to 15W per port, but looking at my laptop's PSU, it's rated at 65W, so supplying an extra 15W to a Firewire peripheral simply isn't practical — and your laptop battery wouldn't hold out very long when powering such a device, either.

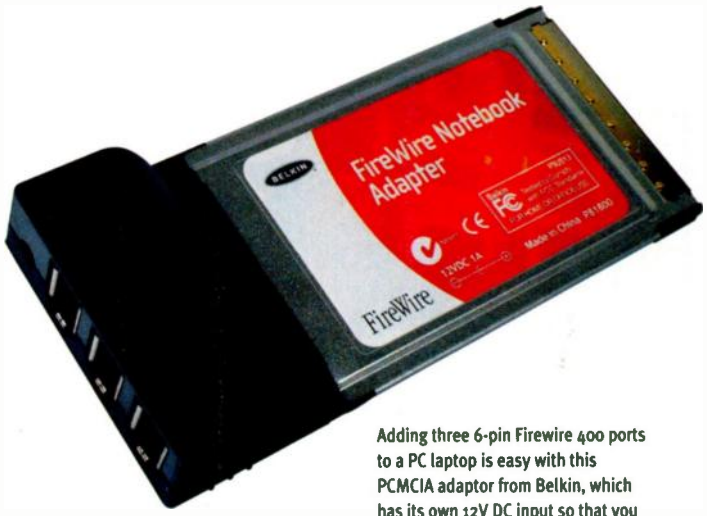
Despite the above, many PC laptop owners point to the 6-pin buss-powered Firewire ports on Apple iBooks and Powerbooks as evidence that buss powering from a laptop can work. However, according to Apple, these Firewire ports only provide up to 7W of peak power when the computer system is on or the power adapter is connected. There seems to be a lot of variation in both voltage and power ratings between buss-powered Firewire ports on both Macs and PCs, so you should be careful not to assume that you can plug any Firewire peripheral into any available port. As long as it requires 7W or less, you should be able to power it from a powered Mac or PC Firewire port, but above this you might suffer reliability problems or find that the peripheral won't boot properly.

Even with the more typical 15W capacity of most PC Firewire ports, many musicians don't realise that you can still overload them. This is a particular problem if you daisy-chain peripherals. Remember that although the Firewire specification theoretically allows up to 63 devices to be daisy-chained from one port, they are still subject to the maximum power drain of that port — and a *single* 3.5-inch external hard drive can typically consume 12W or more. In fact, so many devices are now being powered from Firewire ports that some PCI-to-Firewire adaptor cards, such as Miglia's Alchemy FW800 PCI, even provide optional connectors that you can plug into standard hard-drive power cables, so you can reduce the card's drain on the PCI buss.

However, in general, if you intend to daisy-chain several Firewire peripherals and buss-power them, do check the total wattage required (there should be a power-consumption rating somewhere on their cases), or use the supplied AC adaptors to reduce power drain on your computer.

PCMCIA-to-Firewire Adaptors

In the quest to run buss-powered peripherals, the next port of call for some PC laptop owners is often to consider buying a 6-pin PCMCIA-to-Firewire adaptor, but this route won't work either, as unfortunately the Cardbus specification doesn't allow such ports to be buss-powered. Fortunately, there is a workaround: some PCMCIA adaptors have a separate connection to another source of power, to enable their Firewire ports to offer buss power.



Adding three 6-pin Firewire 400 ports to a PC laptop is easy with this PCMCIA adaptor from Belkin, which has its own 12V DC input so that you can either plug in a mains wall-wart PSU or use a battery to provide buss power for Firewire peripherals.

One such option is Belkin's F5U513ea Firewire Notebook Adaptor. Widely available for about £30, this unit provides three Firewire ports and a 12V DC input that you can connect either to a suitable wall-wart or to a 12V battery. The latter option is an ideal solution for any laptop musician doing mobile recording: without the additional drain of powering an audio interface, your laptop's battery is likely to last another hour or more.

A slightly different approach that will appeal to those who don't want to use wall-warts is the USB 2.0/Firewire Card from Sweex (www.sweex.com/product.asp?pid=452&s=1), which offers two USB 2.0 ports, plus one 6-pin and one 4-pin IEEE1494a port. The clever twist is that it's bundled with a short mini-jack to PS/2 pass-through connector. The mini-jack plugs into the PCMCIA card, and the other end into your laptop's PS/2 port to provide buss power for the card's 6-pin Firewire port, while the pass-through connector still lets you plug in a PS/2 keyboard or mouse. If you're desperate to avoid using a wall-wart, this might provide the perfect solution, although do bear in mind the total load on your laptop's PSU, especially if you're also attempting to plug in USB 2.0 devices that require their own power. If you're considering battery power for a mobile recording session, don't expect your batteries to last very long, either, especially if your audio interface features high-quality mic preamps.

Cubase Compatibility

In *SOS* April 2005, I wrote a PC Musician feature entitled 'Easier Alternatives To Flagship Music Apps', in which I extolled the virtues of entry-level sequencers and maintained that not everybody needs the very expensive flagship versions of major applications, such as *Cubase SX*. After all, while the latest version of *Cubase SX* retails at £499, the *SL* version is only about £250, while the entry-level *SE* typically sells at £99, yet still ▶

Tiny Tips: Keep It Simple

Recently, a PC user reported that after upgrading from *Cubase SX2* to *SX3*, his audio stopped every time he moved his mouse and started again immediately he stopped moving it. The probable cause turned out to be a mouse-pointer utility called *Cursor XP*, from Stardock (www.stardock.com), that lets you create visually attractive mouse cursors. Like Stardock's other products, such as *Window Blinds* and *Window FX*, *Cursor XP* looks wonderful, but as soon as it was switched off the audio problems disappeared. Once again, this reinforces my advice that although it's nice to have a glamorous-looking PC, a stripped-down version is nearly always much more suitable for music production.

ELECTROMUSIC

82 Copley Road Doncaster DN1 2QW

Computer Music

KORG ESX/EMX.....£489
MACKIE SPIKE £145
ALESIS MULTIMIX FW...£225
ESi WAVERACK 1010...£239
ESi JULI@ 24/96 CARD...£79

DIGI 002....£1529
DIGI 002R.£799
DIGI Mbox 2..£329

Alesis SR16.....£79
MOTU 896HD....£769

RME Fireface 800....£899

SAMSON C RANGE FROM £85

CUBASE SE 3 .....£99
LOGIC EXPRESS..£179
ABLETON LIVE.....£239

Recording Studio Gear

TLAUDIO 5051.....£439
TLAUDIO 5050.....£299

MACKIE ONYX 800R.....£839

Mics and Monitors

Genelec 8020.....£399
Genelec 8030.....£590
Genelec 8040.....£950
Genelec 8050.....£1750
Samson Rubicon R5a.....£229
Samson Rubicon R6a.....£329
CAD GLX3000 Studio Pack.£179
CAD GLX2400 Studio Pack.£120
Shure KSM27.....£260
Audio Technica, Beyer, Rode, Call

Keyboards

Yamaha P90.....£649
Yamaha P70.....£489
Korg Triton TR.....£825
Korg SP200.....£489
Korg Microkorg.....£250
Roland Keyboards..... Call
Casio Keyboards.....Call..

Digital Recorders

Tascam DP01FXCD.....£395
Yamaha AW1600.....£739
Korg D3200.....£829
Tascam 2488.....£750
Roland Digital Recorders.....Call

CHECK OUT NEW WEBSITE

Sound Cards

M-AUDIO AUDIOPHILE 2496.....£65
EMU 0404 SOUNDCARD.....£65
M-AUDIO FIREWIRE AUDIOPHILE..£169
M-AUDIO FIREWIRE 1814.....£349
TERRATEC PHASE 88 RACK.....£299
EMU 1212M SOUNDCARD.....£129

Tel: 01302 369999 Fax:01302 738523

MAIL ORDER AND PERSONAL CALLERS WELCOME. 30,000 SQ FEET OF SHOWROOM

www.electromusic.co.uk

▶ has a huge feature set that is more than sufficient for many musicians.

Steinberg make it easy for musicians to upgrade from *SE* to *SL* and *SX* if they subsequently need the extra features, so I was hoping this article might encourage some musicians to buy entry-level software rather then resort to cracked flagship versions. Indeed, we've subsequently had posts on the *SOS* Forums from musicians about to buy *Cubase SE* and *SL* and wanting to check whether they would still be able to collaborate with friends who already had *SX*, and from others who have started projects on *Cubase SX* at college and wanted to continue their work at home by buying their own legitimate copy of *Cubase SE*.

Steinberg detail the compatibility and the interchangeability of project files in their Knowledgebase (http://knowledgebase.steinberg.de/110_1.html), but unfortunately, while *Cubase SX3* and *SL 3* will read each other's project files, you can't open these files in *Cubase SE 3*. Or, at least, not without a tweak discovered by David Dorn, an enterprising *SE3* user trying to help a bunch of students. He examined the difference between the project files, which amounts to a single letter: one contains the words 'Cubase SX' and the other 'Cubase SE' near the end of each file, and if you change the 'X' to 'E' you can load the *SX* files into Cubase *SE3*.

Of course, *SE3* will ignore any *SX3*-specific content it finds in a project file, and you have to be careful not to exceed the *SE* limits for plug-in sends, inserts and VST Instruments, otherwise *SE* might crash (which is probably why Steinberg made these tweaks in the first place). However, everything should otherwise work fine, enabling lots more musicians to officially buy into the *Cubase* range.

David has since written a stand-alone *SLEX* utility, to turn any *SX/SL* project file into an 'SE-friendly' version. The utility is now available for free download from http://cothelsoft.wiffles.net, and it ought not to infringe any copyrights, since it simply modifies your own project files — after all, this seems no different from the multitude of word-processing applications that offer import and export functions for documents created in Microsoft's *Word*, Corel's *WordPerfect*, and similar programs.

The freeware *SLEX* utility will let *Cubase SE* owners load *Cubase SX/SL* project files into their sequencer, so that they can legitimately collaborate on projects with *SX/SL* owners.

SLEX comes with no guarantees, of course, and if your PC crashes as a result of using it, that's nothing at all to do with Steinberg. However, I personally feel that the utility ought to be not only tolerated but accepted by Steinberg, since it should ensure plenty more legitimate owners of *Cubase SE3*, who may later on upgrade to the *SL* or *SX* versions. In the meantime, *SLEX* lets them work within the limitations of their entry-level version of *Cubase*, safe in the knowledge that they can transfer their work to *Cubase SX*, and back again, if and when they need to. SOS

PC Snippets

Free VST Sampler: Lots of musicians seem to have downloaded the freeware VST Instrument version of *Synthfont* (www.synthfont.com) since I mentioned it in PC Notes December 2005. For those who don't necessarily want to use Soundfont sample banks, another more general-purpose VST Instrument sampler to try is Kotkas Soft's *Paax 2 Free*, which not only imports Soundfonts but also Akai S5000/6000 programs. Written by Mexican developer J Andrés Alvarez, this freeware version is limited to 64-voice polyphony and only plays 8/16-bit WAV files, but *Paax 2 Pro* has 128-voice polyphony and also supports 24/32-bit samples — and it only costs $29. Both versions have three envelopes and three LFOs per preset, 'micro-granular' pitch-shifting, and built-in reverb, chorus and delay effects. *Paax Pro 3*, which should also be available by the time you read this, adds a multi-mode resonant filter, advanced time-stretching and pitch-shifting, and direct-from-disk streaming. It costs just $45.
W www.kotkasuniverse.com/paax_index.htm

Keep An Eye On Vista: In PC Notes April 2006, I made some cautious recommendations for musicians who wanted to make sure their new PCs would be suitable for running music software on Microsoft's forthcoming Vista operating system — when it eventually gets released in early 2007. Well, you can now keep abreast of all the latest developments on the official web page, which will not only keep you revved up with tantalising previews but also has a list of detailed hardware requirements with comprehensive footnotes, and even a downloadable beta-version Upgrade Advisor that will tell you what hardware components (if any) you'll need to upgrade to Vista.
W www.microsoft.com/windowsvista

Centrance Universal Driver: Centrance have released their Universal Driver for Firewire audio interfaces, after extensive beta-testing. As I mentioned in *SOS* February 2006, in my feature on using more than one audio interface at the same time, this driver supports multiple interfaces from different manufacturers, and even multiple hosts, so you can combine the I/O from several Firewire interfaces and use them as one across several applications. The Universal Driver supports up to 32 I/O channels on a Firewire 400 port, at sample rates of between 32kHz and 192kHz (24-bit), handles both ASIO and WDM driver formats and costs $79. Apparently it's written at a lower level than many other Firewire drivers and Centrance are confident that it will enable many users to achieve lower latencies than they do with their existing drivers. I haven't yet been able to test it myself, as your audio interface must be on the compatible hardware list and mine isn't. As I write, the current list comprises the Apogee Rosetta; Behringer FCA202; M-Audio Solo; Mackie D2, Onyx 400f and Satellite; Miglia Harmony Audio; Presonus Firepod; and Tapco Link Firewire 4x6. Extra interfaces are likely to be added shortly.
W www.centrance.com

Do you want to trigger samples from a MIDI keyboard, but can't afford a commercial software sampler? Try the freeware *Paax 2 Free*.

Eddie Moors Music

The Music Corporation

Unbeatable Deals on all Samplers, Synths, Modules, Outboard & Effects

679 Christchurch Road, Bournemouth BH7 6AE

We will not be beaten on price

0845 330 6656 www.eddiemoorsmusic.com

Tascam
2488 Exclusive

24 Track Digital Studio with on board effects and CDR/RW

RRP £999 Ex Demo From **£599**

Akai
MPC2500

Latest state of the art sampling workstation

RRP £949 Ex Demo From **£799**

Tascam
US2400

24 Channel USB control surface with motorised faders

RRP £1199 Now From **£599**

Korg
D3200

32 Track Digital Studio with on board effects and CDR/RW

RRP £999 Ex Demo From **£789**

Mackie
Control Universal

Limited number now available from Stock

RRP £999 Now From **£779**

Yamaha
AW2400

24 track Track Digital Studio with on board effects and CDR/RW

RRP £1999 Now From **£1179**

Focusrite
Liquid Mix

Dynamic convolution processor

RRP £499 **£Call**

Boss
GS10

The ultimate guitar effects system with onboard speakers and USB audio interface. Includes Editing software & Cakewalk recording software.

RRP £449 Now Only **£169**

Korg
Ampworks

- 11 classic amp models
- 10 speaker cab models
- 11 digital effects
- Available for guitar and bass

RRP £99 Now Only **£39**

Roland
Fantom S

Powerfull professional synthesizer with onboard sampling, sequencing and digital effects

RRP £1499 Used **£599**

Roland
Fantom X6

Best selling professional synthesizer with on board sequencer and powerful sampling.

RRP £1599 Ex Demo From **£1149**

Roland
V Synth XT

Flagship multi oscillator synth module with colour touch screen. Includes synth & vocal modelling plus external audio processing

RRP £1499 Ex Demo From **£999**

Roland
Fantom X8

Top of the range Fanton synthesizer 88 weighted keys plus onboard sequencing and sampling.

RRP £2299 Ex Demo **£1499**

Roland
Varios

How much would you have to pay today for a Jupiter 8 synth and a TB303 bass machine!! The varios includes the software for both these classics plus VP9000 sampling technology

RRP £999 Ex Demo From **£199**

Roland
V-Synth

Top of the range multi oscillator synthesizer

RRP £1199 Ex demo from **£1299**

Roland
VS2000CD

Top range expandable 20 track digital studio with on board effects and CDR/RW

VGA board and monitor for full visual editing available (not inc)

RRP £1599 Ex Demo From **£899**

Rode
Microphones

Unbeatable Rode Mic Deals

NT1A • NT2A • NTK • NT2000 • Classic II

Best UK Deals **£Call**

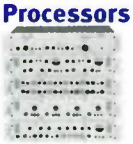
Roland
MV8000

64 track Production studio with sampling and CD burning

Mouse/monitor and analog out boards worth £550 included

RRP £2349 Ex Demo From **£899**

TL AUDIO
Processors



Best possible prices on valve classic and Ivory 2 series

Best UK Deals **£Call**

Mackie
Monitor Deals

Ex Demo HR824 From £799

Ex Demo HR624 From £549

Best Uk Mackie Deals **£Call**

Roland
V Drums

• TD20 • TD12 • TD6 • TD3

New/ used and ex demo in stock

Best UK Deals **£Call**

Genelec
Monitors

Best possible deals on Genelec monitors and subs

Best UK Deals **£Call**

Alesis
Andromeda A6

16 voice real analogue synthesizer

RRP £2299 From **£1299**

Joe Meek
JM47

Legendary Industry standardlarge diaphragm studio condenser microphone

RRP £349 Now Only **£89**

Helicon
Harmonizers

- Voiceworks
- Voiceworks live
- Voice Pro
- Voice Doubler

Call us for an unbeatable deal on any Helicon or TC equipment

Best UK Deals **£Call**

Novation
Synthesizers

• X-Station • Remote LE • Xiosynth

Lowest UK prices on all synths and controllers

Best UK Prices **£Call**

Korg
Triton

New and used Triton synthesizers from £599.Call

Best Uk Triton Deals **£Call**

Logic
Pro 7

• Pro 7 • Express 7 • Upgrades

Call now for a great deal

Best UK Deals **£Call**

Dynaudio
Monitors

• BM5A • BM6A • BM15A

Call for a great deal

Best UK Deals **£Call**

Roland
VG88

Roland Virtual guitar system including GK3 pickup

Limited stock

RRP £899 From **£399**

New/ Used Ex-Demo

Keyboards/Synths

Hammond XK1 Drawbar Organ	£879
Hammond XK3 Drawbar Organ	£1699
Roland VR760 Drawbar Organ	£999
Roland Fantom Synthesizer	£499
Roland Fantom S88 Synthesizer	£899
Roland Fantom X8 synth sampler	£1679
Roland Fantom XR synth/sampler	£739
Roland D50 Synth	£249
Roland Juno D synthesizer	£289
Roland D550 Synth Module	£299
Roland JD990 Synth Module	£249
Roland RD700 Stage Piano	£799
Roland RD300SX Stage Piano	£699
Roland SRX Sound Boards	From £99
Roland SR-JV80 Sound Boards	From £39
Korg Triton Extreme 88	£1899
Korg Triton LE 61 synthesizer	£599
Korg Triton EX 61 synthesiwzer	£1299
Korg MS2000 Synthesizer	£399
Korg Micro Korg Synthesizer	£239
Korg PA1X Pro keyboard	£1499
Korg BX3 dual manual organ	£599
M Audio Keystation Pro 88	£299
Yamaha P200 Stage Piano	£699

MultiTrack

Alesis HD 24 24 track HDR	£899
Alesis BRC ADAT Remote	£199
Zoom MRS1608CD 16 Track studio	£469
Korg D1600 16 Track Studio	£499
Tascam 2488 24 Track Studio	£549
Tascam 122 MkII Cassette Recorder	£129
Boss BR1600CD 16track studio	£679

Mixers

Behringer MX9000 mixer	£749
Behringer DDX3216 digital mixer	£399
Mackie 32/8/2/ 8 buss	£1899
Mackie CFX12 mikxer with effects	£299
Mackie DFX12 mixer	£149
Roland VMC7100 Controller	£199
Gost 24 Channel Meterbridge	£199
Ghost 32 Channel Meterbridge	£149

Outboard/Effects

TC M300	£109
TC M One XL	£289
TC Fireworx	£829
TC Gold Channell	£1299
TC M3000	£1169
Universal Audio 1176	£1199
Electrix EQ Killer	£79
Focusrite Trackmaster Pro	£169
Focusrite Voice master pro	£299
Focusrite Liquid Channel	£Call
TC D Two Studio Delay	£299
TC Voiceworks harmonizer	£399
TC Voice Pro Harmonizer	£Call
TLA VP1 mono valve processor	£1379
DBX DDP Dynamics Processor	£149
DBX 1066 Comp/Lim/Gate	£199
Behringer Tube Ultra Q	£69
TLA Indigo 2051Preamp	£299
Boss GT Pro rack effects	£399
Boss GT8 guitar floor effects	£279

Monitors

Mackie HR624 active monitors	£579
Mackie HR824 active monitors	£799
Tapcoe S5 active monitors	£199
Tapcoe S8 active monitors	£299
Event ASP6 active monitors	£399
Event TR5 monitors	£249
Genelec series 8000 pro monitors	£call
Samson Resolve 50a active mons	£149
Samson Resolve 65A active mons	£185
Samson Resolve 80A active mons	£219
Samson Rubicon R6A active mons	£289
Samson Rubicon R5A active mons	£219
Mackie SRM450 ative PA cabs pr	£849

Drum/Groove

Roland TD12 V Drum Kit	£1899
Roland TD3 V Drum Kit	£499
Roland SP404 Sampler	£219
Roland SP606	349
Akai MPC1000	£399
Akai MPC4000	£1349
Roland TR808 Drum Machine	£599
Joe Mox X Base 09	£399

www.eddiemoorsmusic.com

for sale

classifieds

Classified Dept., Sound On Sound, Media House, Trafalgar Way, Bar Hill, Cambridge CB3 8SQ

Patrick Shelley
Telephone: +44 (0)1954 789888
Fax: +44 (0)1954 789895
ISDN: +44 (0)1954 781023
email: patrick@soundonsound.com

display box
£35.00 + VAT
per single column centimetre
Discounts offered for series.

situations vacant
£40.00 + VAT
per single column centimetre.

lineage
£1.20 per word including VAT
(minimum of 15 words).

(3 column page). Rates include full colour + typesetting. Box Numbers available at an additional £50.00 per insert.

Courses	D/F-L	For Sale	A/B	Rentals	D	Studio for Let	C
Demos & Masters	C	Insurance	E	Services	B	Studio for Sale	C/E
Duplication	M-R	Rec/Prog Room	C	Situation Vacant	S	Tuition/Programming	S/T
Flight Cases	B	Recording Studios	R	Studio Acoustics	C	Wireless	T
For Hire	G	Rehearsal Rooms	S	Studio Furniture	C		

services

for sale

flight cases

demos & masters

rec/prog room

studio furniture

studio acoustics

studio to let

studio furniture

studio acoustics

studio to let

courses

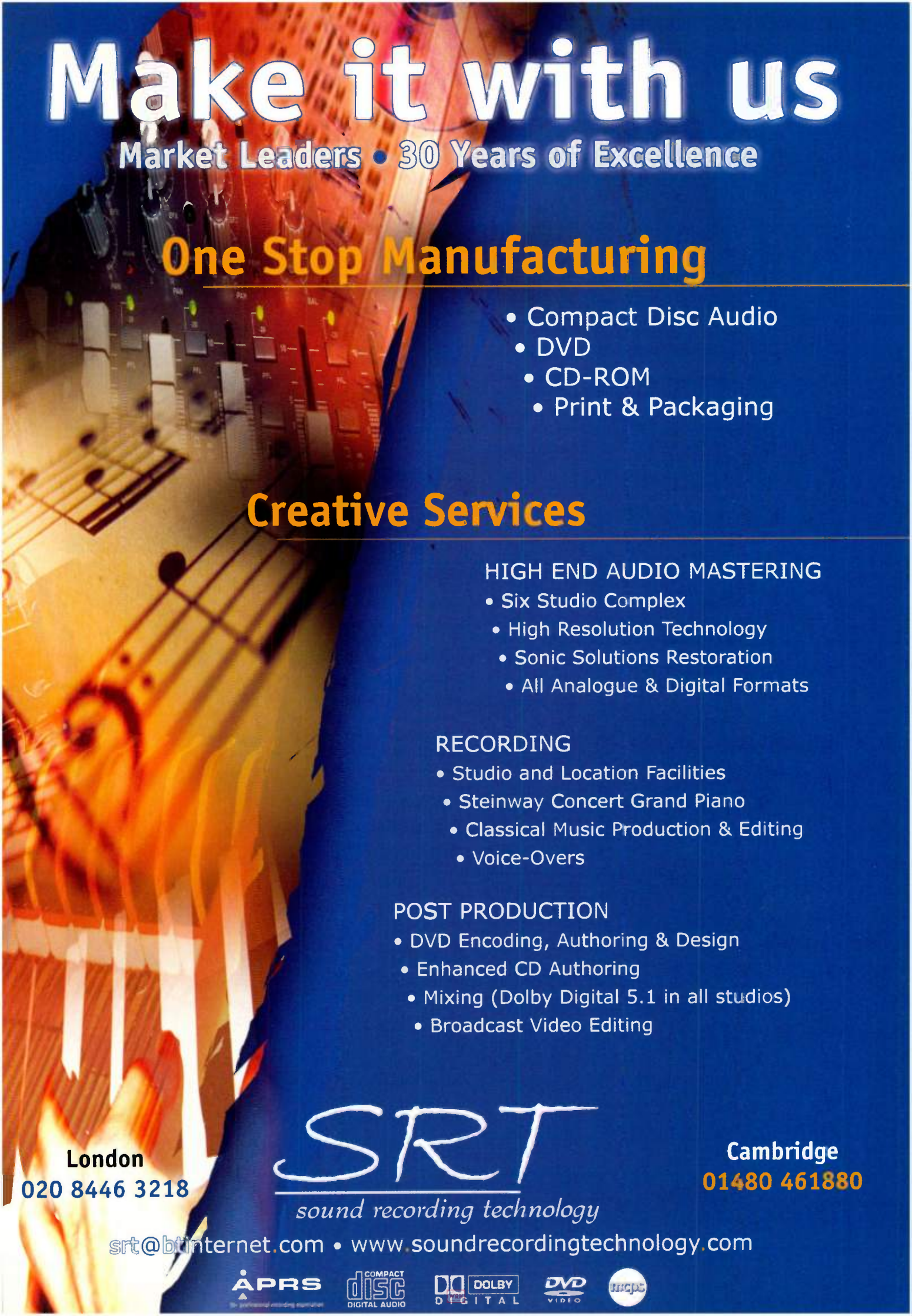
for hire

This could be your classroom. For more information about your future career visit www.sae.edu

Professional audio education
State of the art equipment
Established since 1976
A university degree
Global recognition

SAE INSTITUTE

LONDON 0207 609 2653 - LIVERPOOL 0151 255 1313 - GLASGOW 0141 429 1551
NEW YORK - PARIS - SYDNEY - SINGAPORE - BERLIN - DUBAI - PARIS - AMSTERDAM
plus a further 40 additional locations around the world. Degrees issued by Middlesex University.



Make it with us

Market Leaders • 30 Years of Excellence

One Stop Manufacturing

- Compact Disc Audio
- DVD
- CD-ROM
- Print & Packaging

Creative Services

HIGH END AUDIO MASTERING

- Six Studio Complex
- High Resolution Technology
- Sonic Solutions Restoration
- All Analogue & Digital Formats

RECORDING

- Studio and Location Facilities
- Steinway Concert Grand Piano
- Classical Music Production & Editing
- Voice-Overs

POST PRODUCTION

- DVD Encoding, Authoring & Design
- Enhanced CD Authoring
- Mixing (Dolby Digital 5.1 in all studios)
- Broadcast Video Editing

London
020 8446 3218

SRT
sound recording technology

Cambridge
01480 461880

srt@btinternet.com • www.soundrecordingtechnology.com

APRS · COMPACT disc DIGITAL AUDIO · DOLBY DIGITAL · DVD VIDEO · mcps

tuition/programming

Helping you create

Tuition, Advice and Troubleshooting

Cubase, Logic, DP, Reason and more. All aspects of System and Studio set-up

Experienced Engineer and Tutor

Steve 07986-522595 or 07962-185711 www.swsounds.co.uk email:swsounds@tiscali.co.uk

London and Sussex, Kent, Surrey borders

PC problem solving

cubasetuition.com

joe@cubasetuition.com

Tel 07816 649047

LogicPro/Express(TM)

Garageband(TM)

Consultancy and training on a one-to-one basis

Dov Waterman Tel: 07766768314

Certified Pro

Email: info@forksound.com

web: www.forksound.com

REAL-TIME Solutions

www.real-timesolutions.co.uk

Recording and Production
Pro Tools HD, Control24.
Restoration, Mixdown/Mastering.

Custom Built PC Systems
Systems designed for Audio.
Upgrade/Repair service available.

All services operate at competitive rates.

Studio Installation/Configuration
System setup, Optimisation, Problem solving.
Experience on all platforms.

Technical Assistance/Support
Live callout 24/7, Consultancy, 1 to 1 Tuition.

Tel: 0208 6914578
07901 598504
email: info@real-timesolutions.co.uk

Certified Pro

Logic Pro

Training, Courses & Installation

NINE YARDS Training

Apple Mac Specialist

Dave Gale

Lecturer in Music Technology
Guildhall School of Music & Drama
Approved by Sound Technology PLC/Emagic UK

Tel : 01252 726463 (24Hrs)

e-mail : dave.gale@nineyardstraining.co.uk
internet : www.nineyardstraining.co.uk

Logic (audio)

One-to-One Tuition

Macintosh Hardware & Software Specialist

Peter Dudley

Mobile 07813 657524

email: logictraining@mac.com
web: www.logictraining.co.uk

rehearsal rooms

SSC SOUNDSTAGE

Superb Rehearsal Suite

Live Space
Control Room
VIP Reception Lounge
Kitchen Mini Bar
Parking

For those who deserve the best

synthesiser service centre

www.synthservice.com

020 8961 7890

situations vacant

REGIONAL MANAGER

Music Production Products

London and the South East

The Professional Music Division is pleased to announce that due to increased sales success and an expanding product portfolio we are looking to recruit a further sales specialist for London and the South East.

The successful candidate must have an in-depth knowledge of Music Production products (Yamaha AW-series workstations, Motif synthesisers, Yamaha PA products etc). Applicants will need a strong sales background, be able to communicate in an authoritative, knowledgeable manner and possess good people and administrative skills.

Key tasks:

- Business development of an account based area.
- Work with and utilise given resources in order to achieve set sales targets and long term goals.
- Give considered feedback to the UK based Sales & Marketing department.

We offer all the usual benefits of working for a high profile international manufacturer, including a generous salary and company car.

Please e-mail or send your CV for the attention of:
Elizabeth Beattie
Yamaha-Kemble Music (UK) Ltd, Sherbourne Drive,
Tilbrook, Milton Keynes MK7 8BL
Tel: 01908 369207

elizabeth.beattie@gmx.yamaha.com

YAMAHA

CAREERS AT turnkey

SENIOR POSITIONS AVAILABLE

Turnkey is growing rapidly and we usually find that the limiting factor to our growth is the number of ambitious, talented and driven individuals we have on board.

We are therefore seeking to recruit career-focused individuals, with great business and people skills, in a variety of roles across the company. Previous management experience is preferable but not essential if you have already worked in the industry - so if you want to work for the market leader in a fast-paced environment with exciting future prospects, send us your CV with a covering letter to the Sales Director at the address below.

Competitive salary packages according to experience and position.

SALES ASSISTANTS

Are you fanatical about Music Technology?

Whether you're just out of college, a seasoned professional or an experienced sales person, if you're heavily into the products we handle, we would like to hear from you. Turnkey is the UK's largest and fastest growing outlet for recording and computer music equipment. Manufacturers from around the world have praised the store and this is continually developed with input from all of our staff.

The role requires a good knowledge base of the equipment that we sell. Previous sales experience and / or customer service experience is desirable. The role requires individuals who are able to engage our customers in their buying process. Once in the role, training is given to our sales staff to ensure that their knowledge is as up to date as our customers would expect.

As the country's leading outlet, salary packages are competitive and our most successful sales assistants are amongst the highest paid people in the industry. The company's continual development provides as excellent opportunity for staff to pursue a career within the company.

Turnkey is an equal opportunity employer.

[illegible] applications (which must include [illegible] reference) to: Human Resources Manager, Turnkey, 114-116 Charing Cross Road, London WC2H 0JR or Fax: 020 7692 7614 or apply online at www.turnkey.co.uk/tkweb/employmentform.jsp

tuition/programming

wireless

The Advertisers Index

Subscribe Now

Only £48

for 12 issues

and save over £11 on the newsstand price!

also includes a FREE electronic subscription

☑ **I would like to subscribe to *Sound On Sound* for the price of £48 for 12 issues**

Name: _____ Card No: _____

Address: _____ Card Valid Date: _____ Card Expiry Date: _____

_____ Card Signature Strip (Last 3 Digits): _____ Issue No: _____

_____ Signature: _____ Date: _____

Postcode: _____

Country: _____

Your Email: _____

Telephone: _____

This amazing offer also includes a free electronic subscription and saves you over £11 on the newsstand price.

Sound On Sound, Media House, Trafalgar Way, Bar Hill, Cambridge CB3 8SQ, UK. Tel: 01954 789888 Fax: 01954 789895 Web: www.soundonsound.com

Tel: 01954 789888 **or subscribe online at: www.soundonsound.com/subs**

Grab a bargain!

Browse over £1 Million of second-user gear on-line!

www.soundonsound.com/readersads

UNBELIEVABLE

clearance deals on New, Ex-Demo & Used stock!

If you've ever wanted to own one of these items, these prices represent a once in a lifetime opportunity!

		RRP	SALE
computer hardware			
USB 2 pen Drive 128MB	NEW	£50	£6.99
Belkin Ieee1394 F wire Pcicard	NEW	£0	£9.99
Belkin 4 Port USB Hub		£60	£17.99
RME HDSP9632X	NEW	£50	£19.99
Lindy Firewire 6 Port Hub	NEW	£90	£34.99
DIMM S3200 1G Ram	NEW	£115	£39.99
MOTU Fastlane USB		£65	£35.99
DIMM 1GB DDR400 Kingston	NEW	£175	£69.99
Edirol UA20		£160	£72.99
Line 6 Guitar Port USB		£100	£79.99
Seagate SCSI 72G	NEW	£940	£99.99
Tascam US224		£265	£119.99
Presonus Inspire	NEW	£160	£129.99
Storcase 5.25" U160 FRM & Can		£235	£139.99
M-Audio Quattro		£249	£159.99
Frontier Dakota		£499	£179.99
Mindprint Trio	NEW	£330	£199.99
Neovo M15V Flatscreen Monitor		£436	£199.99
Steinberg System 4		£400	£199.99
M-Audio Firewire 410		£350	£204.99
Mindprint Trio	NEW	£330	£249.99
Focusrite Saffire V stabon and B stabon		£330	£269.99
Tascam FW1804		£480	£289.99
Format Gallery 1740 Adc Flatsc		£630	£329.99
RME Multiface I O Box		£483	£369.99
RME Multiface Mk2 36 Chn I/O		£540	£399.99
Aardvark Q10		£790	£429.99
Universal Audio UAD1 Studio Pack	NEW	£800	£599.99
Digidesign Command 8	NEW	£952	£629.99
Digidesign Command 8		£952	£669.99
TC Powercore PCI Mkll	NEW	£939	£699.99
Steinberg Nuendo Timebase		£1 750	£999.99
Digidesign Pro Tools 192 I/O		£2 938	£2,029.99
Mackie DXB200		£10,500	£7,499.99
DJ			
Numark DM1050		£79	£39.99
Kam GM25		£75	£45.99
Numark DM1002XE		£120	£46.99
Numark TT1650		£180	£94.99
Numark Matrix 2		£150	£99.99
Pioneer DJS	NEW	£139	£99.99
Numark CDN22mk4 Dual Player		£230	£129.99
Numark TT500	NEW	£300	£169.99
Stanton SK 2 F		£380	£179.99
Numark TTX1	NEW	£450	£249.99
Numark CDMIX2		£500	£289.99
Technics SL1210 Mk2		£499	£289.99
Numark CDMIX2	NEW	£500	£299.99
Stanton C504 2 CD MP3 Player	NEW	£800	£499.99
Allen & heath Xone 92		£1,175	£849.99
headphones			
Beyer DT252 Single Ear	NEW	£100	£49.99
Sennheiser HD280 Silver	NEW	£110	£69.99
keyboards			
Korg 8MB Rom For PA80	NEW	£115	£39.99
Behringer KX1200	NEW	£274	£129.99
Korg Microkontrol		£275	£149.99
Fatar TMK88 88 Note Keyboard		£199	£159.99
Casio CTK731		£300	£169.99
Fatar SL960 XP		£470	£269.99
Novation Remote 49	NEW	£400	£269.99
Novation Remote 61	NEW	£450	£329.99
Korg PA50 Arranger		£700	£419.99
Oberheim MC3000		£850	£499.99
Roland Discover 5		£1,299	£499.99
Waldorf Micro Q	NEW	£900	£499.99
Yamaha P140S	NEW	£925	£649.99
Korg Triton TR 76 Key	NEW	£950	£749.99
Yamaha MO6		£950	£749.99
Big Briar Voyager Signature		£2,999	£2,399.99
lighting			
NJD 4 Channel Controller	NEW	£60	£9.99
NJD 4 Channel Par Can Fader Pack	NEW	£126	£19.99
700w Smoke Machine	NEW	£45	£29.99
recorders			
Tascam Porta 02 Mk2	NEW	£145	£74.99
Tascam CDGT1		£150	£99.99
Marantz SD4050/PRO		£219	£144.99
Tascam DP01 FX		£350	£229.99
Zoom MRS1044 HD		£700	£369.99
Tascam 2488		£980	£589.99
Roland VS2000CD		£1,600	£1,079.99
microphones			
AKG D44S CCS	NEW	£20	£14.99
AKG C416L	NEW	£100	£39.99
Behringer B5	NEW	£78	£49.99
Superlux ORKA3C2 Drum Set	NEW	£196	£59.99
AKG C900		£130	£79.99
Samson Q 5 Bundle	NEW	£170	£99.99
M-Audio Solaris	NEW	£230	£119.99
Sennheiser K6P Output Stage	NEW	£179	£129.99
ADK SLPRO		£190	£139.99
AKG SR40 & MP40 Micropen		£320	£149.99
ADK TL With Shockmount & Flightcase		£230	£169.99
AKG WMS 400HT	NEW	£560	£299.99
Sennheiser EW372 Gtr		£515	£339.99
ADK Matched Pair of TL's	NEW	£550	£399.99
mixing consoles			
Yamaha CD8 Y2	NEW	£302	£49.99
Yamaha Dm1000 V2 Upgrade Kit	NEW	£220	£49.99
Alesis Multimix 8 USB Mixer		£150	£85.99
Mackie DXB Digi Adat/TDIF Card	NEW	£395	£149.99
Numark IDJ Mixer For Ipods	NEW	£230	£149.99
Mackie DXB AES/O Card	NEW	£595	£199.99
Mackie DXB Analogue I/O Card	NEW	£595	£199.99
Mackie DXB Firewire Card	NEW	£645	£199.99
Mackie DXB Mic/Line 4 I/O Card	NEW	£745	£249.99
Tascam IFandMIF DM24 8 CH AD		£400	£299.99
Voice Systems Alpha8	NEW	£600	£299.99
Mackie D8B Mon/Mouse/Kbd		£502	£379.99
Mackie CFX16		£872	£419.99
Soundcraft FX16		£840	£419.99
Mackie D2 Firewire	NEW	£586	£449.99
Mackie DXB Mic/Line 8 I/O Card	NEW	£995	£499.99
Yamaha 02R 96 Meterbridge		£950	£649.99

		RRP	SALE
monitoring			
Tannoy Revolution X	NEW	£150	£49.99
Yamaha Ystm100 Monitors (pair)		£199	£49.99
Alesis RA150		£175	£74.99
M-Audio DX4		£130	£79.99
Yamaha MSP3		£149	£82.99
Alesis Monitor 1 Mk2		£250	£86.99
Presonus Central Station Remote	NEW	£194	£99.99
M-Audio BX5		£200	£129.99
Samson Resolve 65 Active	NEW	£329	£169.99
KRK V6		£290	£219.99
Tapco SW10	NEW	£270	£229.99
Alto PS4HA Active (each)		£410	£230.99
M Audio BX8		£300	£239.99
Fostex NX6A Active Monitor Single		£350	£249.99
Tannoy Precision Active 6D	NEW	£400	£249.99
Tannoy Precision Passive 8	NEW	£500	£269.99
Tannoy Precision Active 8D		£480	£269.99
Tannoy TS12		£500	£289.99
SPL Surround Monitor Control	NEW	£474	£299.99
Tannoy Precision Active 8D	NEW	£480	£299.99
Tannoy Reveal Active 6D		£500	£309.99
SPL MTC 2381	NEW	£500	£369.99
PMC DB1S	NEW	£617	£399.99
Mackie HR824		£714	£409.99
Mackie HR626		£713	£419.99
Genelec 1030AM		£1,321	£849.99
outboard			
Focusrite Aes Cable	NEW	£25	£4.99
Samson S COM 2Ch Comp		£129	£69.99
Studio Electronics Modmax Phaser Pedal	NEW	£330	£99.99
Studio Electronics ModMax Filter		£330	£169.99
RME Quadmic	NEW	£313	£229.99
Lexicon MPX1		£699	£369.99
Mackie Onyx 800R 8ch	NEW	£1,000	£799.99
Drawmer 1969 Mercenary Edition	NEW	£1 498	£999.99
Avalon VT737SP		£1 909	£1,349.99
PA			
Behringer CX3400	NEW	£128	£79.99
Wharfedale Evpa10 Single		£119	£79.99
Voice Systems Contact		£150	£99.99
Behringer B1520 Speaker System		£189	£105.99
Voice Systems Contact XB (each)		£180	£109.99
Wharfedale EVPX15B	NEW	£234	£149.99
Wharfedale EVPX15B		£234	£179.99
Wharfedale EVP15PSB	NEW	£248	£199.99
Voice Systems Base 12		£330	£209.99
Carlsbro Gamma 15.400		£430	£239.99
Peavey PV2600	NEW	£559	£269.99
Voice Systems Factotum Mini		£400	£299.99
EV SXA100	NEW	£840	£349.99
Mackie M2600 Power Amp		£1,222	£399.99
PC and Mac software			
Arturia Storm v1 5		£150	£39.99
East West Adrenaline	NEW	£80	£49.99
East West Afro Latin Storm	NEW	£80	£49.99
East West Vapour Synth	NEW	£80	£49.99
Best Service Cult	NEW	£113	£69.99
Digidesign Pt Hd 7 Upgrade	NEW	£118	£99.99
Native Instruments Guitar Combo	NEW	£130	£99.99
Waves Musicians Bundle	NEW	£200	£99.99
Digidesign Impact (HD Only)	NEW	£476	£99.99
East West Bosendorfer Piano	NEW	£130	£99.99
Syntrillium Cool Edit Pro		£330	£99.99
Wizoo Latigo Latin Soft Synth	NEW	£180	£99.99
Native Battery 2		£150	£109.99
E-MU Emulator X	NEW	£299	£149.99
USB Xtreme FX	NEW	£266	£149.99
M-Audio Protools M-Powered		£240	£164.99
M-Audio Factory Pro Bundle	NEW	£280	£169.99
MOTU Mach Five	NEW	£295	£169.99
TC Fabrik R Reverb Plug In	NEW	£241	£169.99
Yellow Tools Culture	NEW	£280	£169.99
Yellow Tools Majestic	NEW	£270	£169.99
Fruity Loops XXL Edition	NEW	£250	£199.99
Sony Oxford EQ Powercore	NEW	£411	£299.99
TC VSS3	NEW	£405	£299.99
Steinberg Wavelab 6	NEW	£500	£309.99
Waves Gtr Guitar Tool Rack Tdm	NEW	£899	£599.99
sample CD's			
G Media Tape Banks 3	NEW	£30	£19.99
Sonic Drum Series 1 Exs24	NEW	£100	£19.99
XXL Club Edition 3	NEW	£70	£29.99
CD XXL String Machine	NEW	£65	£29.99
E-MU Beat Shop 2 Library	NEW	£35	£29.99
IK Sonik Capsule Groove Capsule	NEW	£70	£29.99
CD Synthasia 3 Loopland		£65	£39.99
IK Sonik Capsule Drums 3-Pack	NEW	£170	£49.99
CD Festa Latina Set Vol1-3	NEW	£130	£69.99
QL Voices Apocalypse	NEW	£310	£149.99
East West Orchestral Platinum XP	NEW	£2 000	£1,499.99
sampling & percussion			
Compact Flash 64MB Card	NEW	£20	£9.99
Simm Jedec 72 Pin 32MB	NEW	£176	£9.99
ION IDP01 Expansion Kit	NEW	£40	£24.99
Compact Flash 1GB Card		£100	£24.99
Linn Rm4 Drum Machine Software	NEW	£100	£49.99
Alesis SR 16		£180	£69.99
Boss DR3	NEW	£149	£99.99
Yamaha DD55		£200	£113.99
Jomox Jazbase 001	NEW	£549	£249.99
Boss SP505		£450	£269.99
Akai MPC4000		£2,300	£1,299.99
sound modules			
Korg FMC PCM 02 PA80 Ethnic	NEW	£175	£49.99
Roland SRJV8008 60's & 70's Keyboard		£200	£129.99
Roland SRX10	NEW	£200	£129.99
Roland SRX02 Concert Piano	NEW	£199	£139.99
Voice Systems Genius 8MB		£480	£159.99
Acces Virus Classic		£900	£599.99
Dave Smith Instruments Evolver Rack		£1,000	£849.99
Clavia Nord lead 3 Rack		£1,385	£859.99
Access Virus Ti Desktop		£1 200	£1,029.99
Big Briar Voyager Rack RME		£1 300	£1,099.99
stands & cases			
Furman PL8E Power Conditioner		£194	£129.99
Quiklok ZJ755 DJ Workstation	NEW	£371	£199.99
Novation Remote 61 VS/BS Bag	NEW	£350	£299.99

Call Now! 020 7419 9999

turnkey

For more incredible deals see our FREE 40 page Buyer's Index & Soho Soundhouse guitar catalogue with this magazine.

*Please check full guarantee details at time of purchase

Errors & omissions excepted

WRH

Are DJs musicians? They certainly need talent...

Harry Webley

Some say that DJs are the superstars of tomorrow — but they said that yesterday... The truth is that DJs have always been superstars — how else is fantastic music played to hordes of partygoers and aficionados alike? Magazines can only go so far in explaining what songs sound like, and everyone perceives music differently, so are they exactly reliable? A good DJ will go that extra distance to bring you the best sounds you've never heard.

Some of my favourite DJs were the ones doing the rave circuit in the early '90s, just because the sounds they were playing felt like another universe crashing into my previous conceptions of music. I remember buying tape packs and absorbing sounds which until that moment I didn't even know existed!

It is a great shame that the majority of radio stations operate with identical playlists — the radio DJ is in a unique position to expose unheard music to a much wider audience. It's testament to this that the late John Peel was heralded as one of the most important men in 20th century music. Unaffected by commercialism, his eclectic broadcasts introduced many new acts into the mainstream. This injection of talent kept an increasingly stale industry on its toes, but now a new breed of DJ is emerging.

With more and more music being made but less and less places to play it, the hip-hop mixtape phenomenon has taken hold, with streets and markets being flooded with homemade CDs. This enables new songs to reach an audience literally hours after completion, and the race is on to find the hottest exclusives. In the US some mixtapes even outsell artist albums!

A truly great hip-hop DJ displays musicality, originality, skill and dedication. If you possess these qualities you have the right to call yourself a turntablist; one who uses the turntable as an instrument.

Let me take a minute to break down each aspect of turntablism, beginning with musicality. Many of the greatest musicians have no musical training, they can just feel when something is right, and this is true of most turntablists. From the seamless blending of two songs to the complex arrangement of super-fast scratch patterns, a gift for music theory is essential.

Some turntablists take things further by creating whole compositions using loop pedals and even pressing up their own records! The juxtaposition of pitch and direction of the record with the formation of 'syllables' on the crossfader can produce alien language, and has influenced many eccentrics to push the boundaries of scratching in a quest to communicate with one another.

There are some DJs whose styles are so distinct that within minutes, you can tell who it is. How can that be? Well, maybe they're chopping sounds together in a trademark style, or maybe the scratching displays a distinctive natural finesse. The technique of freestyle scratching is where learnt patterns are strung together in a flow of conscious movement by the hands, so everyone's style is a little different.

Skill and dedication go hand in hand, and as with any instrument, a serious amount of hours need to be spent honing your talents. Practice, practice, practice really is the key to all the above.

A good DJ will go out of their way to find the killer record you need to hear, but the turntablist can make you listen to familiar records in a new context, for example cutting a kicking Beyoncé vocal over a classic James Brown instrumental — there really is no limit to what you can play. Some turntablists make whole tapes playing '80s pop songs, but do it in such a way that the whole mix feels like an adventure back in time.

One rather unique skill I must mention is a trademark of the top battle DJs. 'Juggling' is the live remixing of tracks using two records. By memorising the position of a groove in relation to the centre label, a turntablist can loop short sections of music, punching-in snares or other sounds, creating new fills and rhythmical patterns. Combine this with the unlimited supply of vinyl records and strange things start to happen. 'Disses' can be directed to other DJs by rearranging rap verses, musical phrases can be replayed in different orders and grooves can even be 'locked' using stickers.

This relentless quest to try and get the most out of bits of vinyl has revolutionised the way we make and sell music. From using obscure samples and noises, to uncovering rare tracks and inventing new marketing schemes — some even say the DJ is the musician of the future! SOS

About The Author

Harry Webley, aka DJ Moschops, is one of Ireland's top turntablists. He is involved in youth workshops around Dublin and performs regularly with rap crew 'Urban Intelligence'.

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.

Fostex PM-Series
Have You Heard The Word?

Precisely calibrated bi-amped power in each speaker, innovative driver construction and Fostex's usual attention to detail mean that the PM-Series deliver a full-on bass response that's meaty and dynamic, and a high end that sparkles with life and vitality.

The new compact and supremely attractive PM0.4 with its high-gloss front, exclusively designed 4" woofer, 0.65" soft dome tweeter and 36 watts of bi-amped power offers an incredibly controlled sound for a speaker designed for ultra-tight spaces.

Breaking a new price / performance barrier, the Fostex PM0.4, at only £129 a pair, is the route into professional monitoring you've been waiting for.

Fostex PM-Series

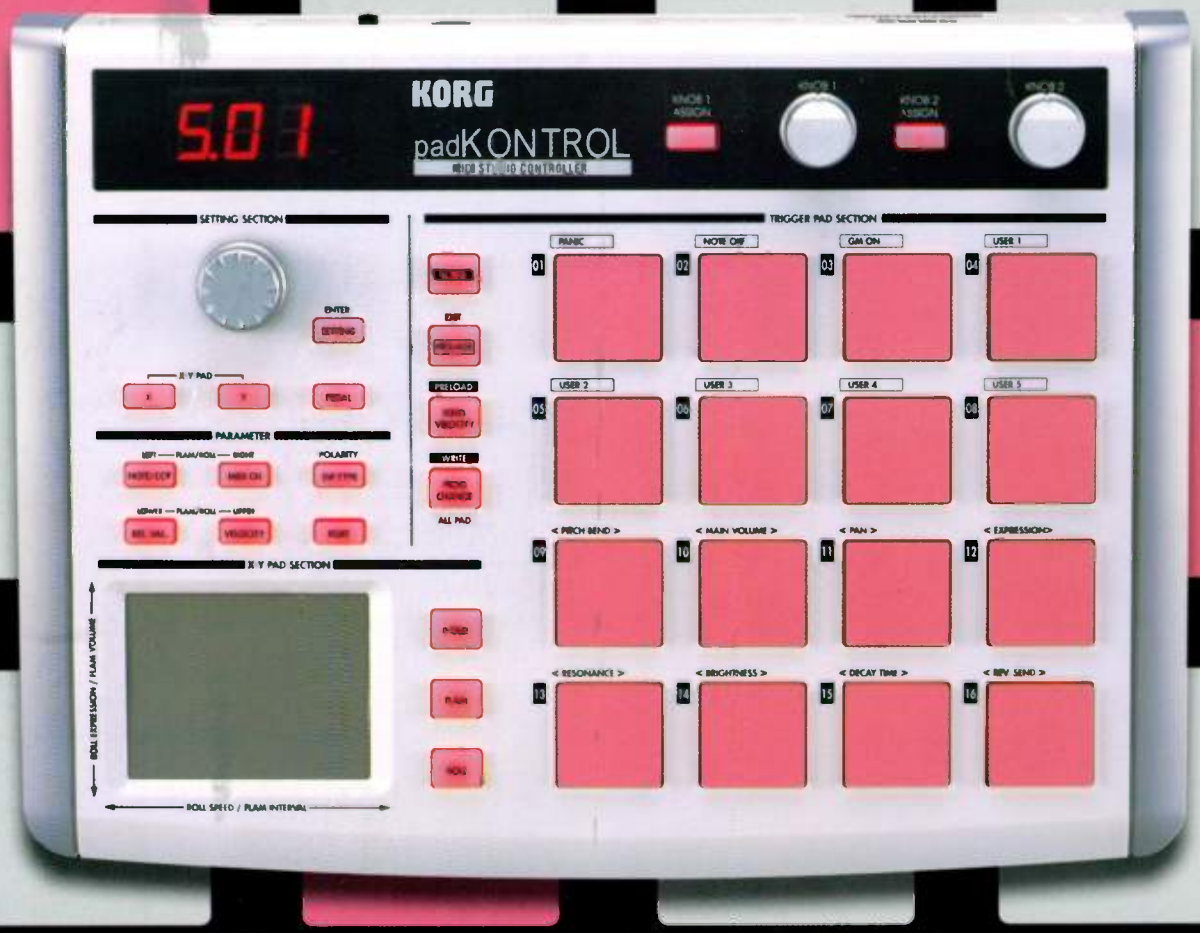
PM-2 PM-1 PM0.5 PM0.4 PM0.5-Sub

Exclusively distributed in the UK by SCV London.

40 Chigwell Lane, Oakwood Hill Industrial Estate, Loughton, Essex, IG10 3NY
Tel: 020 8418 1470 Fax: 020 8418 0624 Email: info@scvlondon.co.uk

bias Hughes & Kettner UA SONY A|A|S muse RECEPTOR Digigram FOSTEX GENELEC hear VOVOX Superlux PRODUCER

Get your hands on padKONTROL



For grooves with maximum expression

The moment you first touch padKONTROL you'll instinctively know that your music production has entered another dimension, taking you beyond the boundaries of keyboard control. With the ultra-responsive padKONTROL MIDI studio controller, you get a unique and highly sensory command over your music that feels instantly natural and will have you creating your hugest, organic-sounding grooves yet. In addition to accessing drum sounds and samples, padKONTROL can control soft synths and effects, trigger loops and even video clips. Combining an assignable X-Y touchpad that produces natural-sounding flams and rolls, 16 great-feeling illuminated trigger pads, assignable knobs, footpedal input plus a massive software bundle - padKONTROL is simply the most expressive and versatile pad controller ever!

padKONTROL. Another touch of genius from KORG.

KORG

www.korg.co.uk

padKONTROL
MIDI STUDIO CONTROLLER