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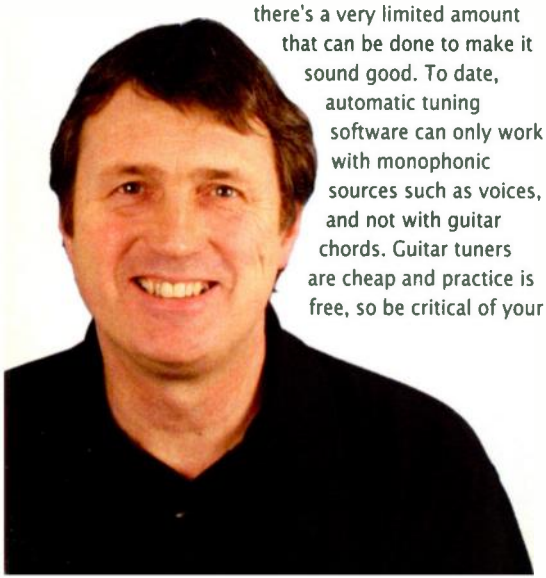
Quality Matters!

We often hear stories from readers who have bought lots of good-quality recording equipment but who are still unhappy with the results they are getting. Sometimes the reasons for this are easily diagnosed, but in many instances, it is because the user has lost sight of the fact that the technology is there only to record music, not to create it. Stick up a microphone and it will record what it hears, and if it is a very good microphone connected to a very good preamplifier, then the chances are that it will record what it hears pretty accurately. The problem is that the performance or the sound of the instrument being recorded may not be up there with the quality of the recording system, in which case no amount of expensive upgrades will solve the problem. For example, switching from a £100 microphone to a £2000 mic when the real problem is worn-out guitar strings or a badly set-up guitar with intonation problems won't help anybody aside from the company that makes the expensive microphone.

During the course of our Mix Rescue series, I've been faced with a number of guitar parts that have been recorded well enough but have had tuning issues. These may have been slight enough to get away with live, but on a recording, particularly where the guitar part is exposed, they stand out like a sore thumb. It can't be stressed too highly that if the tuning and timing of the performance isn't right, there's a very limited amount that can be done to make it sound good. To date, automatic tuning software can only work with monophonic sources such as voices, and not with guitar chords. Guitar tuners are cheap and practice is free, so be critical of your recordings and do them again if there are audible timing or tuning issues.



Another common problem is that rather than get in a bass player, some musicians have a go at playing the part themselves, and while they may hit the right notes, they often don't pick the strings with the required degree of conviction, so the sound becomes messy and unfocused. You can only do so much with EQ and compression — give me a well-played track over a screen full of remedial plug-ins any day.

The other issue of course is that the microphone doesn't just hear the instrument or voice at which it is pointed, but also spill from other instruments, noise from elsewhere in the house, passing traffic and, possibly most serious of all, room reflections. In a small domestic studio, room reflections are seldom constructive and usually serve only to make the sound appear boxy, coloured and difficult to mix. Despite the emphasis on acoustics in our Studio SOS series, it seems that acoustic treatment, both in the control room and in the studio area, receives far less attention than it should. Just by making sure that the room you record in is quiet and that the area around the performer is acoustically absorbent (using foam, sleeping bags, duvets or even proper acoustic baffles), you'll make much better recordings that are a joy to mix, rather than constantly challenging your salvage abilities. It's just like photography in many ways — put something good looking and well lit in front of the camera, press the button and the chances are you'll have a good picture, even if you don't have state-of-the-art equipment. Recording equipment today tends to be very good, even at the budget end of the market, and it all has the ability to capture a musical performance with adequate quality. That performance though, is entirely down to you.

Paul White *Editor In Chief*

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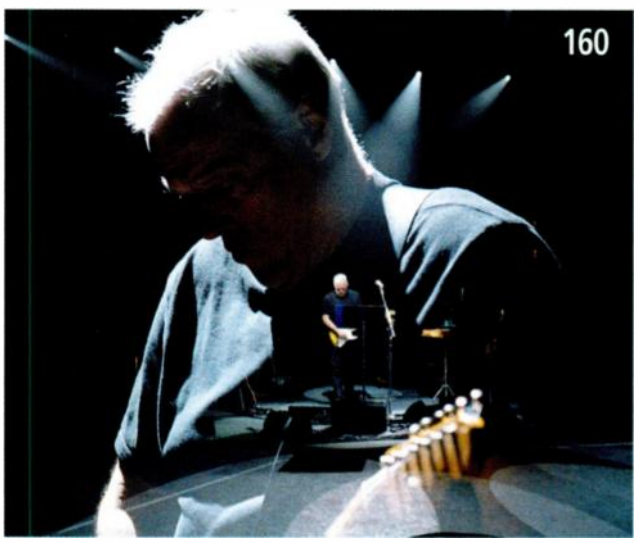

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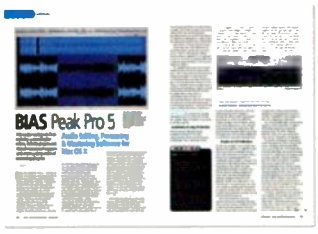

product tests


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Audient completes DAW hat trick with Centro

Following success in the DAW-based market with the ASP008 eight-channel mic preamp and Sumo analogue summing mixer, UK pro audio equipment manufacturers Audient have added a third product to their range of studio-based outboard gear. The Centro is a monitoring controller that features six stereo digital and six stereo analogue inputs, two cue mixes — for flexible foldback monitoring — and built-in talkback. It can route to three pairs of speakers and a subwoofer all from a neat desktop control module. All input and output sockets are housed in a rackmount breakout box, meaning the Centro's desktop unit requires only power and control cables.

The system is expected to be shipping very soon and will cost £995. If you're interested in Audient's other products, check out this month's competition on page 66, where you can win both the ASP008 and Sumo.

Stirling Trading +44 (0)20 8963 4790.
www.stirlingtrading.com www.audient.co.uk

The face of Duende

Following news in *SOS* May 2006 of SSL's Duende Firewire DSP engine, we can now show you what it will look like! The rackmount unit (pictured below) will host 32 channels of audio processing at sample rates up to 96kHz. Its two plug-ins — *Xlogic Stereo Buss Compressor* and *Xlogic Channel Strip* — are based on the SSL C-series console's onboard DSP, featuring stereo compression and full channel EQ, dynamics and filters respectively. Initially, the Duende will only operate on Macs, with VST, Audio Units and RTAS (via an FXpansion wrapper) plug-in support. A PC version is set to follow.

Apart from revealing its front panel, SSL have given us a price and a release date, too. The Duende will cost £999 including VAT and will be distributed by Sound Technology from the end of June. Check out SSL's web site for more information.

Sound Technology +44 (0)1462 480000. www.soundtech.co.uk www.solid-state-logic.com

Summer courses announced at Bath Spa University

This summer, Bath Spa University will host three short courses on *Sibelius 4*, Korg's Triton Extreme workstation synth and Apple *Logic Pro*. They will be run by CENDIA (the Centre for Digital and Interactive Audio), which is based at the University, and will be delivered by Apple-certified instructors. The *Logic Pro* 101 course starts on the 30th July and successful participants will be able to take Apple's on-line exam (at extra cost) upon completion of the course. The Triton Extreme event will commence on the 2nd August and Korg will supply Triton Extreme keyboard workstations, to use during the course, to the first six applicants to apply. Further students are expected to bring their own Tritons. The *Sibelius 4* course starts on the 5th August, covering topics including MIDI recording, transposing and scoring to video. Each student on the *Sibelius* course will have access to an Apple Powermac G5. All courses last three days and will be held at the University's Newton Park campus. Each one costs £360 with accommodation available at extra cost. There's also a £40 discount for under-17s.

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+44 (0)1225 875875.
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So you think you need a record deal?

UK-based download company 7 Digital have launched Indie Store, an on-line music distribution service for unsigned artists and independent labels. The web site is very much like some of the other 'global community' sites in its layout, with multimedia galleries, biographies and message boards. However, Indie Store features the option to sell your music, with up to 80% of the revenue going directly to the artists — depending on which type of account you sign up for. Their starter account is free and has the capacity to upload four audio tracks in MP3, WMA or WAV format. Pro users pay £75 to sign up, but benefit from higher revenue (the full 80% instead of 70% for standard users) and the option to advertise and sell up to 20 tracks. Each track uploaded to a Pro account is also assigned an International Standard Recording Code (ISRC) and is eligible for chart entries in the UK, America and 20 other countries. Digital Rights Management (DRM) can also be applied to tracks on Pro accounts.

www.7digital.com

Tube MP gets USB

The newest addition to ART's desktop preamp range is the Tube MP Project USB. It provides, as the name suggests, USB connectivity alongside all the features of the standard Tube MP Project preamp. The unit is a single-channel phantom-powered preamp with a 12AX7 vacuum tube. It's got line, instrument and microphone inputs, with a total gain range of 70dB. There's also LED metering and buttons for phase inversion, a high-pass filter and a -20dB pad. The preamp has two switchable input impedances and an automatic output limiter.

The Tube MP Project USB links to the host computer via a standard USB 1.1 connection and ships with an external power supply. Both the Project and Project USB can be stacked one on top of the other, allowing multiple preamps to occupy a very small footprint. It will be available in June costing £125. Check out the ART web site for more details.

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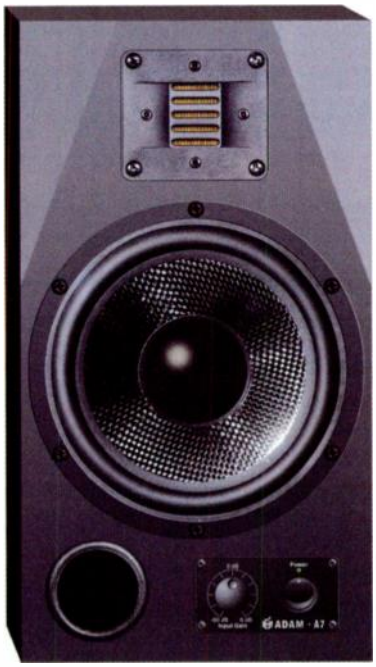
ADAM unveil new nearfield monitors



ADAM studio monitors have earned a strong reputation, being among the select few brands to champion ribbon transducers. Their latest nearfield monitor, the A7 (pictured), features a 6.5-inch woofer below one of their unique ART (Accelerated Ribbon Tweeter) transducers. The A7 is a bi-amped two-way monitor with two 50W (RMS) amplifiers that power the woofer and tweeter independently. Frequency response is quoted as 46Hz to 35kHz with a maximum SPL of around 105dB at one metre. On the back, there are balanced and unbalanced inputs, on XLR and RCA respectively, alongside controls for tweeter level and EQ. The power switch and input gain control are located on the speaker's face along with a bass reflex port. The A7 monitors are available now, costing £702 per pair.

ADAM have also recently announced a new active subwoofer, the Sub8, which features an eight-inch woofer, built-in crossover and 160W RMS amplifier. It has balanced and unbalanced inputs and outputs and claims to cover frequencies from 28Hz to 150Hz (±3dB). Also, an included wireless remote gives you control of the input-level and crossover-frequency controls, which are mirrored on the front baffle in the form of two motorised knobs. Currently, it's only available in black, but ADAM plan to produce a silver one to compliment their Artist desktop monitors. The Sub8 is shipping now, at a cost of £428 including VAT.

Unity Audio +44 (0)1440 785843.
www.unityaudio.co.uk
www.adam-audio.com

Digigram announce pioneering high-resolution soundcards

French soundcard manufacturers Digigram have updated their VX PCI soundcard range with two new high-resolution products, the VX222HR and the VX222HR Mic. They both feature 24-bit/192kHz audio processing and two balanced analogue and AES-EBU inputs and outputs. The 'Mic' version has an additional phantom-powered mic preamp and analogue compressor/limiter. Interestingly, its AES-EBU inputs are also compatible with the AES42 standard for digital microphones like Neumann's Solution D, which features A-D conversion directly after the mic's capsule. Microphone control data, including phase reverse, gain and mic status, can be transferred using the AES42 protocol, which uses a standard three-pin XLR connector. The VX222HR and VX222HR Mic are aimed at the professional market, costing £529 and £694 respectively. But bear in mind a digital mic system will cost you at least £5000! Both new products are available now.

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www.scvlondononline.co.uk
www.digigram.com

Arturia release *Analog Factory*

Following on from a string of successful analogue-modelling software synths, Arturia have announced the forthcoming release of another promising virtual instrument. *Analog Factory* borrows sounds from Arturia's other instruments and includes over 2000 presets from the *Minimoog V*, *Moog Modular V*, *CS80V*, *2600V* and *Prophet V* and *VS* software synths. A neat graphical user interface displays preset settings and features level, filter, LFO, ADSR envelope, chorus and delay controls. For ease of use, there are also four Key Parameter knobs, whose functions change according to each preset, as well as eight snapshot buttons, enabling quick A/B comparisons.

Analog Factory is compatible with Windows 2000 and XP and Mac OS X, including the new Intel-based Macs. It can run in stand-alone mode, using ASIO, Direct Sound or Core Audio, or can be used within a suitable host as a VST, Audio Units or RTAS plug-in. *Analog Factory* is expected in July, costing £149.99 in the UK.

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

AMS Neve launch battle of the bands competition

Hardware supremos AMS Neve have announced 'Nevolution', a global battle of the bands competition aimed at aspiring groups and individuals. Applicants are invited to post 30-second MP3 clips of their work on the Neve web site. There will then be four selection stages, with members of AMS Neve's on-line community choosing three winners from each heat. Then, the remaining twelve participants will enter the final stage, where a panel of top industry professionals will judge their work and choose a winner. Finally, AMS Neve will then transport the winner to the nearest studio with a Neve console, where Rodney Jerkins (aka Darkchild) will produce their track.

There are lots of other prizes as well, with each heat winner receiving a Gibson Les Paul guitar, Sennheiser mics and headphones, and Emu monitors and audio interfaces. The final winner will get a Neumann TLM193 studio microphone and there will also be prizes from Universal Audio, including their UAD1 card and a selection of plug-ins. All terms and conditions can be viewed on-line and all submissions must be the artist's original work. So what are you waiting for? Join the Nevolution!

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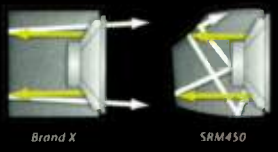

The SRM450's rugged molded cabinet is asymmetrical to trap internal resonances.

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.

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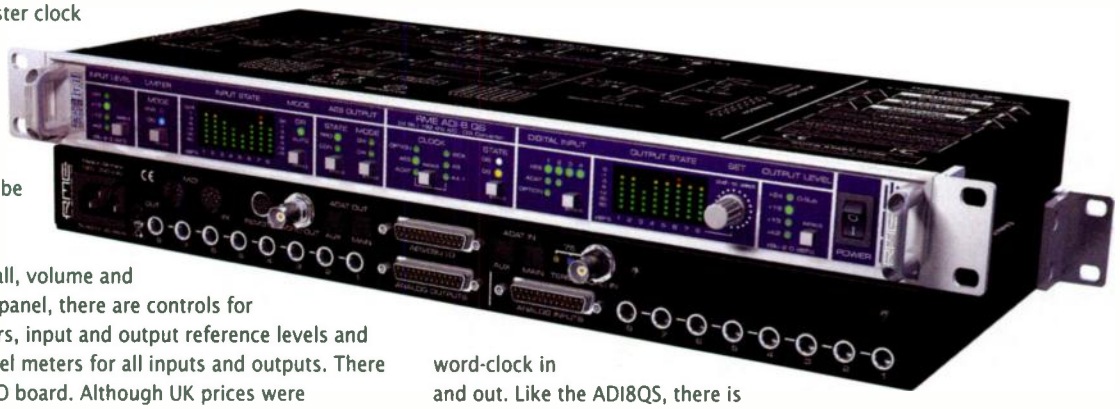
RME ADI8QS and Micstasy

New A-D/D-A and mic preamp units

RME released two new products in their high-end studio equipment line-up at the recent Audio Engineering Society event in Paris. The ADI8QS, the latest in their ADI range, is an eight channel A-D/D-A converter. It features, as you may have expected, eight analogue inputs and outputs, which are on TRS jack and D-Sub, eight AES-EBU inputs and outputs — capable of eight-channel conversion at up to 24-bit/192kHz — and two ADAT inputs and outputs, which are also capable of carrying 192kHz signals using the SMux protocol. The unit also includes RME's Steady Clock, Intelligent Clock Control, Sync Check and Sync Align technologies, which provide a stable master clock signal and apply jitter suppression to incoming sync signals. Word-clock inputs and outputs allow both master and slave operation and the unit can be controlled by MIDI. What's more, an included remote control features handy recall, volume and dim controls. On the front panel, there are controls for analogue and digital limiters, input and output reference levels and clock source, alongside level meters for all inputs and outputs. There is also an optional MADI I/O board. Although UK prices were unavailable at the time of going to press, the ADI8QS's published price of 2000 Euros should give you a rough idea of what to expect. It should be shipping in August.



The second new product is the Micstasy (below), which offers eight of RME's finest microphone preamps in a 2U multiple-purpose unit, catering for both mic-to-line preamplification and A-D conversion. Each mic preamp has an impressive gain range of 85dB, handling input signals of up to +30dB. The Micstasy has flexible I/O options, with mic and line inputs on XLR connectors and line/instrument inputs on front-panel TRS jacks. Analogue line outputs are on XLR, mirroring those of the digital section — effectively acting as a mic splitter or thru box. The digital output section features dual ADAT optical connections (capable of carrying four channels of 96kHz audio each), eight channels of AES-EBU and word-clock in and out. Like the ADI8QS, there is a MADI option board available, and several Micstasys (or ADI8s, for that matter) can be combined to transmit up to 64 channels down a single MADI cable.

Unusually, the front panel is almost completely devoid of the knobs and switches that appear in abundance on other mic preamps — there is only one knob and a single 'on/off' switch per channel, as Micstasy's Auto Set and remote control features handle all the on-board functions. In operation, the user 'sets and forgets' the preamp gain using the front-panel knob, while Auto Set reduces the gain if it senses the signal level creeping too high.

The Micstasy doesn't come cheap, at around 3500 Euros (a UK price wasn't fixed when we went to press), and it should be available in August.

Synthax +44 (0)1664 410600. www.synthax.co.uk

www.rme-audio.com

Associated Board launches remix competition for young people

The Associated Board of the Royal Schools of Music (ABRSM) have launched a new remix competition for young people. As part of their Sound Junction scheme (www.soundjunction.org), the competition encourages youngsters to compose and remix in any genre they like, using an interactive on-line sequencer and sound bites from the web site. There are four age categories, but entrants must be under 20. The overall winner of all four categories will win a day with remixer and producer Colin Emmanuel (aka C Swing — Jamelia, Mary J Blige and Beverly Knight) in London's Miloco studios, and the winners of each age group will get their very own copy of Steinberg's *Cubase SX 3*. Entries must be in by the 10th July and up to five compositions can be submitted per applicant. Check out the web site for more information.

www.soundjunction.org

New facilities announced at MANCAT

The opening of a new recording facility and 300-seater outdoor live music venue has been announced by MANCAT (Manchester College of Arts and Technology) at their Openshaw site in East Manchester. The Amphitheatre — dubbed 'The Amp' — has extensive sound and lighting systems and acoustic treatment, making it the only one of its kind in the North West. The Amp will be opening on the 23rd June with a summer season of over 30 events, using students from two new HND courses as cast and crew. Along with the venue, a new recording facility is being built with two live rooms and a Pro Tools HD3 rig. It's due for completion at the end of August. The new HND courses in production and performance bolster MANCAT's current course line-up, which includes Music Practise and Music Technology. For more information, check out their web site.

MANCAT +44 (0)161 953 5995.

www.mancat.ac.uk

reality **check**

Ready for a dose of reality?

When choosing reference monitors for mixing and music production, accuracy is essential. Speakers that sound "good" on first impression may not necessarily be accurate. You need an honest reference for your mix. Not monitors that have been tweaked or coloured to sound impressive.

The new HS series gives you the perfect reference point. If your mix sounds good on these, it will sound good on anything.

HS Series drivers are housed in bass reflex cabinets, and reproduce a surprisingly tight low end with a smooth, high frequency response to beyond 20 kHz. And, because units are sold and packaged individually, it's easy to build flexible monitor setups and 5.1 systems.

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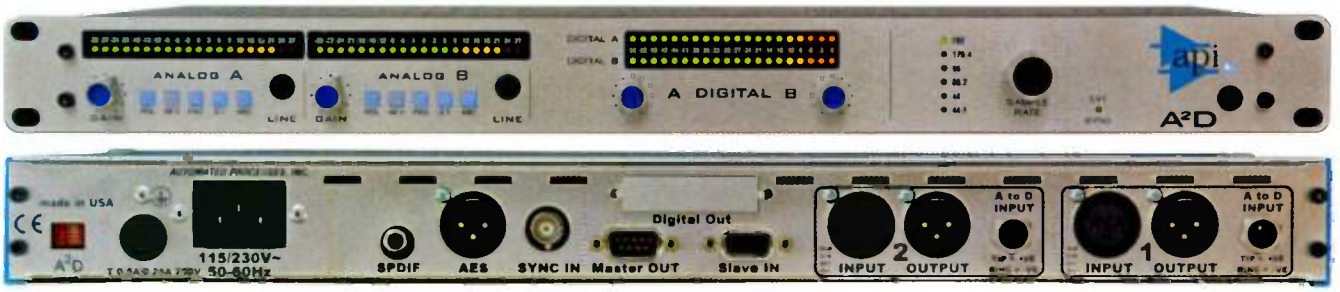
YAMAHA

www.yamaha-music.co.uk

API A2D Dual-channel preamp with digital out



High-end hardware manufacturers API have expanded their range of top-notch analogue equipment with a two-channel mic preamp featuring S/PDIF and AES-EBU outputs. The A2D essentially contains two of API's 312 preamps — as found in the 3124+ and 3124MB+ mic preamps — which feature line and mic inputs, an XLR line output and an insert point on TRS jack, allowing external processors to be inserted before A-D conversion. On the front panel, there are controls for phantom power, phase reversal, a -20dB pad and '2:1' mode, allowing the unit to run at even higher gain. Each preamp has 20-segment LED metering.

Both digital outputs have individual level control and metering, while six sample rates can be selected, from 44.1kHz right up to 192kHz. The unit has BNC word-clock input and useful chaining I/O, which allows multiple A2D units to be linked together. The A2D costs £1410 including VAT and is shipping now.

KMR Audio +44 (0)20 8445 2446. www.kmraudio.com **www.apiaudio.com**

Stage Electrics announce July safety courses

UK-based company Stage Electrics, who deal in all things live, have announced four new training courses aimed at improving safety in the entertainment industry. The courses, which will be held at the Birmingham Hippodrome between 3rd and 6th July, will look at basic health and safety, pyrotechnics safety awareness, electricity at work and 'working at height' regulations. Each one costs £141 including VAT for a full day of training. Successful completion of the pyrotechnics course entitles free membership to the Association of Stage Pyrotechnicians, although applicants must be over 18 for this course. For further information, and to book courses, visit the training area of the Stage Electrics web site.

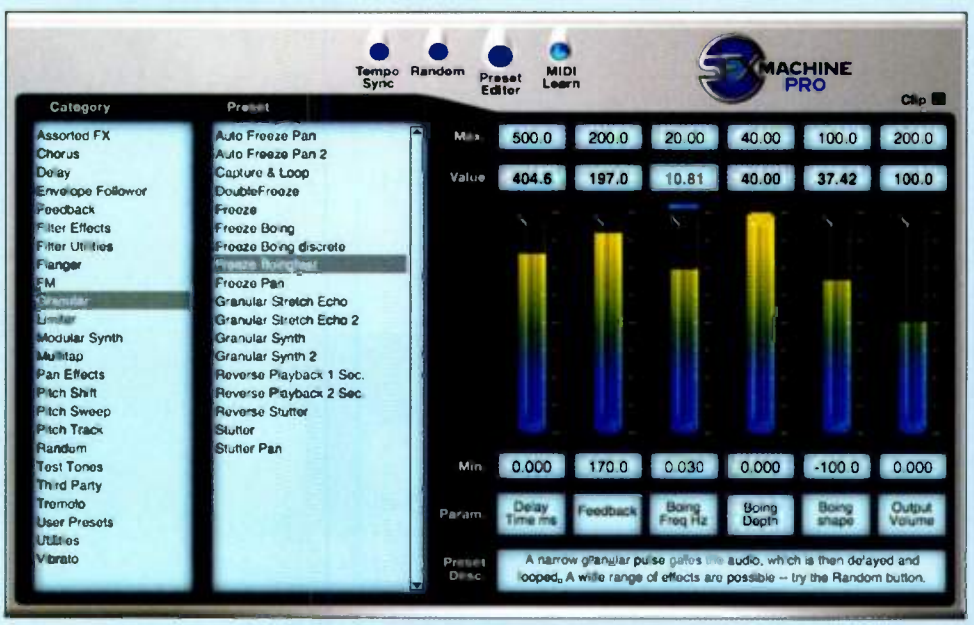
Stage Electrics +44 (0)117 938 4000.
www.stage-electrics.co.uk

SFX Machine goes Pro

The *SFX Machine* concept centres around a multi-effects processor with modular preset editor and eight stereo effects modules. The original *SFX Machine RT*, which was reviewed in *SOS* July 2003 (www.soundonsound.com/sos/jul03/articles/plugin0703.asp), focused on sound design for games, radio, film and TV, at first running only on Macs in Adobe's Premiere format. Since then, it has gained VST and Audio Units compatibility, and has become usable in most popular audio applications on both PC and Mac.

The most comprehensive version of the software to date, *SFX Machine Pro*, has been updated and ships with over 300 presets — which can be arranged using the preset editor — boasting unusual names such as Freeze Boingfest, Fuddizer and Random Pigeons! *SFX Machine Pro* is available now, at a download cost of £132. A demo version can also be downloaded, so you can try before you register.

www.sfxmachine.com



The Price Of Fame Just Came Down

Since inventing the first MIDI Multitrack Sequencer (the Pro 16) in 1984, and introducing Virtual Studio Technology (VST) in 1996, Steinberg has always been the world's no.1 for computer-based music production - the choice of home and professional studios everywhere.

The current Cubase family offers the coolest, most powerful technology available whilst remaining totally intuitive and user-friendly. **Cubase SL3**, for example, combines full-featured audio and MIDI recording and editing, virtual instruments and powerful audio mixing with the added flexibility of loop and pattern-based arranging and mixing.

Add-in realtime time-stretching and pitch-shifting, and the unique 'Play Order Track' arrangement tool and it becomes clearer why well over a million musicians realise their musical dreams with Cubase.

And there has never been a better time to become part of this growing community. Cubase SL3 is now available from your Steinberg dealer at just £199.99 srp.

Further information, specifications and system requirements at www.arbitermt.co.uk

Cubase SL3 - now only £199.99 srp

Also available:

- **Cubase SE3** - Fantastic value entry-level music production system
- **Cubase SX3** - The ultimate professional Digital Audio Workstation

steinberg
Creativity First

Emu announce X2 upgrades and add-ons

Owners of Emu's *Emulator X*, *Proteus X*, Digital Audio System and Xboard products can benefit from a range of newly announced upgrades and add-ons. *Emulator X* and *Proteus X* users can upgrade to *X2* versions of their software for £60 and £40 respectively, while owners of the Digital Audio System and Xboard hardware can buy *Emulator X2* for £190 and *Proteus X2* for just £120. The *X2* software instruments offer a range of useful and creative new features, such as *Emulator*'s Synthswipe function, which automatically creates full sampler patches from any audio input or software instrument; Twistaloop, which analyses and dissects audio into rhythmic components; and Transform Multiply, a convolution DSP process that fuses two sounds into one.

The upgrades are initially downloadable from Emu's web site and are only available to current owners of Emu products.

Creative +353 1 820 6444. www.emu.com

Zero G release 10GB *Total REX*

Sample surgeons Zero G have released their biggest single collection of REX sound files to date. *Total REX* contains over 15,000 REX 2 files, consisting of drum loops, instrumental riffs and 'groove inspiration' from 42 of Zero G's popular sample packages. The files can be used within any program that recognises the REX format, allowing the user to sequence individual slices of samples without changing pitch. Usefully, the files are individually accessible, rather than being packed together in, say, Refill or SAGE format. The 10GB set is available now through Time + Space Distribution for just £99.95.

Time + Space
+44 (0)1837 55200.
www.timespace.com
www.zero-g.co.uk.

Toughen up your mixes

Strong Hire launch 'Pro Tools and Engineer' package

Strongroom studios' rental branch Strong Hire have announced a new package aimed at semi-professional and home studio producers. The 'Pro Tools and Engineer' package is exactly that — a professional Strongroom studio engineer with a Pro Tools HD rig, along with anything else you might want. The package is ideal for anyone who wants a pro-sounding vocal, drum kit or guitar track (the possibilities are endless), but doesn't have the recording equipment — or ears, as is sometimes the case — to achieve it. Each client is treated individually and prices start at around £300 for the most basic setup, although a comprehensive pre-production stage is included for all clients. During pre-production, Strongroom representatives listen to the client's work and advise on processes to be completed before the session, such as removing effects and bouncing files to WAV or AIFF format ready for Pro Tools.

Obviously, pricing depends on the equipment that the engineer brings along, as well as transport and accommodation costs (where applicable). Strong Hire's wide range of equipment is available and the options are almost unlimited. For more information about the Pro Tools and Engineer package, phone or email the Strong Hire team.

Strong Hire +44 (0)20 7426 5150. www.stronghire.com hire@stronghire.com

Don't Panic!

Owners of TC Electronic equipment will be pleased to hear that Panic Music Services have been appointed sole UK service and spares provider for all TC Electronic products. For more information, check out their web site.

Panic Music Services
+44(0)1954 231348.
www.panicmusic.co.uk

Discovery Sound launch download site

Japanese sample creators Discovery Sound have announced an English version of their sample download site. Specialising in ethnic samples, Discovery Download features hundreds of multi-layer and one-shot samples as well as loops and sound effects from a wide variety of sources. Best of all, users can download the samples and patches individually, from $2 each, in formats including *Halion*, *Kontakt* and REX 2.

www.discoverysound.com/dl/en

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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.
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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

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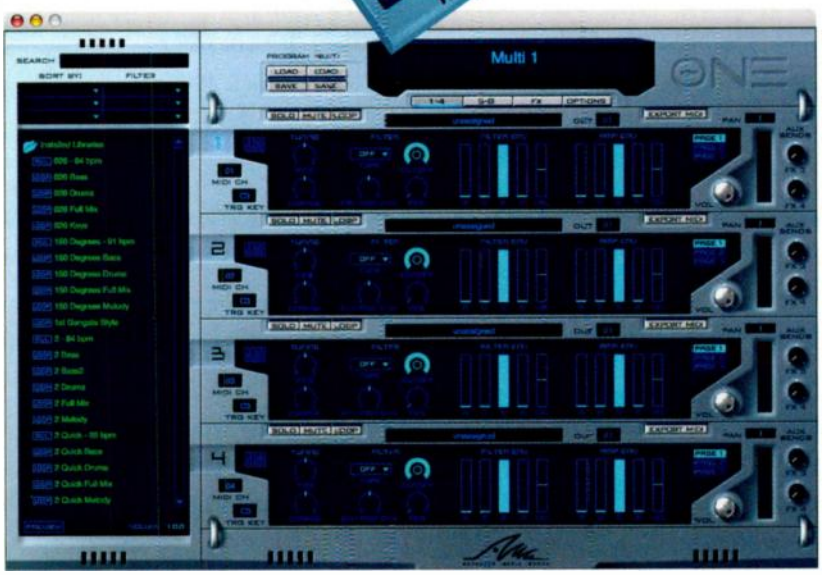
One from AMG

Building on their long history in sample CD production, AMG's latest release is *One*, a loop-based software environment that introduces the company's new range of virtual instruments. *One*'s GUI (Graphical User Interface) has eight individual sample players, allowing up to eight different loops or hits to be independently spliced, reversed, pitch-shifted and nudged in each instance of the plug-in. *One* comes complete with a 4.5GB sample library that includes 3500 loops, over 10,000 individual hits and 312 multisamples. What's more, it can load REX, WAV and AIFF files, allowing you to use your own samples. There is a comprehensive effects-processing section with four effects engines and a multi-mode filter, all with MIDI-controllable parameters. The instrument can be used as a VST or Audio Units plug-in within any suitable audio software on both Mac and PC.

To top all that, there's the price. The *One* virtual instrument with sample library costs just £150 and can be purchased on-line. Watch out for forthcoming virtual instruments from AMG, including cut-down versions of *One* and sample expansion libraries.

www.samples4.com



Chronostream 2 gains Direct X support

Psoft's pitch and time manipulation software for Windows, *Chronostream 2*, has been equipped with Direct X support, allowing it to run as a plug-in inside other software. *CS2* was reviewed in *SOS* March 2006 and was praised for its simple, user-friendly interface and "pretty decent" time-stretching and pitch-shifting, along with an attractive price tag. However, it could only work as a stand-alone effect at the time, with no plug-in capabilities. That has all changed now and, apart from the new Direct X support, the latest version (*Chronostream* v2.25) has had more functions added and its PHISYX algorithm updated — improving the software's low-frequency response. *Chronostream 2* can be purchased from Psoft's web site and works out at about £80, after conversion from the Japanese Yen price.

www.psoft.co.jp/audio/en/products/chronostream

Audio Ease announce *Altiverb* for PC

Plug-in designers Audio Ease have released their widely acclaimed *Altiverb* convolution reverb for Windows XP. The plug-in, which was formerly only compatible with Mac OS X, uses impulse responses from a range of famous locations and studio equipment, as well as some more unusual spaces, such as an aeroplane cockpit. *Altiverb* v5 for Windows XP will run inside any host software that supports RTAS or VST plug-in formats but, like all convolution software, it will need a fairly speedy machine. The new version is available now, costing £399 and is available as a free download for current *Altiverb 5* users. Watch out for new products from Audio Ease, who are in the process of creating cut-down versions of their software, as well as introducing Intel Mac support for the current range.

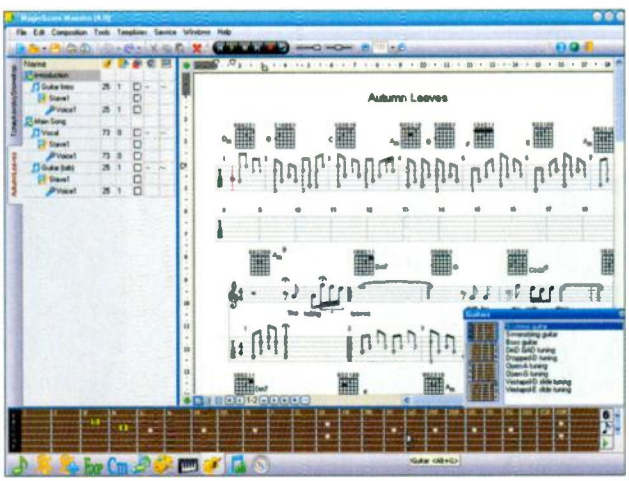
Unity Audio +44 (0)1440 785843.

www.unityaudio.co.uk

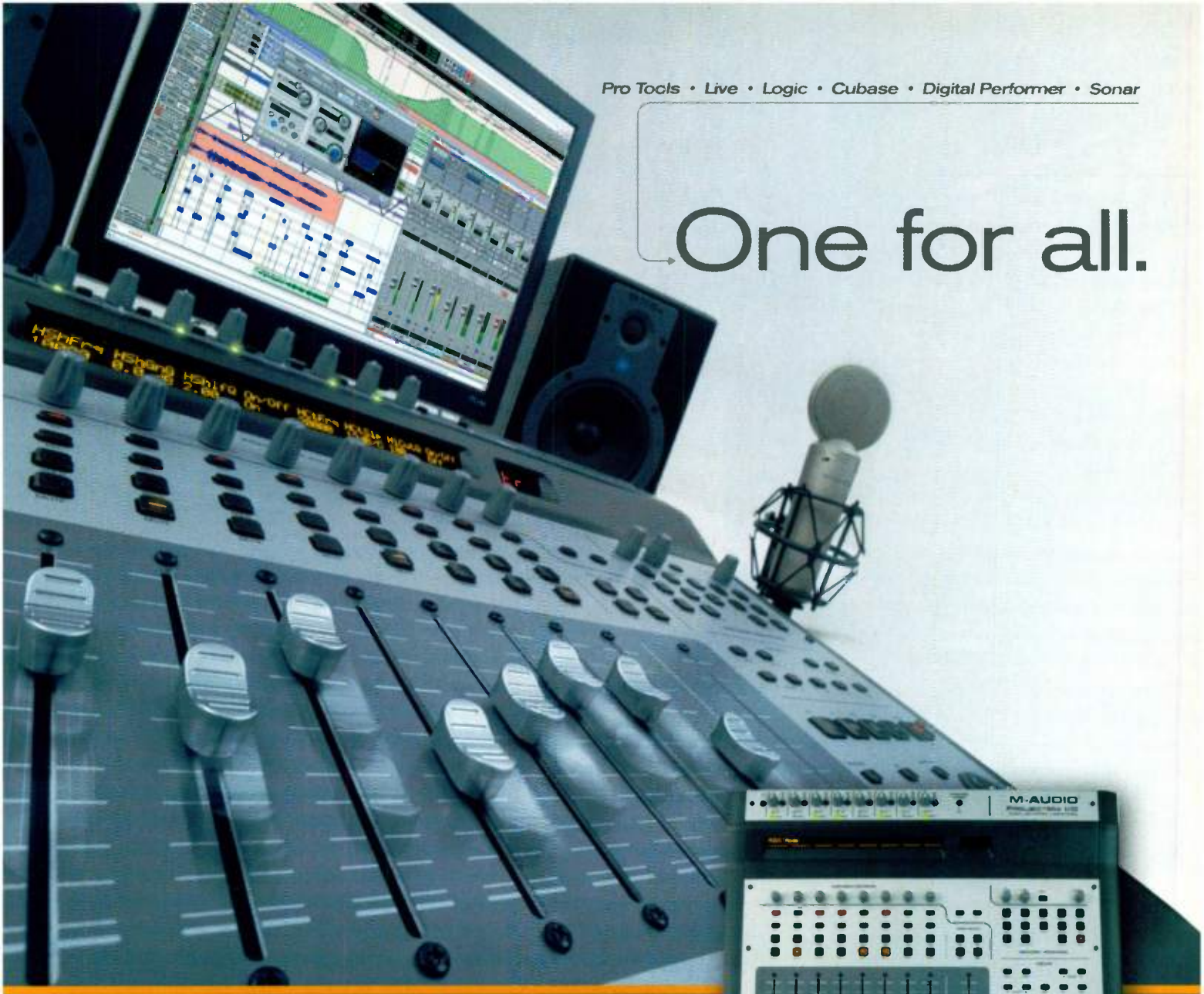
www.audioease.com

Maestro scores!

Web-based software designers DG Software have released *Maestro*, an updated version of their budget-friendly *Magic Score* notation programme for Windows. For a mere $50 (around £26), *Magic Score Maestro* includes all the tool-based editing features you'd expect from a top-spec notation package and, along with the ubiquitous virtual piano, a virtual fingerboard that allows you to program chords for 6-string and 7-string guitars. *Maestro* can also import and export MIDI files and export scores in graphical formats including BMP, JPEG and GIF. It has minimal system requirements and is able to run on anything from a 200MHz Pentium PC to a high-spec Windows XP-equipped machine. For more information, check out the web site, where you'll also find details of *Classic* and *School* versions of *Magic Score*.



www.musicaleditor.com



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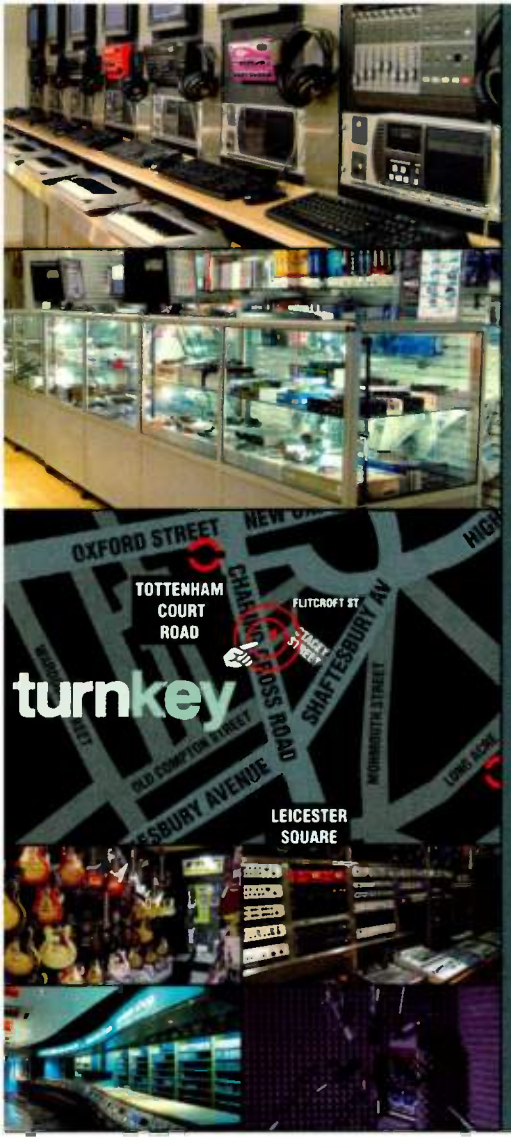

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USB AUDIO INTERFACE

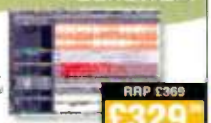
M-AUDIO

The Black Box is a Playstation sized USB audio interface offering balanced TRS jack stereo I/O via 24 bit 44.1kHz converters as well as S/PDIF digital I/O, and complete with guitar jack and mic preamp inputs up to M-Audio's famous quality. The device includes internal audio effects processing DSP with a penchant for guitar amp modelling, offering 12 legendary cab simulations based on well known classic combos including models by Soldano, Fender, Vox, Marshall, Hiwatt, Mesa Boogie and others. There'll also wah effects, a talk box, headphone output for practise amp duty and the handy built in tuner, but that's not all. Black Box is also a complete drum machine, ready to groove with 99 built-in patterns from professional players, with tap tempo ease of use and BPM synced options which can be applied to at least 43 of the effects such as the obvious multitap delay. The drum rhythms are joined by arpeggiator groove patterns too (also BPM syncable) and the effects can work on the drums, voice or guitar inputs. Also has pedal inputs for wah, vol, etc. and includes Live Lite 4 software.

RRP £230 **£99*** MMAN-BLACKBOX

SONAR 5 PRODUCER
MUSIC PRODUCTION SOFTWARE

cakewalk



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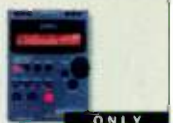

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US-2400
USB HARDWARE CONTROLLER

TASCAM

This outstanding value professional control surface from Tascam gives you a lot of hardware for your money.. with 25 touch sensing 100mm motorized faders, 24 rotary encoders, 8 aux sends, shuttle wheel, joystick for surround panning, footswitch inputs and classic Tascam transport controls with nice red record button! The mix layer architecture lets you control up to 192 discrete channels and perform all track selection, record arming, punch-ins etc, from one powerful unit with Windows XP or Mac OSX compatible USB interface, and template maps matching HUI or Mackie Control.

RRP £1250 **£619** TASC-US2400

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-EMULATPL

AKG SR40/GB40
WIRELESS GUITAR PACKAGE

AKG

It doesn't take much to persuade most musicians or singers to perform live using a wireless radio system, where a good professional quality one is available, but many are put off buying their own because of the fairly high cost of big name brands and the discovery that very cheap ones from obscure manufactures don't work very well. Fortunately AKG have a budget series that really is the exception to this rule ' the affordable Guitar Wireless package is a truly neat and high audio spec solution that won't wreck your tone. Includes GB40 jack plug integrated UHF radio transmitter which plugs discretely into the side of most electric guitars (Les Paul style)or semi-acoustics, flush or swivels round to fit exactly the shape of the Strat type recessed sockets. Gives approx 8 hours use from one AAA standard alkaline battery and inludes the reliable and well proven SR40 single ariel receiver base unit with standard line level output to your mixer. Perfect for guitars or basses! Other AKG wireless systems (handheld mic, tie-clip etc) are also available.

RRP £209 **£99** AKG-SR40/GB40

SONAR POWER STUDIO 25
AUDIO RECORDING BUNDLE

EDIROL

Package includes full software versions of Cakewalk Sonar 5 Studio Edition for pro 64 bit audio production, along with Roland V-Vocal Variphrase effects plug-in, Roland GrooveSynth and TTS-1 GM2 synth, AmpliTube LE plug-in and Lexicon Pantheon LE reverb. USB hardware is Edirol UA25 stereo 24 bit 96kHz audio & MIDI interface.

ONLY **£329** EDIR-PO R25

APHEX 204
AURAL EXCITER

APHEX

The Exciter™ was invented and trademarked by Aphex. Clipping distortion, HPF filtered and mixed back into the dry signal at a subliminal level, increases the sence of source proximity and extends the apparant bandwidth upwards. With the newest 204 model, a wider HF frequency range of adjustment and improved optical compressor for the Big Bottom bass enhancement system is incorporated. The 1U rack dual channel processor has fully balanced TRS ins and outs and is available at a lower than ever price!!!

RRP £349 **£169** APHE-204

2488
24 TRACK HD STUDIO

TASCAM

24 track 24 bit hard disk portastudio recorder with 36 channel digital mixer and 64 note polyphonic GM sound module. Records linear 24 bit 44.1kHz to internal 40 GB hard disk and offers extensive virtual tracks, non linear editing, 3 band parametric channel EQ, 8 dynamics blocks, 3 aux sends, loop effects, guitar multi-effects, 4 mic inputs, CD-RW drive, USB2.0 interface, SMF playback, 25 physical faders, large LCD, Wav import and export and professional Tascam reliability.

RRP £980 **£649** TASC-2488

GOLD MIC
VALVE MIC PREAMP

SPL have produced a 2U rack dual channel valve mic preamplifier with line inputs that uses a hybrid amplification circuit with transformerless solid state input preamp stage to deliver enough clean gain for studio quality recordings. This is followed by a secondary valve gain circuit giving the warmth, subtle THD harmonic distortion and soft overload saturation that valves are supposedly famous for. The preamp includes 48V phantom power, polarity reverse, 10dB pad, high pass filter and 'Flair' which is a presence EQ boost (and phase shift) around 6kHz.

RRP £499 **£299** SPL-GOLDMIC

266XL
DUAL COMPRESSOR GATE

dbx

Two channels of classic dbx compression with new Auto-Dynamic™ Attack and Release controls, program-adaptive expander gates, balanced inputs, precision LED metering and sidechain insert. Front panel selection of stereo or dual mono operation, all in a standard 1U rack design and at an unbelievable price. Now with new auto setup mode! ***Entire dbx range also on demo at Turnkey.***

RRP £160 **£99** DBX-266XL

SAMSON C05
CONDENSER MIC

SAMSON AUDIO

The new C05 mic offers Samson condenser technology in a comfortable to use hand held design. It's 19mm shockmounted capsule with a cardiod pickup pattern provides a smooth flat frequency response, which excels in both live performance and recording applications. The C05 has a quality finish topped with gold-plated XLR connectors. Carry case and mic clip [illegible] a [illegible] mic for [illegible] more.

RRP £120 **£49** SAMS-C05

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FACTORY DIRECT **£849** DAVE-EVOLVEKB

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RRP £849
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TANN-S600A

LITTLE PHATTY TRIBUTE EDITION
RETRO ANALOGUE SYNTH

Since his death in 2005, Bob Moog's own legendary status and classic synthesizers have only gained further respect and desirability among synth collectors and music enthusiasts around the world. The very latest model launched this year, in tribute to the father of the modern synthesizer, is here - the hand built, 3 octave keyboard, fully analogue 2 oscillator performance synth combining many of the best qualities of other timeless designs into a single instrument. It's hard to say whether Bob himself would have viewed the name "Little Phatty" as part of this tribute or not, but at least its a name that won't be forgotten, as neither will Moog. There are 100 usser editable presets and a 37 note keyboard, backlit blue pitch and mod wheels, glide portamento function, ±3 octave shifts and tactile analogue control knob interface with stylish wooden side panels and signature logo on the rear. The synth offers MIDI or Control Voltage compatibility and two ultra-stable analogue VCOs with continuously variable waveforms, plus a classic ladder filter LPF and 4 stage envelope generators.

FACTORY DIRECT
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BIGB-PHATTYT

REASON 3 & OZONIC
SOFTWARE & KEYBOARD BUNDLE

M-AUDIO

Reason 3 is the industry leading all-in-one virtual studio rack and sequencer software, mixing platform, synthesis and sampling workstation, audio multi-effects processing toolkit with open-ended user configurations. Includes Recycle sample looper and REX file player module, vocoder, step sequencer and '808' style drum machine. New to version 3 is the MClass Mastering Suite with Parametric EQ, Stereo Imager, Compressor, and Maximiser with look ahead limiter, plus 'The Combinator' that allows any number and type of Reason module to be combined under a single program for quick access and control. Also boasts huge expanded sound bank with hundreds of new loops and samples. Remote support for hardware integration and new 3.0 patch browser. Included with this is M-Audio Ozonic MIDI controller 37 note keyboard with Firewire for 4 in 4 out audio and MIDI interfacing, self powered via PC or Mac. There are 40 user assignable controllers including 8 pots, 9 sliders and pitch / mod wheels.

RRP £629
£329
PROP-REASON3

VOYAGER V3 ELEC
KEYBOARD SYNTHESISER

The V3 improves both the hardware and the control software for MIDI and the expanded library of 896 new presets in 7 banks. This Limited Edition Electric Blue model is housed in a wooden cabinet finished in sleek black with dozens of cool blue LEDs and blue theory for the control panel. The pitch / mod wheel are also blue light illuminated translucent plastic. Undoubtedly a collector's item appealing to those who prefer a modern studio style look to the traditional wooden cabinet look of the original 70's machine. Obviously on the inside, both synths have the same 100% analogue circuitry for that legendary monosynth charm.

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£1999
BIGB-VOYAGEV3

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WORKSTATION SYNTH RANGE

KORG

Available in 3 sizes the new Triton Extreme sampling workstation synth packs more new features and sounds than ever! 160Mb wave ROM using Korg's HI synthesis at 48kHz native with new concert grand multi-sample, and the many of Korg's best selling libraries. 1344 presets and 50 drumkits, 1280 multis, 256 GM2 patches, 120 note poly, 42x51 modulation matrix, 16MB (exp 96MB) sample RAM, USB, CDRW drive, digital I/O, CompactFlash storage, Wav import, valve preamp onboard, FX, sequencer!

Extreme 76 note... £1399.99 **Extreme 88 note... £1749.99**

RRP £1850
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KORG-TRITEX61

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£49
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PRO TECH

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ONLY
£14
MICA-PSTOP

MPC2500
64-TRACK MIDI SEQUENCER

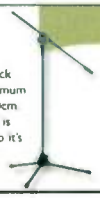
AKAI

The MPC2500 Music Production Center combines a 64-Track MIDI Sequencer with a 32-voice 16bit Stereo Digital Sampler. The MPC is a rock solid sequencing and drum programming machine with 16 velocity and pressure sensitive rubber pads, a wide range of internal effects, large LCD Screen and 16Mb of expandable memory. Designed for professional audio production, the MPC2500's ergonomic interface makes it the ultimate composition studio.

RRP £949
£799
AKAI-MPC2500

MIC STAND

Millenium



Full size tripod base mic stand in black with telescopic boom arm. The maximum height extends to approximately 210cm. This sturdy metal build quality stand is available at an incredibly low price so it's ideal to buy in bulk for your band and singers!

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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.

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

WRH

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 modelling 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.

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Carillon

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Q&A

Your technical questions and queries answered

Q Which PC laptop should I buy?

I'm considering buying a laptop for various audio tasks, including recording school choirs, community music workshops, live DJ'ing (using Ableton *Live*), and writing my own material. I'm looking at a Dell Inspiron 9400 and think I've made my mind up but I was just wondering if you had any advice on this particular machine and if I should be considering any others. I realise you must be very busy writing *the* only music recording mag worth reading but any advice would be greatly appreciated.
Lee Perry

PC Music Specialist Martin Walker replies: Thanks for the compliment, Lee — we do our best! As one of the first models featuring Intel's Centrino Core Duo processor range, Dell's Inspiron 9400 will certainly be fast enough to run lots of plug-ins and soft synths and perform plenty of audio manipulations in Ableton *Live*, whether you choose the 1.86GHz or 2.0GHz model. For recording school choirs and community music workshops I suspect you'll want a quiet laptop, so that no cooling fan noise is picked up by the microphones, and once again a Centrino model should be ideal.

The Inspiron 9400's 17-inch widescreen display and high-end graphics with 256MB of dedicated RAM is generous, to say the least. However, choosing a model with such a large screen does have one disadvantage — weight. At 3.6kg (4.2kg with the AC adaptor) the Inspiron 9400 is generally regarded as too heavy for regular travel which, judging by your list of proposed activities, may influence your final decision.

Don't forget also that you've left out the most important gear item in your query — the audio interface. Recording school choirs and community music workshops could require plenty of microphones, so in a way you're putting the cart before the horse — perhaps choosing a suitable multi-channel interface should come first. Indeed, were you only to be carrying out these activities, many people wouldn't recommend a laptop at all, instead favouring some sort of hard disk recorder or digital multitracker with built-in mixer and mic preamps, so you'd end up with less boxes to cart around.

However, with your DJ'ing and own music composition in mind, let's return to the proposed laptop. Many musicians would look down their noses at one aspect of its performance — the hard drive. At 5400rpm, this seems out of sync with the rest of the specification. Although I'm perfectly happy with the performance of the Seagate Momentus 5400rpm hard drive in my own laptop, which is capable of recording and playing back at least 30 audio tracks at 24-bit/96kHz, most musicians nowadays tend to look for laptops with 7200rpm hard drives, particularly if they intend to record using the more demanding 24-bit/96kHz audio format.

Of course, like lots of other musicians, you could add an external 7200rpm hard drive, but that's yet another box to transport to your live recording sessions. This brings us to another hurdle for PC-music laptop performance — the Firewire controller chip found on the laptop's motherboard that's in charge of Firewire duties, which can be crucial to achieving glitch-free Firewire audio recording and playback.

There are three main audio interface formats suitable for laptop use: USB, Firewire and PCMCIA. Since the Dell Inspiron 9400 features one of the new Express Card 54 slots rather than a PCMCIA one, you're restricted to the first two, but fortunately buying a Centrino-based laptop guarantees you an Intel USB controller for maximum compatibility with USB audio interfaces. However, there are still very few multi-channel USB 2.0 audio interfaces available, so the majority of musicians who need multi-channel recording capability opt for a Firewire audio interface.

It's generally accepted that Texas Instruments' controller chips are more widely compatible with Firewire audio devices, and most interface manufacturers provide some sort of list of compatible chips on their web sites. However, it can be difficult to find out which Firewire controller chip a particular laptop uses, unless its manufacturer is prepared to tell you — and a mistake here will be expensive to rectify.

One final factor can prove very frustrating for musicians when choosing a laptop: ground loops. Many manufacturers use three-wire earthed power supplies for their laptops, particularly if they feature metal cases, and this extra earth connection is notorious for causing ground-loop problems such as background buzzes, whistles and other noises in your audio, which change during hard drive activity, when you move your mouse, and during graphic redraws. I'm afraid Dell are one of the most notorious in this respect, and I specifically mention the Inspiron 8200, 8500 and 8600 models as causing such problems in my PC Music FAQs ▶

Dell's notebooks offer good specs at decent prices, but you may find that they are not ideal for audio.

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Q&A

forum, at www.soundonsound.com. You can cure such problems with a systematic approach and DI boxes between your audio interface and other gear, but it's added frustration and expense.

For this reason, I generally recommend one of two approaches when choosing a PC laptop for music. The safest is to contact a specialist music retailer (many of whom advertise in the pages of *SOS*) and discuss your requirements with them, as they not only choose their laptop components carefully for maximum compatibility and minimum likelihood of ground-loop problems, but can also advise on the most suitable audio interface to go with the laptop, depending on what tasks you want to perform.

You may pay a little more than you would for a mass-market model, but you'll get technical backup from experts who know all about music hardware and software — after all, most mainstream laptop manufacturers have little or no idea of the particular technical requirements of the musician, which are rather different from most ordinary users. I bought my Centrino laptop from a specialist music retailer, as have members of *SOS* staff, simply to avoid all the potential pitfalls and possible strife.

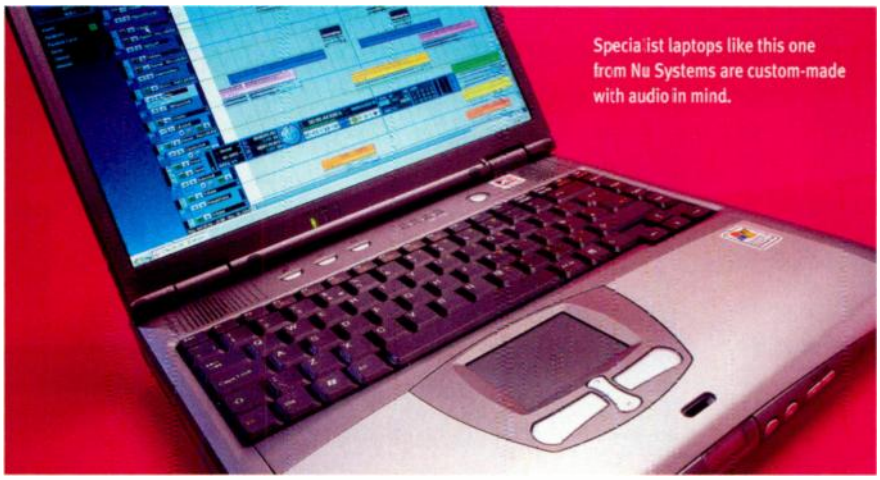
The second approach is to post your suggested laptop model on the *SOS* Forums and ask if any other musician has bought it, and, if so, what experience they have had. Such answers are, of course, no guarantee of a problem-free purchase, and you're likely to have the added confusion of a host of other models being recommended as alternatives, and others to avoid, but at least this approach is better than buying blind. Whatever model you decide on, choose your audio interface at the same time, and try to buy from a dealer who is prepared to give you a refund or exchange on the laptop in the (hopefully unlikely) event of the two ending up in conflict.

Ultimately there are hundreds (possibly thousands) of different PC laptop models available, all including a slightly different combination of components such as motherboard, CPU, RAM, hard drive, graphics, screen, selection of ports, and so on. Such competition keeps prices keen, but it does make choices more difficult for the musician!

Q How do I compensate for latency in *Logic*?

I use a G5 with *Logic Pro*, a TC Electronic Powercore, a MOTU HD192 interface and a variety of plug-ins, but have recently discovered the wonders of running already recorded tracks through outboard units and back into my DAW, to give me some different flavours in my mix. The only problem with working like this is latency. Ideally I'd like to be able to set a nudge value in *Logic* so that whenever I record like this I automatically just nudge the effected track back into place. *Logic*, however, doesn't appear to have a user-defined nudge, and also I've no idea how to work out the exact latency. At the moment I'm moving tracks by eye, but this is a bit hit and miss, especially for drums.

Peter Diggens



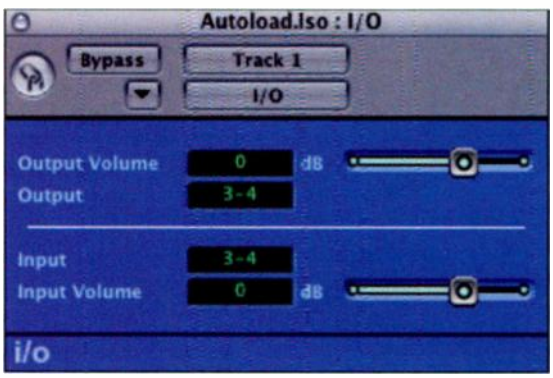
Specialist laptops like this one from Nu Systems are custom-made with audio in mind.

Editor In Chief Paul White replies:

Latency is a complex issue and it varies depending on how many other plug-ins are running and whether or not the delay compensation is set to 'all' or just to tracks and instruments. Latency will also increase when you use one or more Powercore plug-ins. That means that while you can determine an offset time for a given setup, it may change if you bring in new plug-ins or take out one that you used previously. Where the outboard device is a reverb, you don't really need to worry, as the extra delay simply amounts to a bit more pre-delay, and you can then reduce the pre-delay on your outboard gear. If you're using outboard for mastering, the delay is again unimportant, as everything in your mix will pass through the mastering device and so be delayed by the same amount.

Latency is more of an issue when you need to use something like an analogue compressor or EQ on an individual track. The best way to tackle this is to use the *I/O* plug-in, under 'Helper' in *Logic*'s plug-in menu. It works as an insert point, like those on an analogue console, routing the track's signal to an alternate output on your audio interface — say, three and four, as in the screen shot on the previous page — then back in on analogue inputs, before continuing through the signal chain. The *I/O* plug-in works as part of the automatic latency-compensation system within *Logic* but what it won't do is compensate for any delay caused by the device itself. However, in my experience few outboard devices add enough latency to be of concern anyway.

There are further tricks, using some of



Logic's *I/O* plug-in lets you insert outboard gear using your interface's additional inputs and outputs.

The *Sample Delay* plug-in in *Logic* does what it says on the tin, delaying the track down to sample level.

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

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Q&A

▶ *Logic*'s other tools, such as copying the track, phase inverting it (you can use the *Gain* plug-in, which is also in the Helper section), inserting your outboard device, then using the track delay feature to find the setting that gives you the most cancellation. The problem here is that *Logic*'s arrange window doesn't allow you to make adjustments down to single sample level, so the alterations may well be too coarse anyway. If you are really keen to get things spot on, you could use the smallest value of negative track delay, then insert *Logic*'s *Sample Delay* plug-in and adjust that to find the cancellation 'null point'. But in most cases the delay through the outboard device is only a millisecond or two (less if it is analogue), which means that it has no real musical significance. However, if you are prepared to go to this trouble for each bit of outboard gear you plan to use, you can make a note of the delay settings and then apply them again in future, as they will always be the same as long as you are working at the same sample rate and have the delay compensation turned on at least for the audio tracks in *Logic*.

Q Which active nearfields should I choose for my home studio?

I am currently setting up a small recording studio based on a Yamaha AW16G recording system and require some active nearfield monitors. I have been reading your excellent magazine for some months and, although I am more clued up than I was six months ago, I am still no clearer on what monitors are best suited to me. I want to record mainly British traditional music using a variety of acoustic instruments, including guitars, mandolin, bouzouki, Irish whistle and bodhran alongside vocals. I have a budget of about £200 to £500 for a pair of active nearfield monitors but would be happy to pay more if a noticeable benefit could be obtained. What do you recommend? My studio is about 12 x 9 feet in size.

Vince Jerrison

The Alesis M1 MkII actives offer well-balanced sound and good value for money.

Editor In Chief Paul White replies: In a studio of that size, an active two-way monitor with a five-inch to eight-inch bass

Genelec's 8020A is the baby of their active monitor range, but it still packs a punch.

driver would be ideal. For 'acoustic' music, soft-dome tweeters often give the most natural response (although some metal designs also sound very good) and at the budget end of your price range I'd be inclined to look at the Alesis M1 MkII Actives (reviewed in *SOS* way back in August 2002) and the Fostex PM05 monitors, as I feel these would work well for your style of music and your room size. Tannoy Reveals are also well worth looking at. At the top of your range, I'd be tempted to look at KRK's range of monitors, along with products from Dynaudio and Genelec's recent 8020As. If you can stretch your budget further, the Mackie HR624s should work well for you and there are also some great monitors at the more affordable end of the ADAM range. If you can find a shop that will lend or hire the speakers to you so that you can check they're giving the results you want in your room, it's well worth it, even if you end up paying a bit more.

Note that these recommendations are based on speakers that I've used for a meaningful amount of time, so there may be other equally worthy contenders that I haven't included purely because I don't feel I know them well enough. It's also worth mentioning that proper speaker mounting and positioning is important, as is acoustic treatment — even the best monitors won't give great results in an untreated room.

To find out more, check out our in-depth guide to monitoring and acoustic treatment in *SOS* February 2006 (www.soundonsound.com/sos/feb06/articles/studiosos.htm).

Q How can I make use of my interface's ADAT I/O?

I have an Emu 1820M audio interface and I would like to get more simultaneous inputs into my DAW software using the Emu system's ADAT I/O. I will mainly be using the extra inputs for balanced line sources, but it would be good to have extra mic preamps, too. I also want to keep within a reasonable budget. What should I buy?
SOS Forum Post

Staff Writer Chris Mayes-Wright replies:
This question is becoming more common because of the popularity of ADAT inputs and outputs on affordable audio interfaces like yours.

The ADAT protocol allows eight channels of audio at up to 24-bit/48kHz to be streamed down a fibre-optic cable. If your soundcard or audio interface has ADAT optical inputs and outputs, you can use them to get more channels of I/O to use with your ▶

Q&A

Does your soundcard have ADAT interfacing you're not using? The right add-on box can give you easy access to it.

Behringer's ADA8000 — eight channels of mic preamplification and A-D/D-A conversion, all at a remarkable price.

▶ DAW. Of course, you need something to take the analogue mic, instrument or line input and turn it into the digital ADAT output, and vice versa if you want more analogue outputs.

There are quite a few products on the market that supply A-D and/or D-A conversion, with a wide range of input and output options to boot. At the cheapest end, the 1U Behringer ADA8000 supplies eight phantom powered mic and balanced line-level inputs with balanced line outputs and ADAT optical in and out. BNC wordclock comes as standard and it's also very easy on the wallet, available at under £200 in UK shops. M-Audio's Octane is a 2U, eight-channel preamp with a built-in analogue to digital converter. Compared to the ADA8000, the Octane is twice the price, at £419 in the UK, although you can probably find cheaper street prices. It only has ADAT output — providing A-D conversion but no D-A — but it has some useful additional features, such as an M&S matrix and phase invert switches. Also, the quality of the mic preamps is arguably better than the Behringer's. But, as my heating engineer told me recently, you get what you pay for!

If you've got more money to spend, Focusrite offer two preamps with ADAT options — the Octopre and Octopre LE, which both feature eight decent preamps and occupy 1U of rack space. The Octopre LE has all the features of the Behringer ADA8000 plus switchable low-cut filters on all channels, two instrument inputs and a rather fetching blue VU meter that can be selected to show the output from each preamp separately. However, the ADAT input and output is an optional extra and in the UK the whole package, including the ADAT option board, will cost £478. The standard Focusrite Octopre is similar, although it has individually switchable phantom power and dynamics processing on each channel. Like the Octopre LE, the standard version also needs a digital option board to provide ADAT in and out and there is a further option with additional AES-EBU and S/PDIF connectivity. In the UK, the Octopre with ADAT-only board will cost you £798.

Further up the price ladder is the Mackie Onyx 800R, which offers eight Onyx preamps with plenty of extras, including a feature-packed digital output section that supports sample rates up to 192kHz (with lower ADAT channel counts). It costs around £1000 on the street. Even further up is the Audient ASP008, which supplies eight top-flight preamps and ADAT output for £1228. This is beyond the reach of most people's budget, but it's worth mentioning as you can win one in this month's competition, along with Audient's great Sumo summing mixer (check out page 66 for more details).

Another thing to consider when buying any device with digital inputs or outputs, after the quality of the analogue circuitry, is how you will sync it up to your audio interface, as digital audio transfer always requires sychronisation — whether it's from word clock, or down the same cable as the audio signal.

If you want to use your ADAT-equipped preamp as the master clock, you need to be able to trust that the clock will not transmit excessive jitter — arteficts in the clock pulse signal that can cause audible glitches — to other devices in the sync loop. Conversely, if you want to use your preamp as a slave device, you can — depending on the features of the unit — use an alternative input to receive your sync signal. These include ADAT (of course), AES-EBU or S/PDIF inputs, or a dedicated word clock, which is usually on a BNC connector. If you have a device that you specifically want to use as a master clock, you should ensure that your preamp will be able to sync to it. Referring back to my wise heating man, you get what you pay for here as well, as some low-end digital devices are known to kick out the odd glitch here and there.

For more information on clocking and why it's important, our 'Digital Clocking Explained' article, in *SOS* April 2003, is invaluable. SOS

The Mackie Onyx 800R adds an octet of high-quality, flexible mic preamps to any ADAT-equipped audio interface.

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Roger Nichols: Across The Board

What can we, as engineers or musicians, do to prevent our recorded legacy being lost?

Roger Nichols

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.

Audio preservation is a topic that keeps rearing its ugly head, and will not easily go away. The human species seems concerned about its past, but would rather wait 1000 years and try to reconstruct what might have happened instead of dutifully preserving the information for future generations that will remove any doubt. It is more fun to argue about the multiple possible origins based on incomplete data.

It seems like we want our recordings to slowly fade into oblivion. The record companies still have the attitude that they will not spend money to preserve what they have, but will spend tons of money later to recover something that is gone when they need it for a release. Record companies will not pay for extra archival copies on alternate formats during mastering sessions. Record companies and production companies will not pay for the additional time necessary to correctly document and consolidate DAW sessions so that they can be recalled years later for additional releases in new formats. "We will not pay for the additional time or media. We just want the CD master now for release, and send the multitracks to our office." There are exceptions, but they are few and far between.

I, personally, have made a copy of every project I have ever worked on. I decided to do this in 1970 after seeing the storage facility at ABC Dunhill Records in Hollywood. Tapes were damaged or missing after being stored for only a few months. In 1981 all of the Steely Dan two-track tapes were transferred to digital. The record company could not find the 'B' side of the *Aja* album. We had to use the copy I made during the original mixing. We did not make digital copies of the multitracks, and since then the record company has lost the 24-track tapes of that album.

Formats

I don't really care whether someone likes analogue or digital, but you must accommodate the medium to which you are recording. Some digital formats are more robust than others, so why not print your mix to more than one? You want to print your mixes to analogue tape? OK, it has worked for all the years prior to digital, but analogue tape does not reproduce exactly what you put on it.

When an ad agency is producing a magazine ad, they look at the final proof after it has been printed, and then go back and change colour balance and lighting until the results are what they want. I have almost never seen anyone do that when recording or mixing to analogue tape. The engineer spends three days making a perfect mix, then prints it to the 30ips quarter-inch tape and hangs out in the lounge while the assistant plays the tape back to make sure it was actually recorded. I have only once seen an engineer rewind the two-track and play it back in sync with the actual mix to A/B what the tape did to the sound, and then make small corrections to the mix to somewhat compensate for those differences.

Once the analogue tape has been recorded, there is never any mention of the sound change that occurs over time from the minute the recorder was stopped. The message on the analogue tape starts deteriorating as fast as skywriting messages over Brands Hatch on race day.

Overlooked Remedies

All formats have their flaws. The trick is to admit what they are, correct them in future formats, and figure out how to correct for them when migrating the data to the new ones. You don't complain about CD rot until your music disappears: you are supposed to clone the music to a new format while the error correction can still correct.

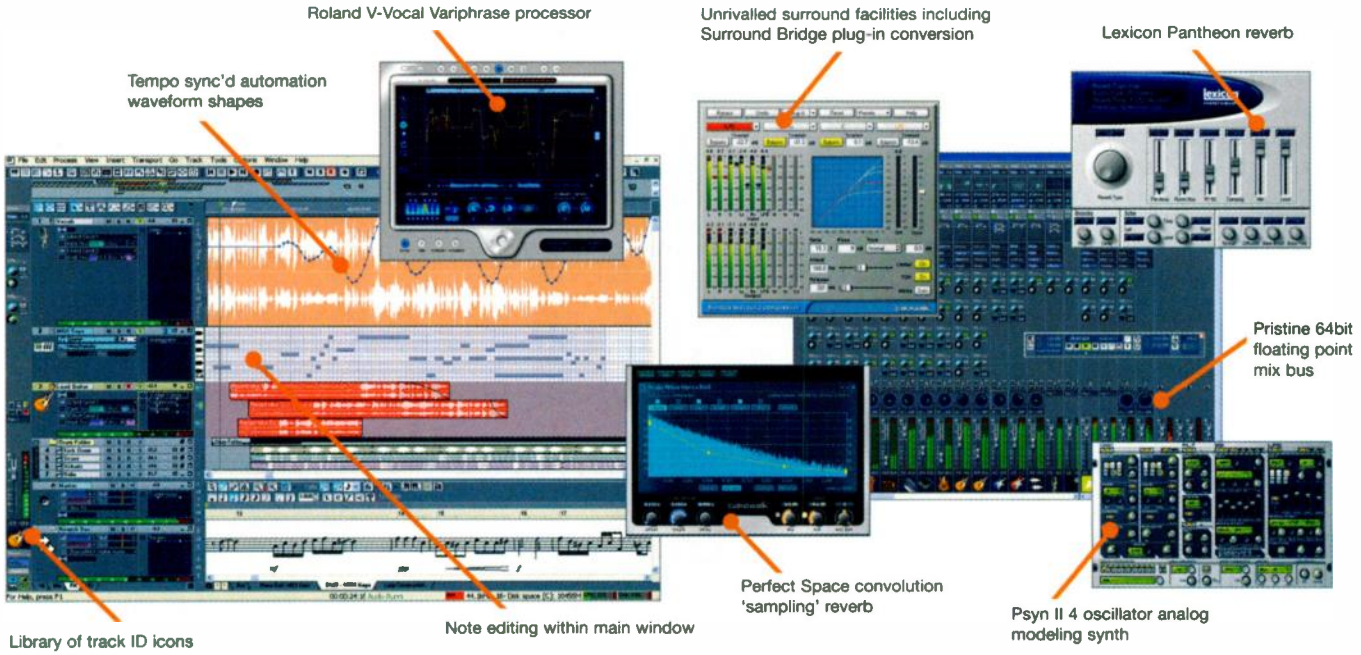
When the only format for mixing to was analogue tape, I did not mix to a piece of tape then copy it. I had a second tape machine recording in parallel to avoid the generation loss. Now the backup was equal to the original. After the advent of digital, I printed the mix to two different formats in case one did not last as advertised. When DAT tape life was in doubt, I transferred to CD-R as audio files and to Exabyte tapes as DDP files while the DAT tapes would still play back. Now whatever I record is also stored on CD-ROM, DVD-ROM, or Blu-Ray as AIFF or BWAV files. Data files have more error correction than audio CDs, allowing for a better chance for recovery.

Ten years ago I submitted to the US Library of Congress a method for recovering audio from cylinders and records photographically. I am glad to see that they ▶

Baking tapes in special ovens can temporarily make 'sticky' tapes playable for transfer to another format, but baked tapes are not usable in the long term.

TIME TO SWITCH?

For years the best selling sequencer in the States, Cakewalk's SONAR has long been at least the technical equal of the other major applications. Although less well known over here, with the release of Version 4 many professionals found that they could no longer ignore SONAR's persuasive combination of powerful features and a superbly easy to learn user interface. Now with the advent of Version 5, SONAR has taken a genuine leap into the lead with a host of new technologies, instruments and effects. This ad isn't nearly big enough to do justice to them all, but check out some of the major features below and make your own mind up - *is it time you switched?*



	Physical Inputs/Outputs	Audio plugin track inserts	Maximum Sample Rate	Audio engine resolution	Freeze Synths & Tracks	A/B Freeze edit frozen data	Full Track Templates	Bus & synth wave preview	Professional Scoring	ASIO loop support	V-Vocal Variphrase	Convolution Reverb	Lexicon Reverb	Synths	Audio hardware support	ASIO	WDM	Digidesign	M-Audio	Edirol	Video output to FireWire DV	WMA lossless encoding
SONAR 5 Studio Edition	unlimited	unlimited	96kHz	64-bit FP	•	•	•	•	•	•	–	–	•	5: (2) Roland synths, groove sampler, SoundFont sampler, analog modeling	•	•	•	•	•	•	•
SONAR 5 Producer Edition	unlimited	unlimited	96kHz	64-bit FP	•	•	•	•	•	•	•	•	•	6: (2)rgc:audio synths, REX synth, (2) Roland synths, groove sampler, SoundFont sampler, analog modeling	•	•	•	•	•	•	•
Cubase SL 3.1	128	5	96kHz	32-bit FP	•	–	–	–	•	•	–	–	–	3: dual-oscillator, virtual bass, drum machine	•	–	•	•	–	–	–
Cubase SX 3.1	256	8	96kHz	32-bit FP	•	–	–	–	•	•	–	–	–	5: dual-oscillator, virtual bass, drum machine, surround pad, mono analog	•	–	•	•	–	–	–
Pro Tools LE 6.9 (and M-Powered)	Digi002:18 Mbox:2	5	96kHz	32-bit FP	–	–	–	–	–	–	–	–	–	4: Reason Adapted 3, Ableton Live Lite 4, SampleTank LE, Expansion BFD lite	–	–	•	•	–	•	•

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have finally funded Berkley National Laboratory to investigate that process. But it took 10 years.

As everyone knows by now, analogue tape suffers from 'sticky shed syndrome'. The tape companies suggested baking the tapes to enable playback temporarily. In 1992 I started using a vacuum process to recover these tapes; I enlisted one of the original scientists who developed Mylar and the oxide binders for DuPont to help develop it. It works perfectly and turns out to be permanent. Tapes processed in 1992 still play back perfectly today, without the increase in distortion as a result of baking.

Everyone also knows that wow and flutter are natural occurrences when recording analogue. You can measure it, but you couldn't do anything about it. Until three years ago. Jamie Howarth developed a system that removes the wow and flutter from analogue tapes. Analogue tape machines use a high-frequency bias signal of around 100kHz during the recording process. This bias frequency actually gets recorded on the tape, and if you manually move the tape by the head at a very slow speed you can hear it as a whistle. Since this frequency is constant, the pitch of it on the playback tape is modulated by the wow and flutter. If you detect the bias and correct the pitch of the program material, you have removed the wow and flutter. Brilliant.

I have heard the process, and it sounds amazing. The difference between the before and after quality is about the same as the difference between 16-bit and 24-bit audio. It is not subtle, it is amazing. The film industry has jumped on it, and has been transferring the audio tracks from old movies that are being re-purposed for DVD. Jamie's company, Plangent Processes (www.plangentprocesses.com) has partnered with Chace Productions in Burbank, California to recover the audio from the six-track mag-tracks.

The record industry is, however, slow to get on board. The comment I have heard most is "Why would you want to change what I have recorded?" The answer is that the process does not change what you have done: it removes an artifact that was introduced by the medium. In the digital world everyone was quick to minimise the jitter caused by inferior clocking, so why do they fight the removal of the analogue jitter recorded by every analogue machine ever used? Maybe a shovel upside the head would help. It works for my mule.

Restoration Works

So what to do if your master recording has decayed and needs work? Forensic audio was a big topic at the Paris AES show this year. Police departments around the world are buying CEDAR systems to clean up bad audio. At the other end of the spectrum there are very inexpensive and even free software programs to clean up your old record collection so you can make your own CDs or MP3s for your iPod. These are great products, but they are only as good as the person using them. As with any new software process, two things happen. The user gets better at using the program as he gains experience, and the software is improved over time. The clicks and pops you could not remove now will be much easier to remove in the future.

To save extra work in the future, make the best flat transfers you can, and save them. Do the processing to the flat transfers instead of processing during the transfer. When the processing improves in the future you can then use the flat transfer and re-process it instead of having to go back and re-transfer the audio. In 1997 we tried to transfer the old Steely Dan two-track masters again with newer technology. Because of the additional 15 years of deterioration, the digital transfers done in 1981 sounded much better than anything we could do in 1997. So, the sooner you make the transfers, the better your results will be.

If you are transferring records, make sure you clean the records first. Learn how to do it properly. There are plenty of companies with helpful information on the Web that will increase your chances of success. Remember, cleanliness is next to high fidelity.

There are flat preamps available without an RIAA or CCIR frequency response curve. Companies like Enhanced Audio (www.enhancedaudio.com) have very good flat preamps, and even the Griffin iMic (www.griffintechnology.com) allows flat transfer with software equalisation. After transferring flat, you can use software curves to recover the original audio more accurately; the curves are complex and inexpensive preamps with curves built-in are most often incorrectly implemented. Also, early recordings before the '60s did not all use these curves, and sometimes the curves were different between record companies and even between different releases from the same record company.

If you are going to remove the clicks, pops and noise from the transfers, the best results are achieved if you perform the processes in the following order: de-click, de-crackle, de-buzz, de-hiss and de-rumble. Performing the processes in the wrong order can mask information needed for the next process or create audible artifacts by not completing a required prior process.

There are hundreds of recordings lost every year because they were not transferred when it was still possible to save them. Take care of your music so others may enjoy it in the future. If you have questions search the Web, contact archival companies in your area, or write to *Sound On Sound*. SOS

Storage And Handling

Audio storage is all too often overlooked. Cool and dry is always good, and vertical storage is very important. Store your records, CDs, DAT tapes, reel-to-reel tapes, videotapes, Mini-DV and DVDs vertically, never flat. Tape, especially, tends to slip down and damage the edges if stored flat, even if they are in a cassette.

Don't touch tape. Don't touch the surface of records or CDs. Handle by the edges or the CD hole. Don't touch the playing surfaces of any recording. Don't store tapes on or near speakers or amplifiers. This may sound intuitive, but every day I see someone lean a tape against a speaker or pick up a CD like it was a ham sandwich.

Make sure the playback machine, be it a tape machine, CD player, turntable or cassette player, is in good working order and adjusted properly. I have many times seen a hungry player eat the only copy of a cassette, DAT or videotape. Demagnetise the machine, clean the machine and use a test tape (and then the tones on the tape that you are playing back) to align the machine. Some new engineers have never even seen an analogue multitrack.

Tape storage done right (top)... and wrong (above).

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Focusrite Liquid Mix

Preview: Firewire-based Dynamic Convolution Processor

Focusrite's Liquid Mix could be the hottest product of the year, using convolution technology to integrate a stellar range of vintage analogue EQ and compressor replicas into your DAW. We got our hands on a prototype to see whether it lives up to the hype...

Paul White

Focusrite started something of a stir in the professional audio market when they released their Liquid Channel, a high-end voice channel that had the ability to emulate a number of different

preamplifiers, equalisers and compressors using a new core technology called Dynamic Convolution, which they licensed from the inventors Sintefex. Straightforward convolution can't reproduce the characteristics of non-linear audio processes such as compression or distortion, which is why Dynamic Convolution was invented. In essence, it analyses the behaviour of the subject piece of audio gear when fed a series of pulses and test signals at multiple gain settings and frequencies. DSP technology is then used to replicate the response of the original device in real time, with full user adjustment of the control parameters. Sintefex are a Portugal-based software development company, though the three directors are British and their hardware R&D takes place in Cambridge not a million miles from the secret hollowed-out volcano lair of *Sound On Sound* headquarters.

Liquid Assets

While the Liquid Channel was aimed squarely at the professional user, technology has a habit of spinning off in more affordable directions, which is why I now have a prototype of Focusrite's new Liquid Mix sitting in my studio. The Liquid Mix combines Dynamic Convolution technology with concepts that are already familiar to many computer musicians. For example, we are all aware of DSP-assisted processors from companies such as Creamware, Digidesign, Universal Audio, Waves and TC Electronic, and we all know about hardware control surfaces that interface with software. The Liquid Mix, in essence, combines DSP horsepower with a hardware user interface that fits into a neat desktop unit and connects to the computer via Firewire. Cosmetically, the moulded unit is very stylish and is reminiscent of a high-end reverb remote controller. All you need to connect up to a six-pin Firewire port is the included Firewire cable, and a PSU (12V AC at 1 Amp) can be connected for those using non-powered four-pin Firewire connectors.

The DSPs run a range of EQ and dynamics processing plug-ins that use Dynamic Convolution to replicate various pieces of classic studio EQ and compressor hardware, so in essence you're getting all the Liquid Channel technology other than the analogue preamp section (which is very expensive to build), available as plug-ins and for a fraction of the price of the Liquid Channel. Furthermore, as all the really hard work is done by the on-board DSP, there's very little hit on your host CPU. The reason this isn't a full-blown review is that Focusrite still need to finish tweaking the code to add the full quota of 'replicas', but shipping of the Mac version is expected by June this year with the PC/Windows version following on a little later.

By the time it ships, the plug-ins that come with the Liquid Mix will be able to run in VST, Audio Units, and RTAS-wrapped VST formats, and so should be compatible with all the mainstream sequencing software. Despite its low cost, the Liquid Mix aims to provide 32 simultaneous channels of EQ and compression, where the user can choose from a large library (20 EQs and 40 compressors) of EQ and compressor types. Additional emulations will be made available on-line and there's even provision to build your own hybrid dream EQ with up to seven bands taken from different equalisers. Each of the 32 channels may be accessed as a separate plug-in within the sequencer in the usual way, and because of the nature of the processors, this will normally be via channel or buss insert points. While the unit can run up to 32 mono or 16 stereo channels at 'normal' sample rates, an optional DSP expander is planned to increase the maximum channel count ▶

▶ at higher sample rates. This will fit into the unit via a panel at the bottom of the case.

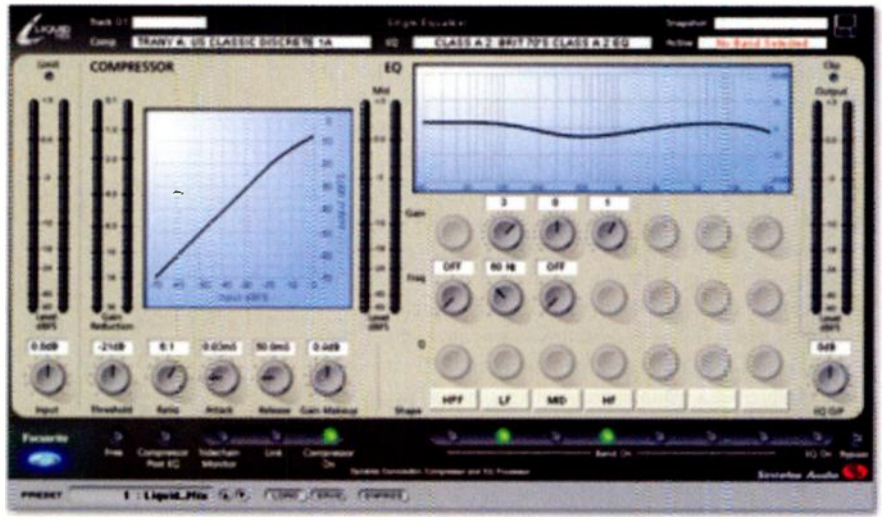
Liquid Control

You can, of course, control the Liquid Mix's processors directly using your computer's GUI if you prefer mousing around, but one of the key aspects of the unit is that it combines all this sophisticated plug-in power with a very practical control surface, which has dedicated continuous controller knobs for the main compressor controls (Threshold, Ratio, Attack, Release and Gain Makeup), and Gain, Frequency and Q controls for a single EQ band. A further data wheel moves between EQ bands and there's also a master EQ output level control, as well as a general input level control for the whole channel. Further functions such as saving and loading patches, selection of model type and even track selection are accessed by small illuminated push switches. There's also a Compare/Go Back function, plus a button that selects the EQ shape for the currently selected band. There's also good metering covering the input level, the amount of gain reduction and the output level, with a further meter covering the level between the compressor and equaliser, which can be switched pre- or post-compressor. LEDs show when limiting takes place and warn of output clipping.

Along the lower edge of the front panel are buttons for switching the positions of the compressor and EQ and for accessing the side-chain monitoring. Stereo linking also has its own button, and there are four further buttons for switching in and out the compressor, the entire channel EQ, a single EQ band or the whole lot. The way the replica plug-ins work is that by default, they have the same control options as the original, so if the original has a stepped ratio control rather than a continuous one, that's what the Liquid Mix gives you. However, if you need more flexibility you can use the Free button to allow you to dial in interim values. A backlit LCD window displays additional information pertaining to patches and parameters, and when the EQ is being viewed, a small version of the EQ curve is displayed.

On the beta model I've been using, the data wheel is used to select the EQ band being worked on, and also to switch between viewing the EQ and viewing the compressor. However, Focusrite have already been asked for some kind of automatic switching system that moves between displaying the EQ and compressor when knobs in the relevant section are moved, and apparently this is on their to-do list.

As I mentioned at the outset, the current version is Mac-only and currently requires Mac OS X Panther (10.3.3 or later), with support for the new Intel Macs included. Obviously, the computer needs a Firewire port, and a minimum spec of an 800MHz G4 with at least 256MB of memory is suggested. Devices that access processing DSPs over Firewire inevitably increase the system latency to some degree because of the need to move data into and out of the computer, but the engineers at Focusrite say the Liquid Mix should not add more than 2 or 3 ms to the latency of a typical system. What's more, as these processors are designed for use when mixing, you can always leave them off until you've finished tracking as latency doesn't matter at all when you're mixing.



If you prefer to control plug-in parameters from the screen, everything is available there too!

Trying It Out

My first attempt to use the Liquid Mix was thwarted by its reluctance to play nicely with the Belkin Firewire hub I had in my system (even when not plugged into it), but once this was removed, the software installed and the Liquid Mix Manager program started, I had the system up and running in moments. In this beta version only the VST versions of the plug-ins are supported so I checked the operation from within BIAS *Peak*, with which the Liquid Mix worked seamlessly. It seems that most or all of the compressors are already implemented, but at the time of writing there's only a handful of EQs, though there will of course be a full set by the time Liquid Mix ships. As with most products that seek to emulate others, there are cryptic clues as to what pieces of equipment have been replicated but no overt naming other than for other Focusrite devices. Already there are Focusrite Green and Red equalisers, not to mention a British Green optical compressor and a wide range of other US and UK models, both tube and solid-state.

That these are available as plug-ins at all is impressive enough, but having knobs to adjust them and real hardware meters to show you what is going on is pure luxury. As mentioned earlier, the controls are presented as on the original with the same size steps where switches were used, but the Free button allows you to override this limitation if you wish to. Where the original compressors had side-chains, and where the host software supports audio routing to side-chains, you can use the compressor for ducking or de-essing as you might with a hardware device, while the side-chain monitor button lets you hear exactly what the side-chain is hearing.

This is only a preview, but already the Liquid Mix is an impressive piece of kit that looks perfectly at home on the desktop. The ability to use so many high-ranking equalisers and compressors at the same time and without imposing an excessive load on your CPU is very liberating. Inevitably, some of the equalisers and compressors will sound more obviously different than others, but Focusrite and Sintefex have gone to great lengths to get these replicas sounding as close to the originals as possible. If you've always hankered after a rack full of classic outboard gear but never had the money or space, the Liquid Mix seems like the perfect solution. We'll bring you the full review when the release version is available. SOS

Information

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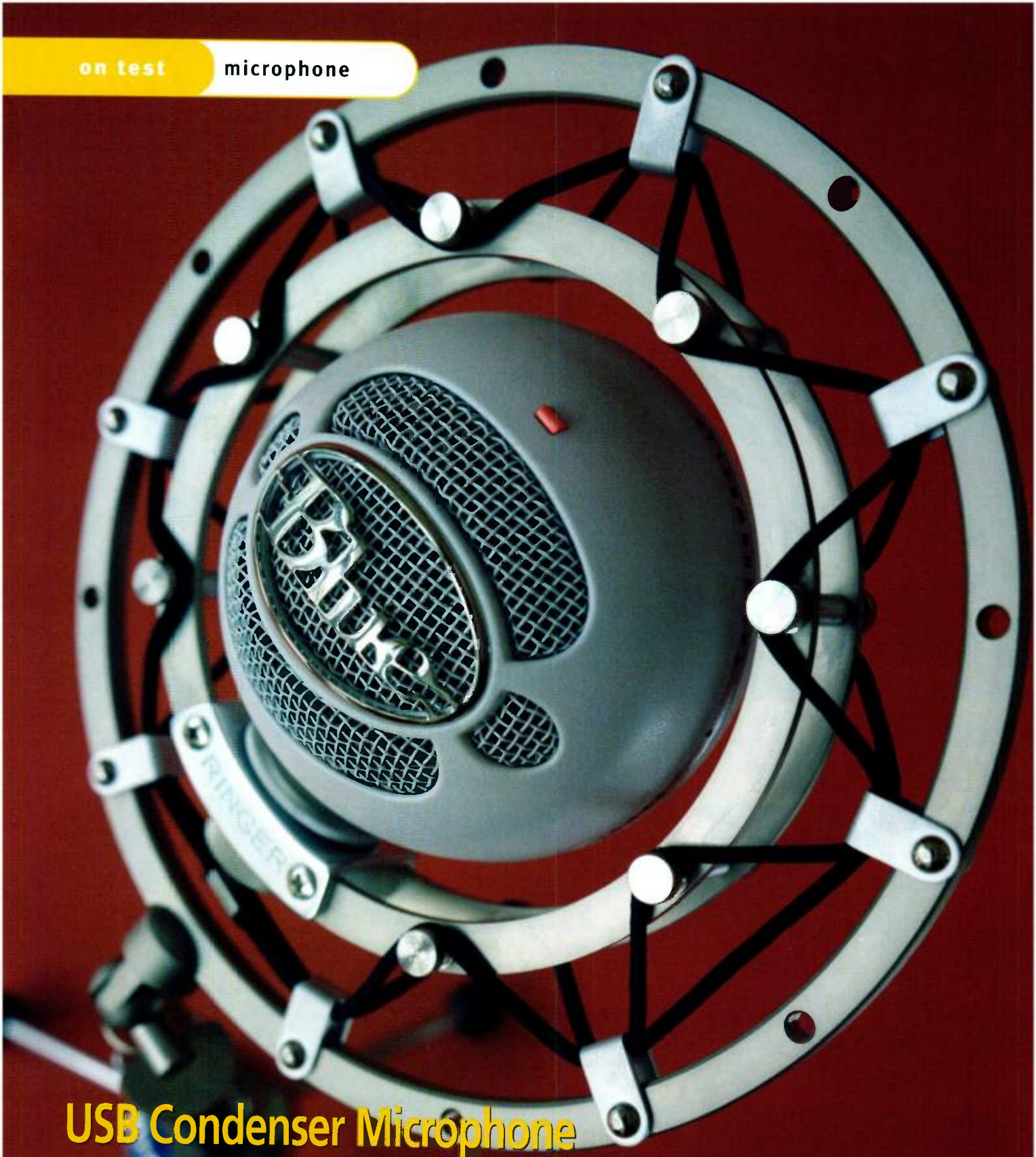

USB Condenser Microphone

Blue Snowball

Paul White

A multi-pattern mic that plugs straight into a computer.

One notable feature of this year's Winter NAMM show was the number of different companies introducing the world's 'first serious USB microphone'. Blue's USB offering comes in their now-familiar Ball format — blue, red and black varieties (the Ball, Kickball and 8-ball respectively) are already available — and because the newcomer is white, the name Snowball seems totally appropriate.

As with the other ball mics in the range, the Snowball has a resin casing with wire-mesh grilles at front and rear. A threaded

stand-adaptor is set into the base and a neat table-top tripod stand comes with the microphone, along with a USB cable for connection to a computer. (The Ringer shockmount pictured here is an optional accessory, costing £99.99 from Turnkey +44 020 7419 9999/www.turnkey.co.uk.) Besides a USB port, the mic's only means of connection, there's a three-position switch at the rear which is used to switch between two cardioid-pattern modes (one with a -10dB pad and one without) and an omnidirectional mode. Overall, the frequency response extends from 40Hz to 18kHz (at -6dB, judging from the response curves given on the Blue web site), with a presence peak at 3kHz in cardioid mode and at around 10kHz in omni mode. A red light on the mic shows that it is active; all necessary power comes from the computer's USB buss.

The converters built into the microphone, which has a capacitor capsule, are fixed at 16-bit/44.1kHz and the mic can be used with computers running Windows XP or Mac OS X without the need to install additional drivers. Inside the host software, the mic shows up as two identical input sources rather than as a single mono source.

According to Blue, the Snowball makes a good general-purpose vocal or instrument mic for project studio recording, but it's also clear that they — like all the other USB mic manufacturers — have their eye on the growing Podcasting market. As all the USB mics I've seen so far have been 16-bit, I can only assume that as yet there are no suitable USB-powered chips that produce a 24-bit output. The quality that can be obtained from 16-bit is excellent, provided you can run at or near maximum level, to make use of all the available dynamic range, but where you use only a small part of that dynamic range, the resolution can suffer. When using a conventional mic preamp, you have an analogue gain control that comes prior to the converters on your audio interface or soundcard, so you can optimise your recording levels. In the case of the Snowball, you only have the choice of pad or no pad.

Testing, Testing...

I set up the mic as part of an Aggregate Device under Mac OS X, so that I could use the computer's built-in audio to monitor the output. When using the Snowball to record conventional studio vocals relatively close to the microphone, the signal level produced was fine, but when it was placed in the middle of a table about 18 inches away to record plain speech (as for a Podcast, for example), the recorded level ended up at around -30dB peak. This equates to a resolution of around 11 bits, or a dynamic range in the order of 66dB. Doing the plain speech test at a distance of 18 inches produced an almost invisibly small amplitude waveform in *Logic*, so I normalised it to see how the sound quality would hold up. The subjective speech quality was actually very good and would be fine for most casual Podcasting or video applications, although there was a certain amount of audible background hiss, which is to be expected when working at such low levels. If you bring the mic closer, so that you get a more sensible level into the recording system, the quality improves markedly. Having a pad setting is useful when you're recording loud sources such as electric guitar amplifiers, but if the designers are serious about the Podcasting market, a +10dB boost switch would also be a good idea.

I was intrigued by the Snowball's omni mode. Typically, omnidirectional mics have an acoustically transparent mesh covering the capsule, but this mic is totally enclosed at the sides and open only at the front and rear. I felt this must compromise the omni mode's performance to some extent. When I checked the polar patterns on the Blue web site, my suspicions were confirmed — the omnidirectional pattern behaves more like a wide cardioid at most frequencies and the sound picked up at the rear is much duller than the sound picked up at the front. The different tonality of the omni mode was also to be expected, as its presence peak is much higher than that of the cardioid modes. When te mic is used on-axis, the omni setting produces a nice airy sound which makes the cardioid modes sound slightly dull by comparison — almost like listening to a typical dynamic mic next to a capacitor mic.

Cool Or What?

There's no denying that a buss-powered USB mic is a very neat solution to recording audio into a laptop or domestic computer system, and this one produces subjectively high-quality results, providing you use it close enough to the source to get a healthy recording level, by which I mean a signal peaking at -15dB or higher. In my view, the omni mode is best thought of as offering a different tonal flavour to the cardioid modes. Its pickup pattern is not truly omnidirectional and I wouldn't consider the mic to be well suited to recording round-table discussions in this mode, but used appropriately its sound quality is fine and the background noise acceptably low. If you are going to use this mic for Podcasting, it will produce the best results if you speak into it from around six inches away, and you can choose omni or cardioid mode depending on which tonality suits your voice best. For loud instruments, the pad is a sensible and welcome addition, but because there is no variable gain control you'll sometimes have to use the mic's distance from the source as a gain control instead, which may mean compromising on placement. I'm sure that we'll see 24-bit USB mics in the future, and hopefully ones with more control over gain prior to the converters, but at the present stage of USB buss-powered technology, where the amount of current you can draw is the limiting factor, the performance of this microphone is probably nearing the limits of what is possible. SOS

Alternatives

Studio-grade USB mics are available or imminent from Samson, Rode, MXL and SE Electronics. To the best of my knowledge, they are all 16-bit/44.1kHz and none offer variable gain control other than the Samson Co1U, reviewed in *SOS* June 2006 (www.soundonsound.com/sos/jun06/articles/samsonco1u.htm). The SE Electronics DO1 is more expensive than both the Snowball and the Samson CO1U, but it can connect via a conventional XLR cable as well as USB and features a two-way audio interface — a headphone output lets you monitor playback from the computer.

SOUND ON SOUND

Blue Snowball £130

pros

- Stylish and simple to use.
- Needs no drivers for Mac OS X or Windows XP.
- Switchable pad and polar pattern.
- Table-top stand included.

cons

- Omni mode is more like a wide cardioid pattern with a high presence peak.
- The lack of gain adjustment (other than a pad) means you need to work close to the mic to get a high enough recording level.

summary

Though subject to certain technical limitations, the Snowball works well for close-up vocals or speech, while the pattern switching system gives a choice of two very usable tonalities. The inclusion of a pad switch means that it may also be used for recording louder sources, such as guitar combos or percussion.

information

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Mike Elizondo

Mike Elizondo has gone from being Dr Dre's right-hand man, co-writing some of the biggest hip-hop hits of recent years, to being an innovative producer in his own right.

Paul Tingen

He might not be a household name, but Mike Elizondo has been involved in the making of an astonishing number of hit recordings. Since his credits first began appearing on official releases 10 years ago he has, as a bassist, keyboardist and guitarist, performed on albums by a strikingly wide range of artists, among them the likes of Eminem, 50 Cent, Jay-Z, Ice Cube, Dr Dre, Snoop Dogg, Xzibit, Macy Gray, Sheryl Crow, Ry Cooder, Avril Lavigne, Ricky Lee Jones, Gwen Stefani and many, many others.

The high proportion of hip-hop artists in this list reflects Elizondo's close partnership with Andre Young, aka Dr Dre. Elizondo worked on the instrumental version of the producer's 1999 solo album *2001*, and the pair ended up co-writing and co-producing. As a team, Elizondo and Dre were the creative forces behind the music for recent mega-hits like 'The Real Slim Shady' and 'Just Lose It' by Eminem, 'In Da Club' by 50 Cent, 'Let Me Blow Your Mind' by Eve featuring Gwen Stefani, 'Family Affair' by Mary J Blige and several more. Since then, Elizondo has begun producing in his own right. Fiona Apple's *Extraordinary Machine* was the first entire album he single-handedly produced, and has attracted universal acclaim, while he also worked on Pink's latest album *I'm Not Dead*.

No Boundaries

Speaking from his home in Los Angeles, where he lives with his wife and three young daughters and has a state-of-the-art recording studio, Elizondo reflects: "My father is a professional musician and an avid record collector, so I grew up with a very broad sense of what kinds of music are available. Everything from P-Funk to Al Green, to Jimi Hendrix, to the Beatles, John Coltrane, everything. I was really fortunate to be able to listen to music without boundaries. In working as a bass player, songwriter and producer I try to take the same approach, which is to have no boundaries and just be open to the different experiences that come up. I don't want to be pigeonholed.

"I began playing piano at age nine, saxophone at age 12, and bass aged 14 at high school because there were no other bass players, so I did not have any competition and could immediately become the best bass player at school! The reason that playing bass stuck was that the role it has, whether you're playing acoustic or electric or keyboard bass, is very appealing to me, because you are the link between the rhythm and the melody. I also enjoy not being the lead singer or lead guitarist, instead remaining in the background, knowing that I have an integral role but that not everyone is watching me."

Similarly, as a songwriter and producer Elizondo plays a fundamental part in the creation of music, without attracting too much attention. For him production and playing bass are complementary activities. "It appears to me that a great number of bass players are able to hear everything in an arrangement, what the drums and the lead vocal and the harmonic instruments are doing, and then wrap themselves around this. Without a doubt this means that you're listening to music in a production sense. I love being around the process of making music, whether it's producing, writing a song, or just playing bass. I get equally excited about these things, which is why I'll continue to do session work. But of course with so many production opportunities now coming my way, I'm very selective about the sessions that I choose to do."

Joining The Firm

Elizondo's career-defining moment occurred when he first joined Dr Dre in 1996, at the age of 24, for the recording of The Firm

(featuring Nas, AZ and Foxy Brown), on the recommendation of an old school friend, Richard 'Segal' Huredia, who was working as an engineer with Dre. Elizondo was at the time also working with big-name producers like T-Bone Burnett, Matthew Wilder, Matt Wallace, Glen Ballard and Steve Lindsey, but it was his work with Dre that took him to another level.

"I came in and played some bass lines on a few songs, and fortunately we hit it off musically," recalls Elizondo. "He kept calling me for more sessions, and this evolved into me participating more with songwriting, beginning with Eminem's 'The Real Slim Shady' [*2000*], and then moving into production. The things I did with the other producers opened up opportunities as well, but Dre gave me my first writing and production credits."

'The Real Slim Shady' was released as a single and as part of Eminem's *The Marshall Mathers LP* (2000). "The album had already been mixed," explains Elizondo, "and we had some time to kill in the studio, and that song arrived in the same way all the others had. Basically, we began with a drumbeat that Dre programmed into an MPC3000. At the time we were working with a keyboard player called Tom Coster Jr, who had this harpsichord-like sound, and I was playing bass, and we simultaneously came up with the parts that evolved into the track. Eminem was in the studio, and when he heard the piece he immediately reacted to it, and had a concept ready to go. I later doubled the electric bass with a keyboard bass that I had programmed on a Nord Lead.

"The MPC is Dr Dre's priority instrument, above anything else. It has become a staple in so many forms of music because it is very musical and intuitive. We have a lot of sound libraries for it at our fingertips. We sometimes use a sample from a record to create a small loop and maybe layer it with a kick or snare, but in general Dre likes to have all parts separated, and so prefers to program everything in the MPC. When you have a loop there's only so much you can change in the balance by adjusting low end or high end. So the drumbeats were usually started from scratch in the MPC.

"The MPC is the session brain from which the MIDI clock is run as well. During the sessions with Dre we set up with maybe two or three musical stations, and my station will have a guitar rig and a bass rig and a soft-synth rig, which is basically a G5 with everything from *Reason* to *Logic* and a whole library of sounds and samples and soft synths. I have Native Instruments libraries and the *ESX* sampler in *Logic*. I also have a loop machine that receives MIDI clock and that allows me to loop a guitar or bass. It might be a four- or eight- or 16-bar idea, and I can punch myself in and out of a particular loop that is synced up to the MPC.

"The main loop machine that I use is called a Repeater, made by a company called Electrix. It's primarily made for DJs to grab bits and pieces of records, have MIDI if they're programming drumbeats, and they can then loop things and have it be in time. You can also plug your bass into it, or a Fender Rhodes, anything, and it allows you to record in time to a MIDI clock. As for the G5, at the time it was the only Apple machine with a processor powerful enough to allow for extensive layering of sounds without crashing. Now that Apple have come out with laptops with the Intel Core processor you can get the power of a G5 in a laptop."

Several other mega-hits in which Elizondo was involved, like 'In Da Club' by 50 Cent and 'Family Affair', by Mary J Blige, came into being according to a similar process, with Dre, a keyboardist and Elizondo working on backing tracks. "On a good day we'll get anything from 10 to 15 ideas in one session," says the bassist. "Rather than separate all the sounds, we make a decent stereo mix of an idea, record it, and move on to the next idea. We used to record to DAT and now record to CD-R, but the aim is the same: to quickly catalogue all the ideas we have. The engineers make notes of all the sounds, so that when the artist comes along and he wants to work on a certain track, we can immediately recall everything.

"Dre and I had written 'In Da Club' for a different artist and it sat around for a few months, until 50 Cent came into town and we played it for him. He immediately gravitated towards it, started writing, and very quickly got the hook and the intro. That track was finished pretty quickly after he started to work on it. 'Family Affair' was again Dre with an MPC3000, myself and a keyboard player throwing out ideas. Mary J Blige added her vocals to that track."

Over The Top

"Once the vocals are laid down, we get into a different mindset, focusing more on the overall arrangements, where you try to make certain sections, like the chorus, come alive. With hip-hop you have four or eight bars repeated over and over again, and the trick is to make that interesting, trying to get an ▶

Mike Elizondo's main instrument is still bass guitar, and he is an in-demand session player.

arrangement where there may be a sound in the second verse that's nowhere else in the song. It may be subliminal, but might give a certain mood or texture. Even though the basic track in hip-hop tends to stay the same throughout, you are embellishing parts, based on the rhythm of the vocals or what is being talked about lyrically.

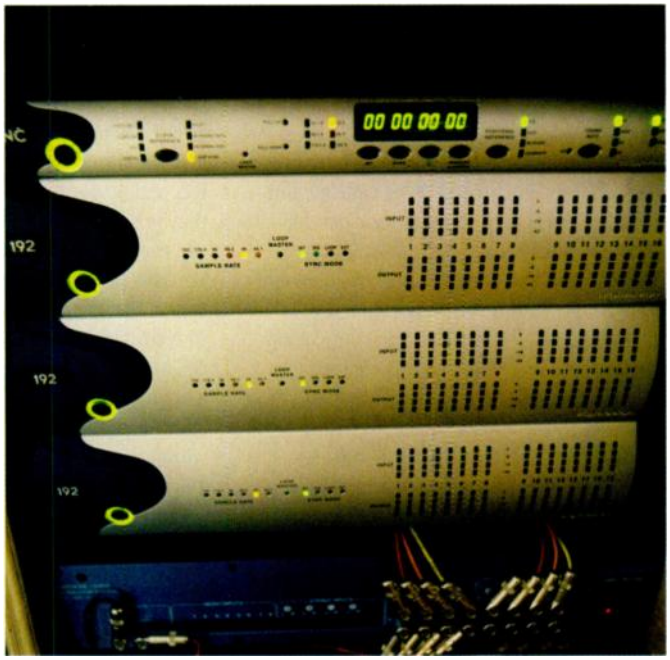
"Embellishing is a matter of trial and error, trying different parts, different keyboard sounds and sections, for instance, on 'In Da Club', an eight-note guitar part in the chorus to make that come alive, and so on. With 'Family Affair' we adjusted the arrangements in relation to her verse, chorus and bridge. I think we added a string part after the second chorus, where she goes into this really swirly vocal thing. We probably also added some Fender Rhodes chords here and there or some string pads, to embellish the mood of what she sings. By contrast, Eminem's 'Just Lose It' didn't need a lot of extra stuff, because his vocals carried most of it.

"Each song has its own challenges. Even though it is the same harmonic progression, you can do things to really lift certain sections, to make it sound as if there is some sort of form to the song, as opposed to just a bland loop without any changes. It's actually very challenging to make a four-bar loop or chord progression sound interesting for three and a half minutes, to make it sound like the chorus is lifting up for instance. A lot of these embellishments happen as we are simultaneously mixing. Dre is very hand-on with the mix, and he'll make suggestions to add parts or bring certain things out."

Going Digital

Surprisingly, it wasn't until Eminem's fourth album, *The Eminem Show* (2002), that Dre and Elizondo made use of digital recording. Elizondo gives the lowdown on why they stuck with analogue for so long. "The first two Eminem records we worked on, *The Slim Shady LP* [*1999*] and *The Marshall Mathers LP* [*2000*], were recorded to two-inch analogue, basically for sonic reasons. Sound quality is really important for Dre, and he's extremely in tune with how things fit in a track. I learned a lot from watching him and the attention to detail he has — the amount of time he can spend on just getting a snare or a bass drum sound. But once Pro Tools stepped up their game in terms of sonic quality with the 192 audio interfaces and HD, we began using that, mainly because of the speed with which you can work.

"With analogue you have to take out the razor blade to do edits, but with Pro Tools you have this instantaneous ability to move things around. When we used analogue we were more or less obliged to play parts from beginning to end, all the way down. Even as you tried to play it like a loop, you still had a human element to it. Now with Pro Tools we have the option of looping things. It depends on the track whether you want that human element to it, or have everything just totally locked. On 'The Real Slim Shady' I played the bass line all the way down, but 60-70 percent of 'In Da Club' was sequenced.



Recording at Phantom Studios is handled by a large Pro Tools rig and an SSL AWS900 controller/mixer.

"By the time we did [*Eminem's 2005 hit*] 'Just Lose It', we had completely embraced the Pro Tools process of working. That song had a lot of sequencing. I think the bass sound came from a Minimoog soft synth made by Arturia, while the main keyboard riff came from a soft synth called *Plastix*, and there was another sample that [*keyboardist*] Mark Batson played, which was a random snippet of a percussion part from a sample library. The song was mainly soft synths, all done in the G5."

Phantom Power

By the time Elizondo worked on 'Just Lose It', he had established an impressive pedigree as a co-writer and co-producer on a range of massive hits. The co-production credits with Dre were in recognition of, said Elizondo, "my involvement in the bulk of the music and having a hand in the arrangements". All this resulted, one imagines, in the Californian having a tidy sum of money at his disposal, and like many musicians in his position, he set up his own studio. Called Phantom Studios, it is based around an SSL AWS900 analogue mixing console/DAW controller and three Pro Tools 192 units.

"The studio has about 1000 square feet of floor space, and there's a live room, a control room and a lounge," explains Elizondo. "Because I have a family, I wanted to have a studio close to home. I also wanted a studio where I would have everything I need, and am able to create on the spot. And I wanted a studio that would be comfortable enough for established artists, yet where I can also

The Human Machine

Perhaps the most crucial piece of kit in Mike Elizondo's arsenal is the son-of-MPC3000, the MPC4000. "The MPC4000 is the machine I learned my sequencing skills on, and it continues to be the quickest way to sequence for me. I've done a lot of sequencing in *Logic* as well, but I know how to operate the 4000 without thinking. So I often use sounds in *Logic*, but sequence them with the MPC. For me, working with a computer is very different than with a workstation. There are certain inaccuracies that allow dedicated machines to feel almost human-like. They are very intuitive, you can get a great feel out of them, and it's easier to manipulate things to feel a certain way. The MPC4000 has a certain feel, just like the old SP1200 drum machine had a certain feel, or the Roland 808.

"One issue big for me are the computer screens. What did we look at while making records not so long ago? Nothing. Now we all look at screens and sometimes I wish we could go back to the days when we just listened with our ears. I have my screens set up in such a way that I can move them away from the mixing monitors. That definitely helps at times. Computers are great tools, but I don't necessarily like looking at them. I have separated my computers into two separate rigs. One computer is dedicated to soft synths and keyboard sounds, and another is set up for Pro Tools. When I'm working in Pro Tools I don't want to be dealing with keyboards, and *vice versa*. Even so, I'm still most comfortable working with the MPC4000. I may abandon it in the next few years and only do things in my computer, but my journey hasn't led me there yet."

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"When I first built the studio I had a Pro Control in here, and soon after the studio was completed SSL announced the AWS workstation, so I had to move things around to have that wired in. Having that console is a huge advantage to me: it's an analogue mixer with SSL EQs and everything, and it also enables me to control Pro Tools. It's great to do everything in Pro Tools and then be able to separate things out across 24 channels for mixing.

"I like to embrace both the analogue and the digital worlds. I like to juxtapose some of the digital harshness with analogue warmth. I have lots of different analogue boxes, like Urei 1073, API 5050 and 5060, Avalon, Universal Audio, Urei 1176, Neve 2254, Manley line mixer, and so on. It's a combination of older and newer gear. I also have a lot of microphones, like the Telefunken M250 that's great for vocals, the Royer 122, which I love, and I'm a big fan of the Blue Bottle, which is probably the most versatile microphone I have. Unbelievable. I also have a couple of C12s, U67, U47, Sennheiser microphones, SM57s and so on."

While the AWS plays a crucial role in integrating this wealth of analogue kit with Elizondo's digital system, he's far from a retro man. "I'm plugged in like crazy," he says. "I have every plug-ever in made, everything from Waves stuff to Digidesign stuff. I recently received the Waves SSL plug-in, which gives you an amazingly accurate compressor and EQ based on the signal path of what I think was a G-series SSL strip.

"I'm also a big fan of soft synths. There's nothing that beats the real thing, like a Fender Rhodes or a Wurlitzer, but in the context of a track you're not going to pick up all the missing nuances between reality and a soft synth. Not everybody can afford every synthesizer in the world, let alone have space for them. At the same time I have picked up a number of keyboards over the years, like the Fender Rhodes, a Wurlitzer, an old Clavinet, Minimoog, Prophet 5, Roland Jupiter 8, Oberheim OB8 and Juno 106. I love the original keyboards and fortunately have the space and means to put them here."

A Second Bite Of The Apple

Elizondo has had his studio up and running since the autumn of 2004, and immediately put it to use writing, recording and producing songs with Pink for her *I'm Not Dead* CD (from their sessions, only the track 'I Got Money Now' is included on the album), and most notably, as player, arranger and main producer on Fiona Apple's

Some of the rackmount gear at Phantom Studios. Left, from top: Manley line mixer, MOTU MIDI Timepiece synchroniser (x2), MOTU 828 interface (x2). Right, from top: Alesis Masterlink CD master recorder, TC Electronic Fireworx effects, Yamaha SPX200 effects, API parametric and graphic EQs, Tube-Tech CL1B compressor and Teletronix LA2A compressor.

Extraordinary Machine — an album which had a tortuous gestation period. Despite the fact that Apple's first two albums had gone platinum in the US, Sony/BMG refused to release her third album when she delivered it in the beginning of 2003. Made with arranger/producer Jon Brion, the label judged it too arty and abstract. Furious fans began a campaign for its release, and eventually Apple and her record company started talking again. The singer was perhaps moved by the fact that when tracks from the album were leaked on the Internet, many agreed with the record company that the carnivalesque orchestral arrangements were laboured and indulgent. Apple agreed to re-record her songs with another producer, and in the beginning of 2004 hooked herself up with Elizondo.

"I was presented with the material that

The live room at Phantom Studios.

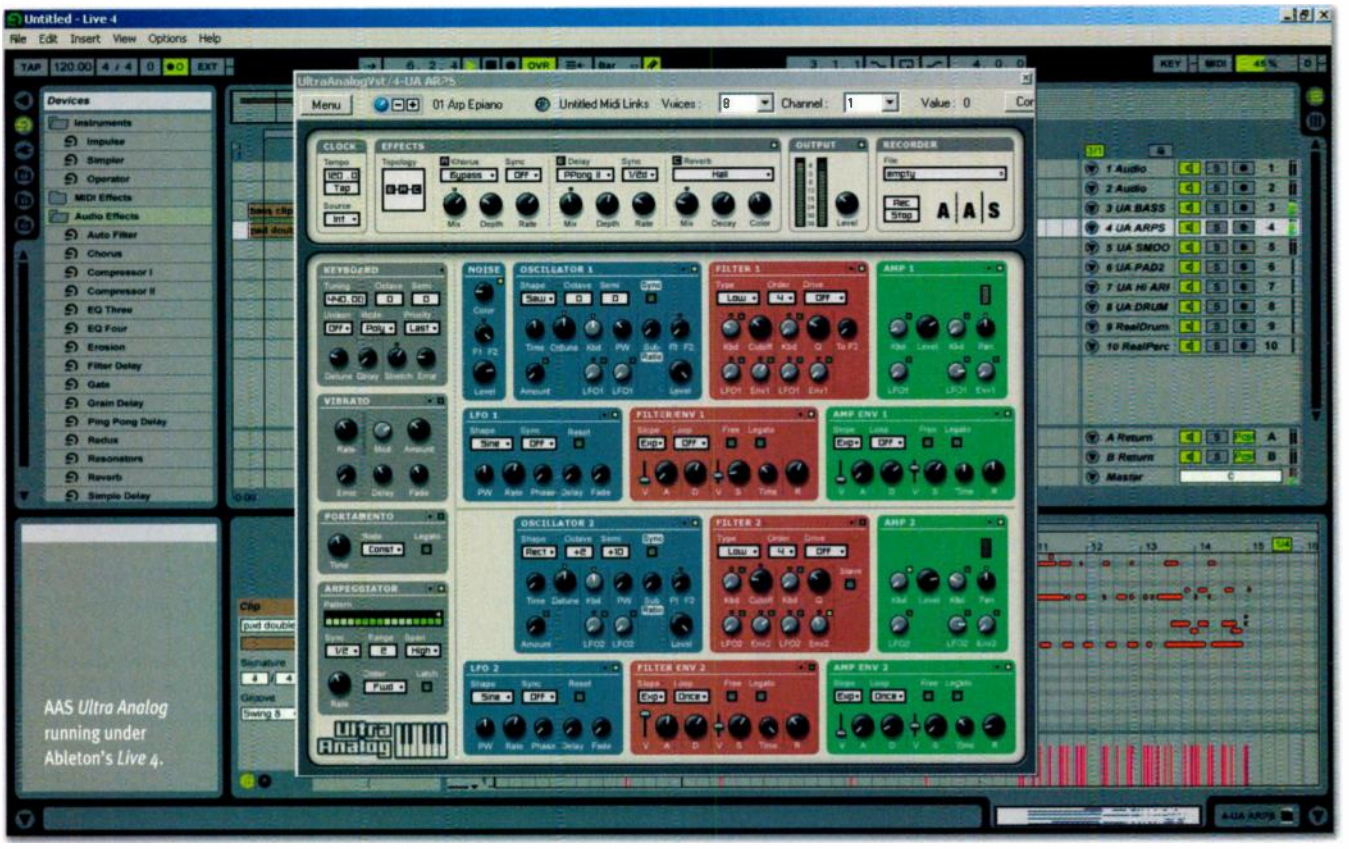
Fiona had worked on with Jon Brion, and for one reason or another she wanted to hear different interpretations of them," recalls the bassist/producer. "I used his versions to learn the material, sitting down at the piano to work out the chord progressions and get a sense of what Fiona was doing vocally. I didn't use or borrow anything from what Jon had done. Not to say that Jon's versions were demos, but I get presented with artist's demos to work from as a springboard a lot of the time, and working like this was no different. I would do two or three songs at a time, and then presented them to her."

Elizondo declares himself a big fan of Mitchell Froom, and his production on the Fiona Apple album certainly recalls Froom. Strange sounds come in and out all the time, without distracting from the songs. "The album was actually recorded before my studio was finished," explains Elizondo. "So it was done in a bedroom here in the house while my studio was being built, with Pro Tools and a small little controller, and some Neve modules.

Some of Elizondo's many vintage keyboards: from top, Moog Opus 3 string machine, Moog Prodigy synth and Wurlitzer EP200 electric piano.

"I would lay down the basic tracks and then I brought in other musicians to embellish things. We used a combination of soft synths and real keyboards for the keyboard sounds. There were some Mellotron and Chamberlin sounds from a library, and in other cases we used the real thing. Sometimes we went to *Gigastudio* for some cool string sounds or some harp sounds, thing like that. To be honest, we used whatever was available to us on the day. Those weird sounds at the beginning and end of 'Get Him Back' for instance are from the library of Keefus Ciancia, one of the keyboardists. On the track 'Timps' I programmed a rhythm on the MPC4000 that I thought would sound cool with Fiona's vocal, and then we layered that with real marimbas, vibes, piano, a whole load of keyboards, and I played Minimoog bass on it. I felt that Fiona's piano parts called for something ethereal.

"We tried to create something unique for each song. I like to push the envelope and come up with interesting sounds and arrangements, while at the same time serving the artist's vision." At this, he's been more successful than most. SOS



AAS Ultra Analog

Modelled Analogue Synth For Mac & PC

Like the analogue modelling of AAS's *Tassman* but don't need its complexity or fancy its price much? The much more affordable *Ultra Analog* could be right up your street...

Derek Johnson

Once upon a time, the idea of IRCAM — the Paris-based music and acoustic research establishment — producing anything populist, commercial or, in some cases, comprehensible seemed faintly ludicrous. Not that their work wasn't fascinating, but in recent years, the computing power available to the rest of us has caught up somewhat with those at the cutting edge. The result is that the likes of IRCAM *do* release commercial products and of course, those who have passed through their doors have often gone on to more commercially minded concerns.

Such is the case with Marc-Pierre Verge and Philippe Dérogis, the names behind Montreal's Applied Acoustics Systems. Both are musicians with a heavy background in physics, mathematics and acoustics. AAS's products, however, are *not* heavy, but put a friendly gloss on some serious coding.

A lot of the work done at IRCAM would now be classed, broadly, as physical modelling: simulating real acoustic events with computers. Indeed, AAS's first product, *Tassman*, is a powerful physical modelling synth. Various incarnations of *Tassman* have been reviewed in *SOS* (see the July 2000, June 2003, and November 2004 issues). Since the release of *Tassman*, AAS have moved sideways into modelling classic electric pianos with *Lounge Lizard* and taken a stab at replicating string-based instruments with *String Studio*. Their latest product sees them returning to their synth roots.

Ultra Analog costs less than *Tassman*, but

SOUND ON SOUND

AAS *Ultra Analog* £150

pros
- Good value.
- Great sound.
- Easy to use.
- Deeper than it looks.

cons
- Some of the on-screen graphics are a bit small.
- Reverb not to my taste.

summary
Given the price and target market, I've no serious complaints with *Ultra Analog*, as you can see from my cons above! If you only have money for one serious commercial synth plug-in, make it this one.

The browser window, which can be displayed to the synth's left.

it's also rather simpler to use. Where *Tassman* is a fully-modular creative environment, *UA* is a 'boot-and-play' affair, though no less inspiring in the right hands. Unlike many software synths of today, *Ultra Analog* wasn't modelled on any particular hardware synth, but if you think of any two-oscillator analogue synth, you'll get the basic idea. *UA* is not monophonic like most two-oscillator analogue synths (in fact, it's up to 32-voice polyphonic) but it can be *made* monophonic if you want to be purist. And though not fully modular, it has modular elements.

Look Out

UA's look is clean and approachable, with the various synth elements grouped into logical, colour-coded sections. Oscillators, including LFOs, are blue. Filters, of which there are two, are red; the same goes for the filter envelope generators. All the modules related to amplitude (level and audio output), including amp EGs, are green. There's a greyness to the rest of the front panel, if not the facilities: portamento, overall vibrato and tuning are joined by a great arpeggiator and a handful of effects processors.

There's only one window, though this can be expanded to include instant access to the patch library (see the screenshot on the left). Your own work can easily be saved and organised from this expanded window.

In common with the rest of AAS's family, *Ultra Analog* is cross-platform, running under Mac OS 10.2 or higher (on a Mac with a minimum 733MHz processor) or Windows 98SE, 2000, ME or XP (on a PC with a minimum 800MHz processor). There is no compatibility with Intel-based Macs yet, but as of this review AAS had announced that development was under way. Most plug-in and audio standards are covered, such as ASIO, MME, VST, ASIO, AU and RTAS, and the software will even run as a stand-alone synth, which is great if you just want to do some virtual tweaking without the distraction of your audio host.

You might be thinking that *Ultra Analog* is in some way related to *Tassman*. While there are sure to be features in common, *UA* has apparently been designed completely from scratch. It's a different beast to the more expensive synth and you won't, for example, be able to load patches from one into the other.

Oscillators & Filters

First, let's have a look at the basic signal path — old hands will know what to expect. A quick glance reveals two more or less identical sets of main sonic modules, each set offering oscillator, filter, LFO and a pair of EGs. The PDF manual is good at explaining all this, although some newcomers may find it a little obtuse. It covers technical details in a comprehensible way, but I say that as an experienced synthesist.

The oscillator starts well, offering sawtooth, rectangular, sine and white noise waveforms plus an octave-below sub-oscillator. AAS have obviously had a lot of experience of different oscillator types; to me theirs is more ARP than Moog, though the warmth and fatness of the latter are present (not to mention the metallic potential of a Korg MS20). With a rectangular wave selected, a pulse-width control becomes available, and this can be modulated by an LFO for the classic effect. I wonder why AAS didn't go the extra mile and offer pulse-width control over other waveforms? Tuning control, however, is comprehensive: there's course and fine tuning, and oscillator detuning options. Keyboard tracking is completely controllable — equal temperament is at 12 o'clock — and this parameter is also modulatable by an LFO. A sort of glide effect offered by the (unlabelled) ramp generator is ▶

The Arpeggiator

Arpeggiators, it seems, can be found in almost all synth-based software except for Propellerhead's *Reason*. AAS have designed a nice example (in the bottom left corner of the *UA* window) which offers a little more than the most basic models. It lets you break up held chords in a rhythmic fashion, but with control over the basic rhythmic pattern via a 16-step grid. There are some other simple controls: transpose range, span (which determines whether the transposition goes up, down or over the whole note range) and note order, and you have the option of sync'ing to MIDI or internal clock as required, but that's about it. I'm pleased to see a 'latch' option, so arps keep going once your hands leave the keyboard, but on my AAS request list is a 'transpose' option, so you can move the pattern up or down with one finger — I miss that from classic hardware such as Roland's SH101.

made using Free Run and Legato switches. The former bypasses the sustain portion and loops the release stage automatically. Legato lets you decide how newly triggered notes are treated when an envelope has already been triggered, and looping options allow you to control which stages loop while a key is held.

Look Beyond

That's not all there is to *UA*, however. As I've mentioned, the oscillators (and the noise generator) can be routed to either filter. Having this option means that a single-oscillator patch can have rather more depth than you might think: the two filters can have radically different characteristics and modulation routing, and the amplifiers may be panned left and right. The synth certainly seems more modular than it actually is, and a few minutes tweaking can soon produce quite complex soundscapes that sound as if they're coming from something much more complicated and expensive. And that's before we add the effects...

On a more mundane level, it's worth noting that most individual modules, and many of the parameters within the modules, can be enabled or disabled at will. This has the dual advantage of letting you be very precise about programming and allowing you to remove unused elements to save on your computer's DSP power. Furthermore, settings from like modules can be copied from one to the other (as shown in the screenshot on the previous page), or reinitialised.

All that remains, from a strictly voice-creation point of view, are the Global modules. The one labelled Keyboard governs overall tuning, and mono/poly mode, with a Unison option thrown in. The Stretch and Error parameters add controlled or random tuning variations, emulating something genuinely old and in need of a service, although stretch tuning, of course, is a term from the world of piano tuning. As if two LFOs weren't enough, the Global Vibrato module adds a little extra to a patch, and delay and fade options help make the effect more natural. Finally, portamento, or glide, also gets a Global module to itself.

Effects

Effects aren't really a necessary addition to *Ultra Analog*'s sonic arsenal — most of us will be working with a software host that offers plenty in this line, and freebies abound on the Internet. But here they are, and AAS have kept it simple so that the effects don't get in the way of the core business. Modulation, delay and reverb modules are provided, each with a handful of significant parameters. Four choices of parallel and linear routing include two options that treat the outputs of the two rows of synth 'modules' with a different effect each — a nice touch.

The Chorus module offers chorus, flanging and vibrato with stereo options, and these effects can be sync'ed to tempo. Needless to say, there are Rate, Depth and Wet/Dry mix controls. The delay effect is also sync'able, and the options here mix realistically retro with precisely modern in effects labelled 'ping pong', 'slap back' and 'tape', which should give you an idea of what's available. Finally, the reverb module will not replace anything your host software can provide, but approach it as part of *Ultra Analog*'s editing facilities and you won't be disappointed. It's a little ringy and crunchy, but can help add air and space to a heavily delayed or arpeggiated patch. Algorithms range from small room to various sizes of hall.

Look Now

Working with *Ultra Analog* is a doddle. The interface is really approachable, and great sounds just pour out of the synth as you get to grips with it. I'd barely auditioned the factory presets before I was creating folders full of my own.

On the subject of the factory set, it's hard to summarise the huge collection. Even AAS must have seen the trouble coming, and have created a hard-to-miss folder of patches called '!!Guided Tour!!' which gives you a taster of thunderous basses, arpeggiations that will seem impossible until you've played with the synth, full-frontal leads and some of the warmest, huggiest pads this side of an ARP Quartet. And any 'voice'-themed patch will have you whizzing over to the 'formant' filter options for some experimentation of your own.

UA appears to be a fairly efficient plug-in that doesn't make huge demands on its host. But a really busy patch with lots of delay repeats and a long reverb can get a bit much, and you'll be hard pressed to get multiple instances to work (*UA* is definitely not multitimbral!). If I had to nitpick, I'd have a go at the size of some of the on-screen buttons — those that disable modules and parameters are particularly small. And the stereo output level meters (to the right of the effects) can be really faint and hard to read. Some other elements — the 'LEDs' which indicate the normal or central position of some knobs, for example — are also really faint.

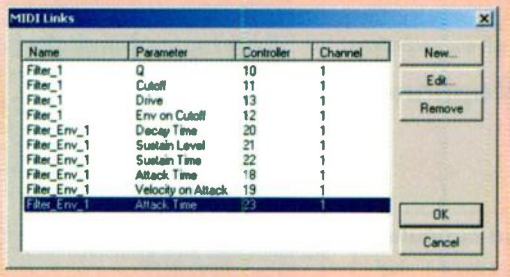
Ultra Analog's main selling point has to *be* its selling point; for the money, this is a lot of synth. There's stuff missing when compared to more expensive software, but it's nothing that you'd notice while you're having such a ball making new sounds (if you're feeling a bit flush, though, check out the AAS modelling bundle: *all* their current plug-ins are included, offering an insane saving on the individual packages).

In short, I heartily recommend AAS's *Ultra Analog* to anyone who likes working with computer-based synths. The simple interface hides the heart of an analogue beast — from throbbing bass to supersonic leads, it's all here. SOS

MIDI Control

It would definitely be remiss of AAS if they hadn't provided a way to access on-screen parameters via MIDI controllers. But they have, and it's simple and powerful. Links to outside controllers can be made by holding down the Control key and clicking on a parameter (on the Mac version — PC users click with the right mouse button as usual) and sending the desired controller data. Nearly everything on screen can be so addressed. The parameter learns the link, though the user can edit this as required and customise how the parameter responds to any incoming data.

What's best is that a screen full of MIDI links can be saved as easily as if it were a normal patch. This works because *UA*'s on-screen elements are always in the same place — so if you have a way of controlling *Ultra Analog* with a particular hardware device, you can import that MIDI map into the plug-in for use with another patch at another time.



MIDI controller links to on-screen parameters can be made manually in this window.

Test Spec

PC REVIEW SYSTEM
- 3.06GHz Pentium 4 PC with 1.25GB of RAM running Windows XP.
- Ableton *Live* v4.
- Mackie *Tracktion* v1.5.
- *Ultra Analog* version reviewed: v1.02.

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Studio SOS

The team create portable acoustic treatment for a composer who has plans to move house.

Rack and stack: with 10 traps to build and hang in one day, there's no time to lose.

Paul White & Hugh Robjohns

This month's Studio SOS features the studio of Hilgrove Kenrick, a professional TV and film composer who recently wrote a series of articles in *Sound On Sound* about producing music for picture (see *SOS* December 2005-March 2006). While writing the series, Hilgrove decided to move his studio from a room in his house into a new double-garage construction and asked Hugh and I if we'd take a look to advise on acoustic treatment. The twist was that he ideally needed a portable approach, as there was a chance he'd be moving house within the next year or two. We came up with an idea for some wall-mounted hanging panels and thought that this rather construction-intensive project would make an interesting Studio SOS — which is why mid-December saw us at Hilgrove's home with a car full of electric saws, nail-guns and cordless screwdrivers. Hilgrove had already bought in all the raw materials we'd specified for the job, other than the foam, which was generously provided by Auralex, via their UK distributor Audio Agency.

The Task Ahead

Hilgrove's studio space is large enough to make most people envious — roughly 23 x 21 feet, and 10 feet high — but other than 10mm Noisestop Solutions rubber matting (the type made from recycled car tyres) that Hilgrove had fitted to the floor, all we had to work with was four painted plaster walls. At the back of the studio are two wooden garage doors separated by a masonry pillar. To save doing anything too permanent there, we suggested that Hilgrove buy a couple of large, thick duvets to hang in the garage-door recesses. He could later hang some thick curtains across the entire back wall, to improve its appearance, if necessary. If it turned out that more bass absorption was required, Hilgrove could easily hang barrier mat directly behind the duvets, to form a limp mass-absorber (providing it was allowed to hang freely). This damped wall would be useful for recording vocals, as the singer could stand with their back to it. Barrier mat is quite expensive to buy in quantity and the shipping cost can add another 50 percent to the bill, but at least it can be taken down and re-used in a future studio. We decided to leave this part of the job for later, as it didn't involve any serious DIY work, to which Hilgrove has a self-confessed allergy!

DIY allergy nothwithstanding, Hilgrove had already sealed the spaces around the

doors, and the walls of the garage are triple-skinned. As his house is reasonably secluded, soundproofing isn't a major concern. Hilgrove has a background in IT, so he's put all the noisy computing gear in a separate machine room with backup UPS power supplies and hooked it up to his keyboard, monitors and Sony digital mixer via cable extenders and MADI looms (running to RME interfaces), which makes his control room quiet enough for serious recording work.

Our plan for the day was to build 10 sound-absorbing traps to hang on the front and side walls, then to fit two further pieces of acoustic foam to the ceiling, to kill reflections from above the mixing position. We could simply have used foam panels on the walls, but their low-end absorption is only worthwhile if they are very thick or spaced from the wall. They can also be difficult to fix up temporarily in a neat and professional way, although Auralex do have some Velcro fixings that work adequately on smooth, painted walls. As Hilgrove was also concerned about aesthetics, we decided to build something more substantial, based on a 125mm-deep wooden frame with 600 x 1200mm internal dimensions. This would exactly accommodate a piece of rigid rockwool cavity-wall insulation slab (30mm thick in this case, but deeper slabs would have been even better, had they been locally available), on top of which would be fixed a sheet of Auralex wedge-profile foam.

The idea was that the rockwool/foam layers would be at the front of the frame, leaving a decent air-gap behind to help improve the low-end absorption. It also left space to add a sheet of freely-hanging barrier mat behind the rockwool, which we fitted to the two traps that we angled across the front corners of the room. As the type of rockwool slab we used is more dense than foam, it provides better absorption than using foam alone and it is also relatively cheap, which is another reason for taking this approach. The foam panel on the front would do its share of absorbing, as usual, and give a tidy, professional appearance to the finished units. Slotted wall-hanging plates fixed to the back would make each trap easy to fix to the wall using just a couple of screws each, and would also allow the panels to be lifted down without the need to undo the screws.

Making The Traps

When we first encoountered the room almost empty, the reverberation was impressive — but certainly not appropriate in a control room! To make matters worse, all three room dimensions are quite closely related, so prominent standing waves would be a potential problem, although we knew that the wooden garage doors should let a lot of low-end pass directly through to the outside world, which would help. As mentioned earlier, sound leakage is not an issue, fortunately, as there are no close neighbours.

The 10 traps we made helped enormously to tame the reverb and seemed to improve the bottom end to a useful extent, but from the outset we realised that a large area of absorption (duvets!) on the back wall would be a vital component, as the wall was all bare wood and plaster. It might also prove necessary to extend the front corner traps and possibly add more corner trapping on the wall/ceiling boundaries. Real Traps Mini Traps, for example, could be a practical way to trap the wall/ceiling corners, as they are easy to put up and take down if you move house. Essentially you can never have too much bass trapping, and it is easy to add more in stages until the optimum balance has been

found. Hilgrove could add these components later as necessary: our intention on that day was to make the room usable for composing and mixing.

To make the frames for our traps, we cut two long sides and two short sides from planks of planed pine timber (using 15mm thick, 125mm wide, 2.4m pine planks), then glued and screwed the corners. No serious cabinet-making skills are required here. You just have to remember that it is the inside dimensions that are important, as the rockwool and foam fit inside the frame. To secure the rockwool slab, we then fitted 20mm wooden batten around the inside of the frame, set back from the front edge by 35mm, so that when the 30mm rockwool was fitted it would sit 5mm behind the front of the frame. This meant that when the Auralex foam was glued to the front of the rockwool, it would be recessed into the frame very slightly, leaving the edges looking clean and tidy.

Having 90mm of air-space behind the rockwool makes a significant improvement to the low-frequency absorption of the trap, and while the wooden frames of the trap are acoustically reflective, we felt they could add a little useful scattering. Traps of this type can be even more effective if you cut holes in the edges to let sound into the back more easily, but as we had to build 10 in a day, we didn't have the time!

We used a couple of wood offcuts to make up some 35mm spacers, upon which we could rest the internal battens as we glued and pinned them in place. This ensured that we got the battens in the right place and avoided the need for measuring every time we fitted a new length. As we had a relatively short space of time in which to finish the construction, we needed to set up an efficient production line, with me sawing and nailing, Hugh drilling and screwdriving and Hilgrove making endless coffee, accompanied by clinically dangerous quantities of chocolate Hob Nobs. Hilgrove also played a few Pink Floyd albums over his studio monitors to keep us motivated.

Hilgrove decided to leave the pine frames unfinished, so we just gave them a light sanding prior to fitting the rockwool and foam. However, I have since built similar structures for my own studio and found a suitable water-based satin-finish varnish (from Wickes) that is very easy to apply and also dries quickly.

We found that the simplest way to fix the rockwool into place was to apply a solvent-free 'Look, there aren't any nails, honest guv' type of adhesive via mastic gun to the frontmost edge of the batten, then simply push the rockwool into place. (As explained earlier, the frame size was decided to ensure that the rockwool slab was an 'interference fit', so the glue didn't have to do much anyway.) Auralex's own spray adhesive was used to fix the foam to the face of the rockwool and we applied it to both surfaces prior to bringing them together. It stuck very positively and also looked very neat, but we had to tuck the edges into the frame as we worked, as the foam panel was just slightly larger than the rockwool slab. If you do get the spray adhesive on a visible section of foam or other surface, we discovered that you can usually remove it by dabbing at it with the sticky side of a piece of gaffa tape — as long as you notice it before the glue sets.

Fitting The Traps

Hilgrove had power trunking running along the front and one side wall of his studio, so we simply stood the traps on top of this, marked the wall through the keyhole slot in the hanging plates, then drilled the necessary holes using a masonry bit and a hammer drill. Plastic plugs were hammered into the holes to take the screws, in the usual way. Two of the traps were fitted across the front corners of the room to give more low-end absorption, so we used scraps of wood cut at 45 degrees to fix the keyhole plates at the correct angle. Filling the space behind these corner traps with more loose rockwool would improve their low-end absorption further, but we settled on fitting a piece of barrier mat behind each, as described earlier, to act as limp-membrane absorbers, as these are effective at low frequencies. In a room of this size, such a small area of absorber could only be of limited help, but it certainly wouldn't do any harm.

We hung three of the absorbers on each side wall, two in the front corners and two behind the monitors. The two foam panels we had left over were fixed to the ceiling,

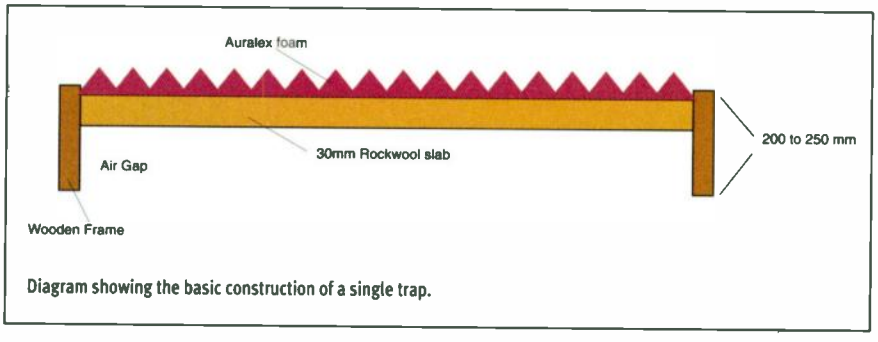


Diagram showing the basic construction of a single trap.

Paul makes a start cutting pine planks for the frames of the traps.

Hugh glues and screws the corners of the frames.

Wooden batten is fitted to the inside of the frames to help position the rockwool and foam correctly.

using a light dash of spray adhesive. We hope it will be possible to remove them at a later date without damaging them beyond the point where they can be re-used.

From Hilgrove's listening position, we had all the 'mirror points' covered, but the amount of bass trapping required was still undetermined. However, once the panels had been hung, the acoustics of the room dried up very noticeably and the stereo imaging improved significantly.

Studio SOS: The Return

A few weeks later, we returned to Hilgrove's room, to check on progress and to hang two thick, 13.5-tog, hollow-fibre duvets in the garage-door recesses. (While duck-down is good for sleeping under, it tends to settle to the bottom of the duvet, which is why we went for hollow-fibre duvets, which keep their shape.) Standard kingsize duvets fitted perfectly into the recesses, and we tacked them in place in the door frames using large-headed nails. The duvets dried up the sound to a useful degree.

With the 10 traps, the ceiling foam panels and the two duvets the room was surprisingly well behaved, and although more bass trapping would undoubtedly be a good thing, the garage door, room doors and windows contributed to the natural bass trapping of the structure, so there were no really hot or dead bass notes evident.

As an experiment, Hugh and I held up a large rug at the back of the room, to see if it would function as a limp-membrane absorber to remove more bass energy. It was surprising how strongly the bass vibrations in the material could be felt, meaning that the rug was indeed drawing energy from the sound field. The bass was slightly but audibly tightened up when we held the rug in place, so we suggested to Hilgrove that he might want to try hanging a couple of rubber-backed rugs in the garage door recesses to provide further bass trapping, and to look nicer than the bare duvets — although I gather Spiderman duvet covers were being considered...

Hilgrove's studio space before the team started work...

... and looking and sounding rather more inviting afterwards.

To avoid reflecting mid-range energy back into the room, the rugs should really be hung behind the duvets, but given the distance to the rear walls it probably ▶

Fitting the rockwool slabs into the frames.

The foam panels go on top of the rockwool, secured with spray adhesive.

One finished trap!

The picture-hanging plates used to mount the traps on the walls, so that they can easily be lifted down.

Two traps in place on the wall, one of them across a corner.

wouldn't make much of a subjective difference if the rugs were in front, and they'd certainly look nicer than exposed duvet. Our impromptu test with the rug didn't seem to affect the mid-range tonality at the monitoring position.

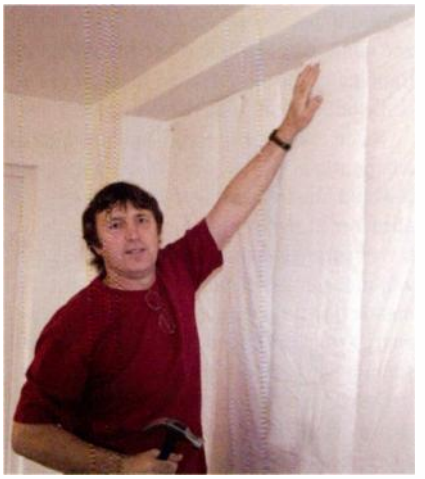
Hilgrove had brought in a soft sofa, which he positioned at the back of the room. This is always a good idea, as large soft furnishings (and the clients who sit on them!) provide a little extra free trapping.

After a final trial of the room, using the theme music from *Batman Begins* (which has monstrous bass end) as a test source, we were generally very happy with how much of an improvement we'd brought about for so little outlay. However, there were still some little jobs for Hilgrove to do to optimise his studio, so we left him with a 'to do' list.

We had noticed that the cavity under the closed-back monitor shelf of Hilgrove's work desk seemed to be acting as a resonator, which had the effect of reducing definition towards the low end of the spectrum, so we suggested relocating his Mackie HR824 speakers onto suitably high rigid stands behind the desk, then either filling the offending cavity with foam or other absorbing material, or otherwise removing the shelf completely. As the monitor shelf was directly behind Hilgrove's computer screens, the space below it couldn't be used as practical storage space anyway.

Taking the speakers off the desk would almost certainly tighten up the low-end performance of the HR824s. I had already reset the switches on the back of the monitors to full bass extension and half-space room loading, which seemed to work best for their current position and should be even more appropriate when the speakers are moved to stands a little closer to the wall.

The wall behind the monitors has two windows in it that Hilgrove didn't want to block, so slatted vertical blinds should ideally be fitted to these, to help deflect and scatter reflections. Setting the blinds half-open usually works well. There were also two noticeably untreated side sections of wall left just behind the frontmost side-wall absorption panels, due to the presence of a door in one wall. This door leads to Hilgrove's machine room, which houses the impressive rack of computers that he needs for his extensive use of *Gigasampler*, running *Vienna Symphonic Library* and other libraries, for classical compositions. We suggested that Hilgrove buy a couple of panels of matching Auralex foam and fix these directly to the door and to the facing wall, to maintain symmetry. This would improve the sound, particularly the stereo imaging, further back in the room where his clients tend to sit. SOS



A return visit enabled Hugh and Paul to hang duvets in the recesses of the wooden garage doors, to further improve the room's acoustics.

Hilgrove's Comments

"When I left the big smoke for a life of creative peace and quiet in the countryside, I hadn't realised I'd end up within 20 miles of the gods of the Studio SOS, Hugh and Paul! I was also lucky enough to have Peter and the team at Total Audio Solutions close by, to advise on MADI and remote extenders, so that I could pile all my noisy PCs, network gear and I/O in the server room, leaving my studio free of extraneous noise.

"The studio room ended up very nearly square — a big no-no — but the builders managed to 'find' another inch or two of width at the last minute. The room previously sounded like an oversized bathroom, but now that the acoustic traps are in place the listening experience is very nearly perfect. I rarely have to record live instruments, so the space will be used primarily for tracking and editing, but having the capability to record live if I need to is most welcome.

"After our second meeting I bought two speaker stands for the Mackies and ditched the overbridge from the desk. Thus I now have space for even more TFT screens and can load the bass as much as I need without the desk resonating its way through the floor.

"Ironically, although everything was done with 'pick up and go' portability in mind, I now like it so much that I'm content to stay!"

▶ well indeed, and is quickly configured from the Record Device Selector drop-down menu.

Cycle recording is now also possible. If a loop region is defined and recording is activated, individual audio or MIDI clips are created for each pass through the loop region. When recording is complete, right-clicking on the topmost clip brings up a drop-down menu from which, amongst other things, other clips can be bought to the front to be auditioned. As with the mainstream MIDI + Audio sequencers, this sort of functionality is great for cycling through a song section until the perfect take has been captured. It is also fairly straightforward to use multiple clips to compile a perfect take from the best bits of each pass. For example, if cycle recording has been used to capture multiple takes of a lead vocal, the Split function can first be used to divide the clips between each sung phrase. The topmost clip for each individual phrase than then be changed until the best comp of the vocal performance is identified.

The major improvement in *Acid*'s MIDI features is the inclusion of in-line editing. As anyone who has used similar features in something like *Cubase SX* will know, this makes editing of MIDI data much easier as it can be viewed in context with the timing of events on other tracks. In either the Piano Roll or Drum Grid, note position, velocity (through the use of very intuitive 'velocity stems'), pitch-bend and controller information can all be edited in place. In use, I found this worked very well, offering enough features to make the most common MIDI editing tasks very easy, but without getting bogged down in a glut of features that the majority of users would never need.

Kompakt For Acid

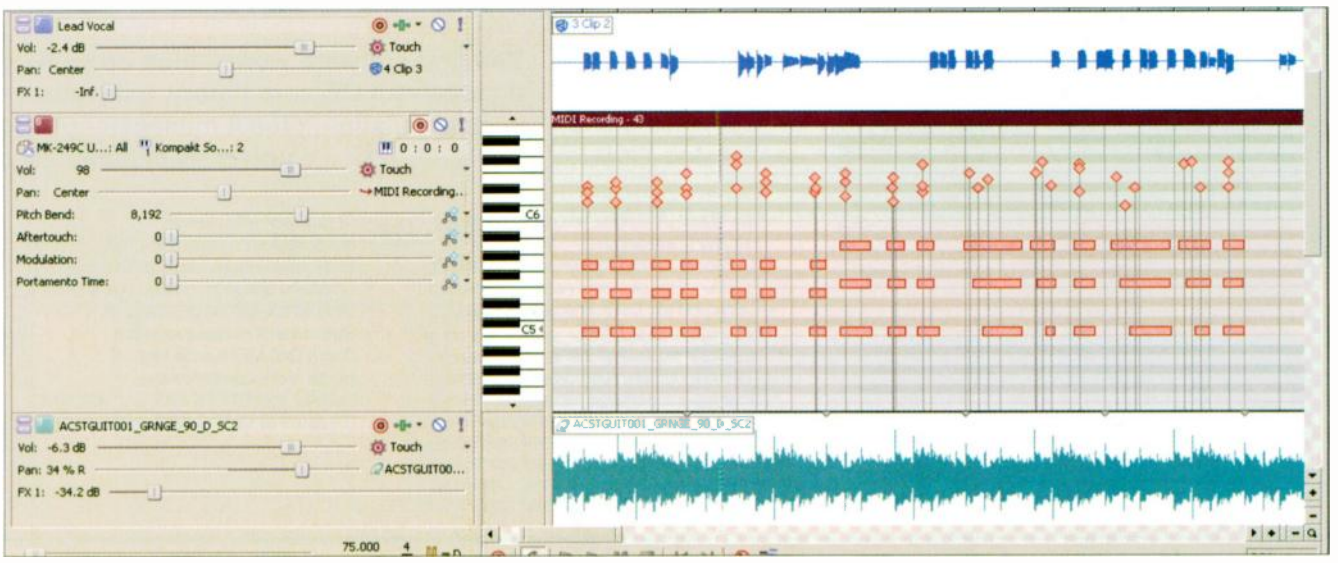
As part of an ongoing relationship between Sony and Native Instruments, *Acid Pro* is now shipped with a custom version of NI's *Kompakt* software sampler. This is not dissimilar to the various *Kompakt*-based sample libraries that *SOS* has reviewed over the last year or so (for example, *Nostalgia* or *Vapor*) in that it contains a fixed library of sounds that cannot be accessed via another sampler; nor can this version of *Kompakt* load sounds from other libraries. However, I was able to use this *Kompakt* VST Instrument within *Cubase SX*, so while it is a custom version supplied with *Acid*, it can be used under other hosts.

I suspect some of the samples may be derived from East West's *Colossus*, and they cover bread-and-butter territory — pianos, keyboards, guitars, basses, drums, synths and ethnic instruments — with well over 100 different instruments in total. Free or not, some of the sounds are very good. There are some very respectable piano sounds and the various synth and bass patches are also good, but my personal favourites were the various drum and percussion programs. Here there is plenty of choice, ranging from acoustic kits through to basic orchestral percussion.

The custom version of NI's *Kompakt*, with its 2GB sample library, is a nice addition to *Acid*.

Bits And Bobs

These headline features aside, there are a whole raft of smaller but nevertheless worthy additions and improvements in *Acid Pro 6*. For example, *Acid* now includes support for external control surfaces; a generic template is supplied that can be configured for any external controller, and there's native support for Mackie Control and the Frontier Designs



In-line MIDI editing improves *Acid*'s potential as a useful MIDI sequencer.

Tranzport. Improvements have also been made to the automation, the most important of which is the addition of support for VST Instrument parameter automation. Given the improvements in MIDI recording and editing, this better VST Instrument support will be welcomed by anyone who now wants to use *Acid* as a complete music production environment.

Options for filtering and processing MIDI data have also been improved, and MIDI File import and export is now supported. In addition, there is now a Drum Map Editor that can be used to create a Drum Map for editing — this is much easier to use than the Piano Roll editor when working with a drum synth or sampler.

I suspect that the audio engine within *Acid* has undergone significant changes in order to provide support for multiple audio clips on a single track. In use, I certainly did not notice any performance hit because of this — comparison of playback of a range of projects in both *Acid Pro 5* and *Acid Pro 6* produced similarly smooth results. However, the engine in the new release now also provides support for dual and multi-processor PCs. I didn't get the opportunity to test this (I'm currently building a new dual-core Athlon system, but at the time of writing, it is not yet fully configured) but I'm sure power users will find Sony's explicit statement of support for dual-core systems reassuring.

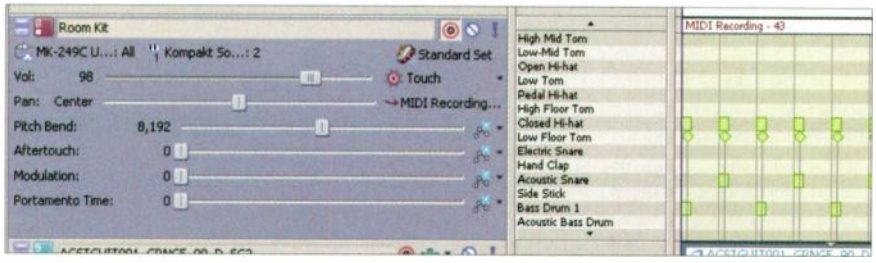
What Next?

As with any software application, most *Acid* users probably have a wishlist of additional features that would be useful. My top two would be a more comprehensive mixer window that would allow both track and buss faders to be viewed together, and a function equivalent to *SX*'s Detect Silence for automatically splitting audio files. The latter would be very useful when doing jobs such as vocal comps in conjunction with *Acid*'s improved cycle-mode audio recording. It would be even better if there was also a function that could automatically shift all the clips stacked on one track to a series of identical tracks each containing one of the takes. These could then be more easily auditioned, before cutting and pasting the best phrases for each part of the vocal onto a single track to create the 'master' take.

Conclusions

As with version 5, this new release of *Acid* contains some big new features that make it both revolutionary and evolutionary, and I would think the majority of existing *Acid* fans would consider the upgrade essential. While I still wouldn't choose *Acid* over something like *Cubase SX* if the majority of my work was based predominately around live audio recording or serious amounts of MIDI such as complex orchestral arrangements, Sony have now moved *Acid Pro*'s functionality firmly towards that of the mainstream MIDI + Audio sequencer, and this ought to broaden its appeal considerably.

The audio and MIDI recording options are now certainly good enough for serious multitrack work and I can easily imagine myself creating complete projects combining loops, audio recordings and MIDI-driven soft synths/samplers entirely within *Acid Pro 6*. This is not something I would have contemplated in version 5. What's more, *Acid* remains, for me at least, the only audio software in my studio that manages to both tremendously powerful and also downright fun to use. If you have never taken *Acid*, there has never been a better time to try it. Highly addictive and highly recommended. SOS



Drum Maps can be defined for MIDI editing when using drum synths or samplers.

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Win

Audient Sumo & ASP008

Worth £2400

Over the past nine years, analogue mixing stalwarts Audient have built a reputation for producing high-quality studio equipment, specialising in no-nonsense analogue products with top-notch sonic quality. Two of their most recent launches, the ASP008 eight-channel microphone preamp and the Sumo summing amplifier, have caused a stir in the recording industry by offering sensibly-priced, top-quality analogue circuitry in products designed as front-ends (and rear-ends) to digital audio workstations. Thanks to Audient, both units, with a total value of around £2400, are up for grabs for one lucky winner in our competition this month.

The ASP008 and the Sumo have both been reviewed by Paul White in *SOS* this year — the Sumo in February and the ASP008 in April. Check out those reviews for full details. In the meantime, let's whet your appetite with a few facts and figures...

The ASP008 crams eight of the same preamps found on the high-end ASP8024 recording console into a 1U rack. All channels can handle line and microphone signals and have individually-switched phantom power. Other front-panel features include phase invert, LF cut, variable input impedance and line/mic switches, with additional instrument inputs and -20dB pads on channels one and two. There are signal present and clip LEDs on all channels and input impedance can be switched between 200Ω, 1200Ω and 5000Ω, meaning that the ASP008 can cater for a range of microphones — including ribbon varieties. Line inputs and outputs are on 25-pin D-Sub connectors, while the mic inputs are on standard XLR sockets. The two front-panel instrument inputs take a quarter-inch jack.

To allow better integration with DAWs, there's also a range of digital output options for the ASP008, and part of our prize is the ASP008 ADAT-only card, which streams all eight preamp outputs down an optical cable at up to 48kHz. You can also add AES-EBU and S/PDIF outputs, plus word-clock sync on a BNC connector, with another card that's available from Audient (AES-EBU, S/PDIF and BNC option not included in prize).

Audient's other analogue rackmount unit is the Sumo, which is a summing mixer designed to take multiple DAW outputs and sum them to stereo with greater fidelity than a computer's internal digital summing techniques. Using the same mix-buss architecture as the ASP8024 console, the Sumo delivers a crystal-clear stereo output free of digital summing artifacts. It features

eight stereo inputs on 25-pin D-Sub connectors and stereo XLRs for mix and monitor outputs, and also has switchable balanced insert points, allowing external processors to be used before or after the on-board dynamics section. The monitor section has an output level control and mono compatibility-checking button, and up to four Sumos can be linked together to achieve 64-channel mixing. There's also an impressive metering section that has a very accurate 26-segment LED display for the mix signal and an eight-step gain reduction meter for the dynamics section. The lucky winner of the Sumo may want to add its digital option card, providing AES-EBU and coaxial S/PDIF outputs alongside BNC word clock, at a later date (digital card not included in prize).

For more information on Audient's product range, including the forthcoming Centro master section, mentioned in this month's news, check out the Audient web site, at www.audient.co.uk.

If you would like a chance to win both of these fabulous prizes, simply fill out the entry form at the bottom of this page and post it to the address on the coupon. Alternatively, you can enter via the electronic form on the *Sound On Sound* web site, www.soundonsound.com. Please make sure you answer all the multiple-choice 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 the 31st August, 2006.

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1. Only one entry per person is permitted. 2. Employees of SOS Publications Ltd, Audient PLC 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.

questions

How many individual mic preamps does the ASP008 have?
a. Nine ☐
b. Eight ☐
c. Seven ☐
d. Six ☐

What is Audient's forthcoming master section called?
a. Centrino ☐
b. Centrana ☐
c. Centrum ☐
d. Centro ☐

From which Audient console is the technology used in the ASP008 and Sumo derived?
a. ARP2600 ☐
b. AMP8816 ☐
c. ART2001 ☐
d. ASP8024 ☐

Audient tie-breaker

If you were a Sumo wrestler, what would be your special move? Answers in 30 words or fewer, please.

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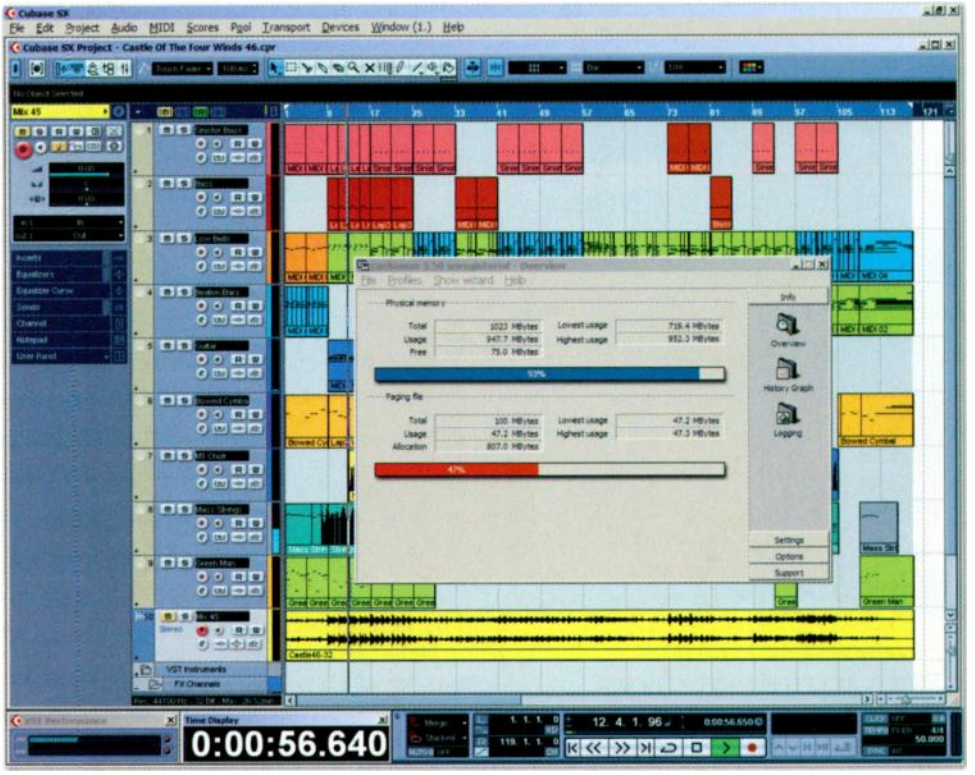
Would you like to receive more information on Audient products? If yes, please tick or cross this box. ☐

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

Overcoming Overload

PC System Overload Problems & Workarounds

There are many factors that can cause your PC to struggle when playing back your songs — including RAM, your hard drive, your CPU and your system settings. But how do you know which is to blame, and do you have to upgrade or can you work around the problem?



You don't need to make 'guesstimates' about how much of your system RAM is being used in total by all your currently running applications — just use a freeware utility such as *Cacheman*, shown here.

Martin Walker

In my PC Musician feature of *SOS* June 2003, I explained how the various PC resources (CPU, RAM and hard drive) are used up by audio tracks, soft synths, soft samplers and so on, to help PC users decide on the spec that would meet their needs when buying or building a new model. However, many musicians face the opposite dilemma: they have a PC that has problems running some songs and are not sure whether they need to upgrade their soundcard, install more RAM, buy a faster processor or get a larger, faster drive to resolve them.

As you can see from the *SOS* Forum survey results in the box on page 70, there's no such thing as a typical song, so if you're in the situation described above you need to do a little detective work to find out what's causing the problems in your particular case. To help, I decided to take some typical problems reported by *SOS* Forum members, explain their most likely cause and point out the best solution, as well as offering temporary workarounds for those who can't afford to upgrade at the moment or have a project that urgently needs to be finished.

Running Out Of RAM

The Problem: "I'm having big problems ever since I bought Spectrasonics' Atmosphere to plug into my *Fruity Loops Studio* software. It takes up a lot of space on my hard drive (3GB) and it's now come to the point where, if I add more than, say, six channels of it into *FL Studio*, the whole piece stutters so much that I can't play it for more than three seconds. The CPU meter also hits 100 percent every time it starts stuttering. I have 512MB RAM in my computer. Should I get more? I think my processor is running at 3.19GHz. If I don't need more RAM, would an external soundcard be the answer? (I have a laptop.)"

The Diagnosis: This user suspects his RAM, processor and audio interface. As an *Atmosphere* user myself, I instantly know which of these three is the real culprit: lack of RAM. This is simply because *Atmosphere* (and its stablemate *Trilogy*) load their patches entirely into system RAM and each one can consume up to 125MB. With six instances, a PC that has 512MB RAM could well be struggling, and once the RAM is nearly all in use Windows will start to ferry whatever data it can to virtual RAM (a cache on your hard drive). Thus every time the song tries to access a different *Atmosphere* patch, some of the data may have to be retrieved from the hard drive — hence the severe stuttering. The CPU meter hits 100 percent because the PC can no longer process the song in real time, due to the extra time it takes to keep reloading that data.

If you suspect that lack of RAM may be the cause of your own problems, there's never any need to guess, or to blindly upgrade just in case — just load in your most complex songs and monitor their RAM usage for yourself. You can read the Available Physical Memory in Kilobytes from the Performance

page of Windows Task Manager. However, I've always found that the freeware *Cacheman* utility (www.outertech.com) provides a much clearer and easier-to-use display of both Physical and Paging File memory, as a percentage bar-chart, so you can see at a glance what proportion of your RAM is already used and what remains (see screen on left). Just launch this periodically to see how much RAM is still available.

Out of interest, I ran *Cacheman* as soon as I reached the Windows XP desktop on the Internet-enabled General Purpose partition of my PC, and 300MB (30 percent) of my 1GB total had already been used up. However, on my stripped-down, music-only partition, only 200MB (20 percent) had been used. An extra 10 percent of available RAM is well worth having, and it's yet another reason to create a multi-boot setup.

Once I'd loaded *Gigastudio* 3.1, the 'RAM in use' percentage rose to about 30 percent, and to about 40 percent once I'd launched *Cubase SX* 3.1, leaving my system with 600MB for instruments. My typical projects have about half a dozen soft synths and the same again of *Giga* instruments. As long as I'm careful, I can keep loading stuff in until the *Cacheman* memory gauge reads 99 percent without any juddering or other mishaps.

The Workarounds: Once a RAM shortage is indicated, the obvious cure is to install more RAM, although there are some limitations when running Windows XP, if you need more than 2GB (see PC Notes May 2005 for more details). Some soft samplers offer engine adjustments that reduce RAM consumption in favour of more CPU load, which may be worth a try if you're a heavy sampler user.

If you're running close to the edge with RAM, try saving your song, closing your sequencer, and then relaunching it and reloading your song. If you've been making lots of edits and trying lots of soft synths, Windows may be caching unused data that will be released using this method. However, the easiest way to release RAM is to choose different soft synths — as you can see from the chart below, some can need only a few MB per instance, while others can swallow 100MB!

Hard Drive Dramas

The Problem: "My *Cubase SX* audio glitches when it's playing back just one audio track, when there are another 52 muted. I'm using my Audiophile 2496 ASIO driver and have a 3.2GHz Pentium 4 PC with 1.5GB RAM."

The Diagnosis: Here's an intriguing one. You wouldn't expect any PC to have problems replaying a single audio track. The twist in this tale is that while the muted audio tracks aren't contributing to the mix, they are still being streamed by *Cubase*, just in case you hit those mute buttons and want the tracks instantly added to your mix. Whichever sequencer you use, you can confirm such behaviour either by watching its Disk Meter load, or by saving your song, deleting all but the one track and trying again. Your hard drive should now show minimal loading. Even the CPU overheads of plug-ins used by muted tracks are still weighing down your song, whether you hear them or not, although in this case you can temporarily disable them via their on/off switches.

So in reality this person is running 53 audio tracks and, assuming that these are 24-bit/96kHz, is beginning to aproach the limits of most single 7200pm hard drives. I've personally managed to run 76 mono 24-bit/96kHz tracks on my Seagate Barracuda SATA 80Gb ST380013AS audio drive before it ran out of steam, but this was with long tracks, each lasting ▶

RAM Overhead for Pentium 4 2.8GHz CPU PC

Make	Model	1 patch (MB)	4 patches (MB)
Steinberg	Pro 53	3	6
Steinberg	Hypersonic 1	20	22
AAS	Lounge Lizard 2	16	37
AAS	Tassman	18	51
AAS	String Studio	20	61
AAS	Lounge Lizard 3	23	71
Korg	Wavestation	38	73
Korg	M1	95	95
Edirol	HQ Orchestral	111	111
Emu	Emulator X	147	147
NI	Reaktor 5	30-64	111-185
Spectrasonics	Atmosphere	18-160	60-617

Many musicians don't realise how much soft synths vary in their RAM consumption. Here's a chart showing how some of the ones in my collection compare. This demonstrates that while simple analogue-style synths may only take a few megabytes, those that rely on samples can take considerably more, while synth 'designers' such as Native Instrument's *Reaktor* can vary considerably from Ensemble to Ensemble. Notice also that multitimbral synths such as Edirol's *HQ Orchestral*, Emu's *Emulator X* soft sampler, Korg's *M1* and Steinberg's *Hypersonic* may take lots of RAM for the first patch, but no more when you access several instruments on different MIDI channels using the same engine.

PC SYSTEM OVERLOAD PROBLEMS

In this *Cubase SX* Project, you can see that the Disk Meter (lower left) is displaying an overload, even though only a single audio track is actually playing. Muting the other 50 or so tracks makes no difference, because they are still being streamed in anticipation of you un-muting them and wanting to hear them immediately.

the whole duration of a song. With shorter parts being dropped in and out, some audio editing and some inevitable file fragmentation, this figure will certainly drop.

If you're also running a soft sampler such as *HALion* or *Kontakt* inside your sequencer, this can greatly increase the load on your hard drive. You need to be especially careful if you're running a separate soft sampler such as *Gigastudio*, since its additional load (both CPU and Disk) may not show up on your sequencer's performance meters.

There's another fairly common reason for a hard drive to conk out when it's only playing back a couple of dozen audio tracks. This problem is often exhibited as intermittent glitches (even though the sequencer's CPU meter displays a low overhead reading), and it's due to an unsuitable Buss Master DMA mode. You can check this inside Device Manager:

- Select 'View Devices By Connection'.
- Locate the Storage Controllers on the list, expand their entries, and you'll find your hard drives and optical drives attached to either Primary or Secondary Channels.
- Right-click on the Channel connected to each drive and select 'Properties'.
- Click on the Advanced Settings tab. For each of the two devices that can be connected to each channel you'll find a Device Type (normally set to 'Auto Detection'), and a Transfer Mode, which should read 'DMA if available'.
- If it reads 'PIO Only', your hard drive will be running well below par, so change it to the other setting, click on the OK button and then reboot your PC. (When this Transfer Mode does read 'DMA if available', the 'Current Transfer Mode' box beneath it should typically display 'Ultra DMA Mode 5'.)

A quick way to check the performance of your hard drive is with a utility such as *Dskbench* (www.sesa.es/us/dl/dskbench.zip) or

The Typical Song?

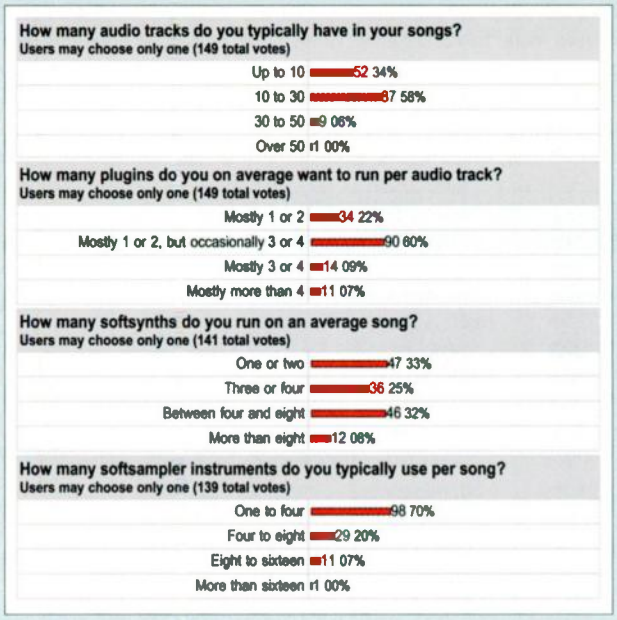
I think it's fair to say that each PC musician will approach each of their songs in a completely unique way. In the case of sound sources, for instance, some people will exclusively record audio tracks, some will mostly rely on triggering external MIDI synths, some will solely use soft synths and others will rely almost totally on software samplers (particularly for orchestral creations). Where effects are concerned, some purists may rely totally on mic positioning to capture the live audio sounds they want, and not use a single EQ, compressor or other effect, while others may load an EQ, compressor and gate plug-in on every audio channel, as well as other insert and send effects.

With this in mind, some months ago I organised a small survey on the *SOS* PC Music Forums, to find out just what the spread was. You can see the results so far in the screenshot on the right, and if you want to add your votes you can still do so, at www.soundonsound.com/forum/showflat.php?Cat=&Board=PCMus&Number=217340.

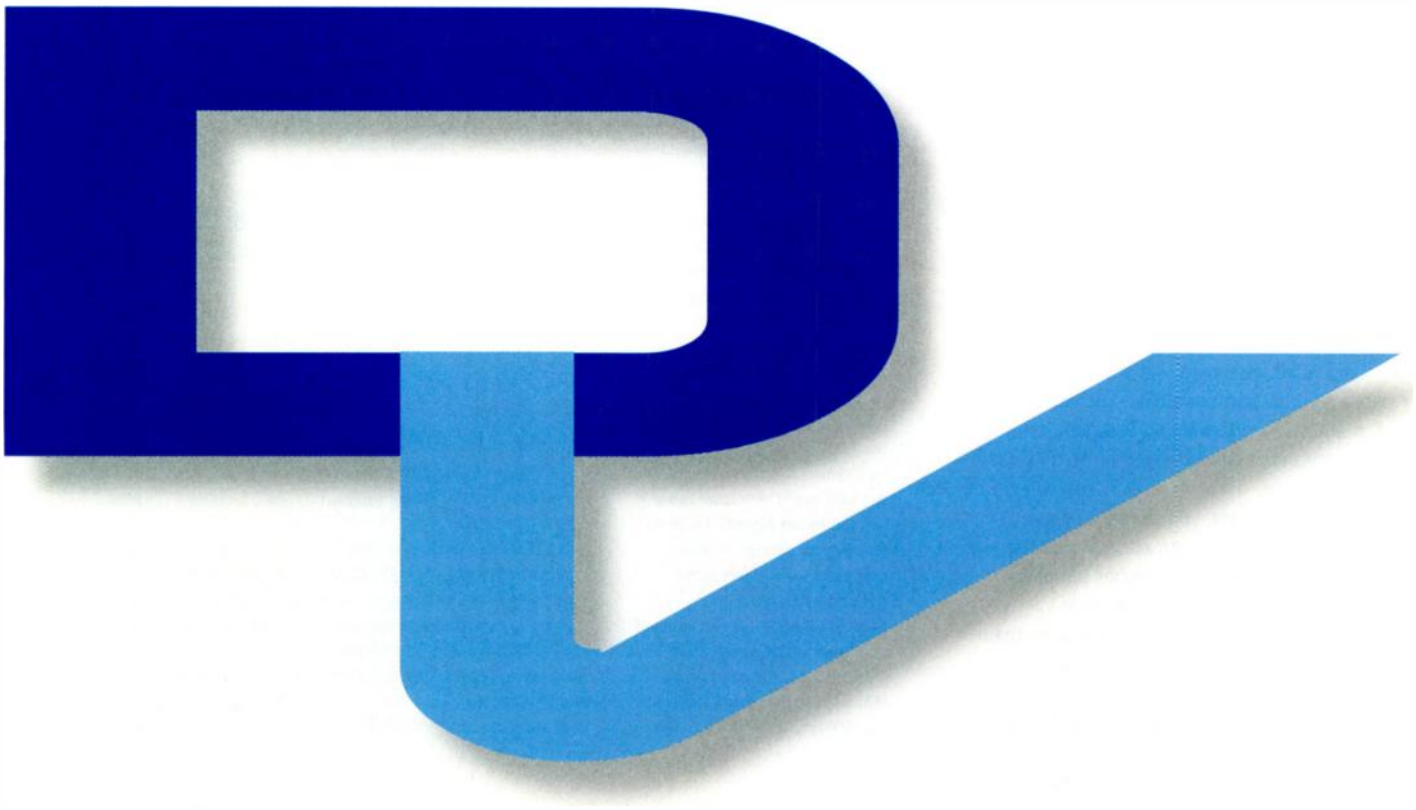
As you can see, a large majority of musicians who replied seem to use between 10 and 30 audio tracks and mostly one or two plug-ins per track, but occasionally three or four, and very few software sampler instruments. However, there's a huge variation in the typical number of soft synths used in each song, although most people seem to use less than eight.

Given the very different demands of specific soft synths, few musicians are therefore likely to stress their PCs in exactly the same way. However, if your PC is set up sensibly, and isn't running loads of background tasks, we can deduce from the results of the survey that everyone who took part should find a lowly single 5400rpm hard drive quite sufficient to fulfil their audio track requirements and would probably get away with 1GB of RAM (although 2GB is always welcome, as memory doesn't cost that much nowadays).

The limiting factor in most cases is likely to be processing power, which is why so many musicians find their sequencer 'freeze' functions so handy, as these let you return the CPU overhead of a track to the pool, by converting the track to a new processed audio version.



Here are the results of an *SOS* Forum survey I set up to find the typical content of a PC musician's typical song. As you can see, the biggest variable is soft synths, which can also consume the biggest proportion of your CPU power. The reason why there are fewer votes in the soft synth and soft sampler categories is that some musicians run none of them.



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HDTach (www.simplisoftware.com/Public/index.php?request=HdTach). These will show up whether or not your hard drive is using the most suitable DMA mode. You should achieve an average or sustained read speed of at least 40MB/second with most modern hard drives, with a CPU utilisation of two percent or less.

The Workarounds: If your hard drive is operating in the correct mode but you still regularly find it beginning to struggle, there are several ways forward. For a start, defragmenting your audio drive may improve matters (although occasionally this may make things worse — see PC Musician June 2005 for more details).

If you're running a VST soft sampler and have plenty of spare system RAM, you may be able to offset some drive load into this, courtesy of its disk-streaming parameters. Alternatively, it may be time to investigate the partitioning options I discussed in the 'Partition Decisions' feature in *SOS* May 2005. Creating a separate partition just large enough for the current project, on the fastest 'outside' portion of your drive, may let you run more audio tracks without spending any more money at all, assuming that you've already got a suitable partition utility, such as Symantec's *Partition Magic*, Paragon Software's *Partition Manager*, or Acronis' *Disk Director Suite*.

In the long term, you should also ask yourself whether you can actually hear the improvements offered by higher sampling rates such as 96kHz on your system. Remember that not only will these more than double the hard-disk load over using a 44.1kHz sample rate, but also that any plug-ins and soft synths you use will consume more than twice as much CPU overhead, as proportionally more calculations are needed per second. At the very least, anyone whose final format is to be Red Book Audio CD should consider moving from 96kHz to 88.2kHz on future projects, if their audio interface offers that sample rate. This will reduce drive loading by over eight percent.

If you're running absolutely loads of audio tracks (one *SOS* Forum poster admitted to a song containing 330, but with only 50 to 75 at any one time!) you should perhaps take a closer look at your working methods, unless you're mocking up an orchestral score and layering multiple instruments for each part. If this isn't the case, perhaps rethinking your working methods will result in significantly less hard drive torture.

However, ultimately it may be time to invest in faster hard drives: a 10,000rpm model should boost your audio track count significantly, or you could investigate RAID (Redundant Array of Independent Disks) by installing two identical drives configured as RAID 0, to potentially double the overall transfer rate. Just remember that your data is more precarious on such a system, because it's spread across multiple drives. Also bear in mind that both 10,000rpm and multiple drives are likely to increase the overall acoustic noise level of your PC.

Soft Synth Overheads

As discussed in the main text, running out of processing power is probably the most likely reason why songs might start glitching or coming to an untimely halt. Buying a faster processor, or a completely new PC with faster everything, is the obvious answer. However, not all of us can have that luxury, and fortunately there are various other things you can do to stretch the CPU you do have as far as possible.

- Cap polyphony: Many soft synths let you set a maximum number of playable notes, and you can often greatly reduce CPU overhead by simply reducing this number. Solo the track in question, slowly reduce the number of notes until you hear the first signs of note-stealing, and then advance the setting to the next highest value. You may be surprised at how low you can go without this capping becoming audible, and it may even (depending on the algorithm used) clean up your mix by removing notes late on in their decay phase that contribute little to the composition.
- Adjust Sequencer Settings: Some sequencers (*Cubase*, for instance) provide a similar 'Restrict Polyphony' function, this time shortening any overlapping notes when required, to avoid overstepping the mark. However, this may produce audibly different results than directly capping the soft-synth polyphony, because the soft synth may have more sophisticated algorithms at its disposal to determine which notes are least likely to be noticed when removed.
- Examine Sustain: If you've used a sustain pedal during your performance, it can often be difficult to know what the true polyphony of your track is, so look for special functions in your sequencer, such as *Cubase*'s 'Pedals to Note Length', which scans for MIDI sustain controller on/off commands, lengthens affected notes to match the pedal 'off' position and then deletes the sustain commands. Once you see the true length of your notes it may become obvious why a particular track is so resource-hungry!
- Select Appropriate Instruments: Each soft synth takes a different number of CPU cycles for each note, depending on how many oscillators, filters, LFOs and so on it uses, or the complexity of its physical modelling. Spend a little time getting to know the appetites of your favourite soft synths, so you can bear this in mind when choosing your sounds. For instance, I love the AAS *String Studio* software, but its physical modelling can eat processors for breakfast, so I only use it for exposed 'lead' instruments. Take a look at the chart below for a breakdown of how a selection of different soft synths compare.

CPU Overhead for Pentium 4 2.8GHz CPU

Make	Model	Voices	CPU	16 voices
Korg	M1	64	25%	6%
AAS	Lounge Lizard 2	32	25%	13%
Steinberg	Pro 53	32	30%	15%
AAS	Lounge Lizard 3	32	35%	18%
Spectrasonics	Atmosphere	48	80%	27%
Korg	Wavestation	44	80%	29%
Emu	Emulator X	24	64%	43%
AAS	String Studio	8	60%	120%

Here's a chart showing how some of the soft synths in my collection compare when running on my Pentium 4 2.8GHz processor, arranged in ascending order of CPU 'appetite'. In each case I either ramped up the polyphony to its maximum value, using a specially prepared sequence, or took it to a point where the CPU load was fairly high. In the fourth column I've calculated the relative amounts of CPU power that each would need to play 16 simultaneous notes. It's quite revealing, so choose your instruments carefully!

Processor Pitfalls

The Problem: "I have a PC laptop with a 2.8GHz P4 processor and 1GB of RAM. Recently I've been using a lot more VST Instruments than I usually use in *Cubase SX 2* and also in *Fruity Loops*. My CPU use hits around 50 percent and is fine for a couple of minutes, then will suddenly jump to 95 or 100 percent, where it stays (slowing the computer down immensely) until I finally close *Fruity Loops* or *Cubase* (stopping the song from playing doesn't help). I've loaded the same song on my housemate's 1.8GHz Athlon desktop, it only registers 40 percent on his CPU meter and the spikes and crashing aren't an issue. What's going on? Is this a Pentium 4 issue, an overheating issue, or possibly a problem with some software on my computer?"

The Diagnosis: Some PC laptop owners have experienced their CPU meter slowly rising for no apparent reason, and the surprising cause turns out to be overheating due to dirt and muck completely clogging up the cooling fans underneath the laptop. As the processor temperature rises, the processor clock speed is automatically throttled down to prevent damage. The cure in such cases is either to have the computer dismantled and properly cleaned by the manufacturer, or to attempt DIY cleaning using compressed air or a vacuum cleaner.

PC SYSTEM OVERLOAD PROBLEMS

However, in this case the CPU rise is sudden, which points instead to 'denormalisation' problems (discussed in PC Notes October 2002 and affecting several processors, most notably the Pentium 4 range). Most plug-ins have long since been tweaked to avoid the issue, but if your CPU usage suddenly jumps at the same specific point in the song each time, this could be when an elderly plug-in causing the problem has an extremely low audio-input level or silence to deal with. You can track down which one by selective disabling, and cure it by inserting a corrective plug-in such as Sascha Eversmeier's freeware *Normalizer* (www.digitalfishphones.com/binaries/normalizer_PC_Win.zip).

Another major cause of laptop CPU problems is the increasing sophistication of technologies such as Speedstep, Smart CPU and Cool 'n' Quiet, used by processor manufacturers to minimise the power consumption of their products. These work by throttling CPU clock speed down to a lower value when you're not using 100 percent of your processor, as well as in some cases reducing CPU voltage and fan speed, keeping temperatures and fan noise down and extending laptop battery life.

Although these are wonderful features for most computer users, unfortunately musicians have to battle with the audio consequences of them — such as clicks, pops or longer interruptions. These normally happen when the CPU clock speed is ramped up or down, so the safest option is to disable such behind-the-scenes trickery and leave your CPU at its top speed. Some laptop users have also found their CPU being reported by Windows as having a much slower clock speed than the one they bought — once again, this could well be due to throttling.

The easiest way to ensure maximum performance is to open the Power Options applet in Control Panel and select the 'Always On' Power Scheme. If you want a more versatile way of switching between the various Power Scheme options, including those normally hidden from the user by Windows XP, try downloading the *Speedswitch XP* utility (www.diefer.de/speedswitchxp/index.html) that I discussed in some detail in PC Notes December 2003. You may also want to disable any special drivers involved in throttling, if there's such an option.

The Workarounds: Apart from the issues just mentioned, most CPU 'maxing out' problems are simply a result of attempting to run more plug-ins or soft synths than your PC can handle. Since soft synths are the most likely culprits, I've dedicated a box to Soft Synth Overheads (on page 72). Elsewhere I also mention track 'freezing' and reducing project sample rate, as the most radical cures.

However, there's another step you can take before having to get out your credit card to upgrade your processor or buy a DSP card such as the TC Powercore or Universal Audio UAD1. It's now fairly well known that below about an audio buffer size of about 12ms the CPU load rises due to ASIO driver overheads, so if you're running out of CPU and using a latency of less than this you'll reclaim processing power by changing the latency value. If you're running at 3ms or below you may be able to reduce your CPU load by 50 percent or more! SOS

CPU & Disk Meters

Most sequencers offer some sort of meter displaying what proportion of your processor is currently occupied with playing a song, and some also offer a similar meter for the load on your hard drive (not how much space is left on it, as one musician thought — the manual can occasionally be useful, you know!). However, there's a lot of confusion about what these meters actually measure, particularly when the processor figures may vary significantly from Windows' own CPU Usage meter (found on the Performance page of Task Manager).

Steinberg's *Cubase* range doesn't actually measure CPU overhead, but instead times how long it takes to calculate all the plug-in and soft-synth contributions in your song, mix them together and use this data to fill one of your audio interface buffers. If (for instance) you've set your buffer size to 6ms and the calculations required to fill it take 3ms, then the meter will read 50 percent. Similarly, drive meters measure the time taken to retrieve the audio data that needs to be sent to your audio interface and compares it with the total available. A few applications 'hang on' to remaining CPU time, so while their own CPU meter may register a realistic figure, the Windows one remains permanently at 100 percent.

You can't always rely on CPU meters to tell you the whole story. Here's AAS' *String Studio* running as a stand-alone synth on a Hyperthreaded CPU. As you can see, *String Studio* itself measures an 80.9 percent CPU overhead, while Windows Task Manager measures just 41 percent. The reason is that only one of the two processors is being used, Task Manager indicating the 'average' overhead of the two. If you were running *String Studio* inside a multi-threaded application such as *Cubase SX*, the various soft synths would be more evenly shared between the two processors.

Interface Issues

Judging by posts on the *SOS* Forums, quite a few musicians point the finger of blame at their audio interface when songs run into problems, but apart from occasional driver issues the interface is rarely the culprit — except at low latency. As mentioned in the main text, if you reduce your interface buffer size below about 12ms your CPU overhead will nearly always rise, and will do so radically below 6ms. So to rule out your interface from song problems just increase latency to 12ms or above, if only temporarily.

Nevertheless, some musicians have, over the years, complained that a song they were working on one evening that played back perfectly won't do so without severe glitching when they load it up the next morning. Others may find this scenario hard to believe, but a new benchmark is now demonstrating that this behaviour is indeed possible at low latency with some interfaces (see this issue's PC Notes column for more details).

If your songs suffer from occasional clicks and pops you should note down when the clicks happen and what you (and your song) were doing at the time. If the clicks are fairly regular it could be a background task cutting in, so examine the ones that are running, using Windows' Task Manager, and try temporarily stopping them, to see if that cures the problem. If clicks only happen on some songs and not others it could be a rogue plug-in or soft synth. You can temporarily move the contents of your VST plug-ins folder elsewhere and methodically move them back until the problem recurs. If the clicks seem totally random, the problem might unfortunately be some incompatibility between your audio interface and another hardware component in your PC, so try searching on-line audio forums for anyone suffering from similar problems.

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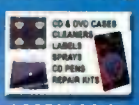
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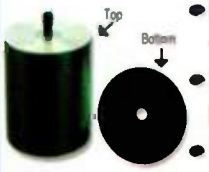

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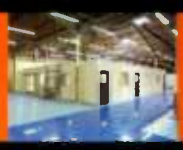

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NHT Pro M60 & XDA

Monitors & Digital Amplifier System

This new pair of high-resolution active monitors features sophisticated Class-D amplification and DSP processing options.

Hugh Robjohns

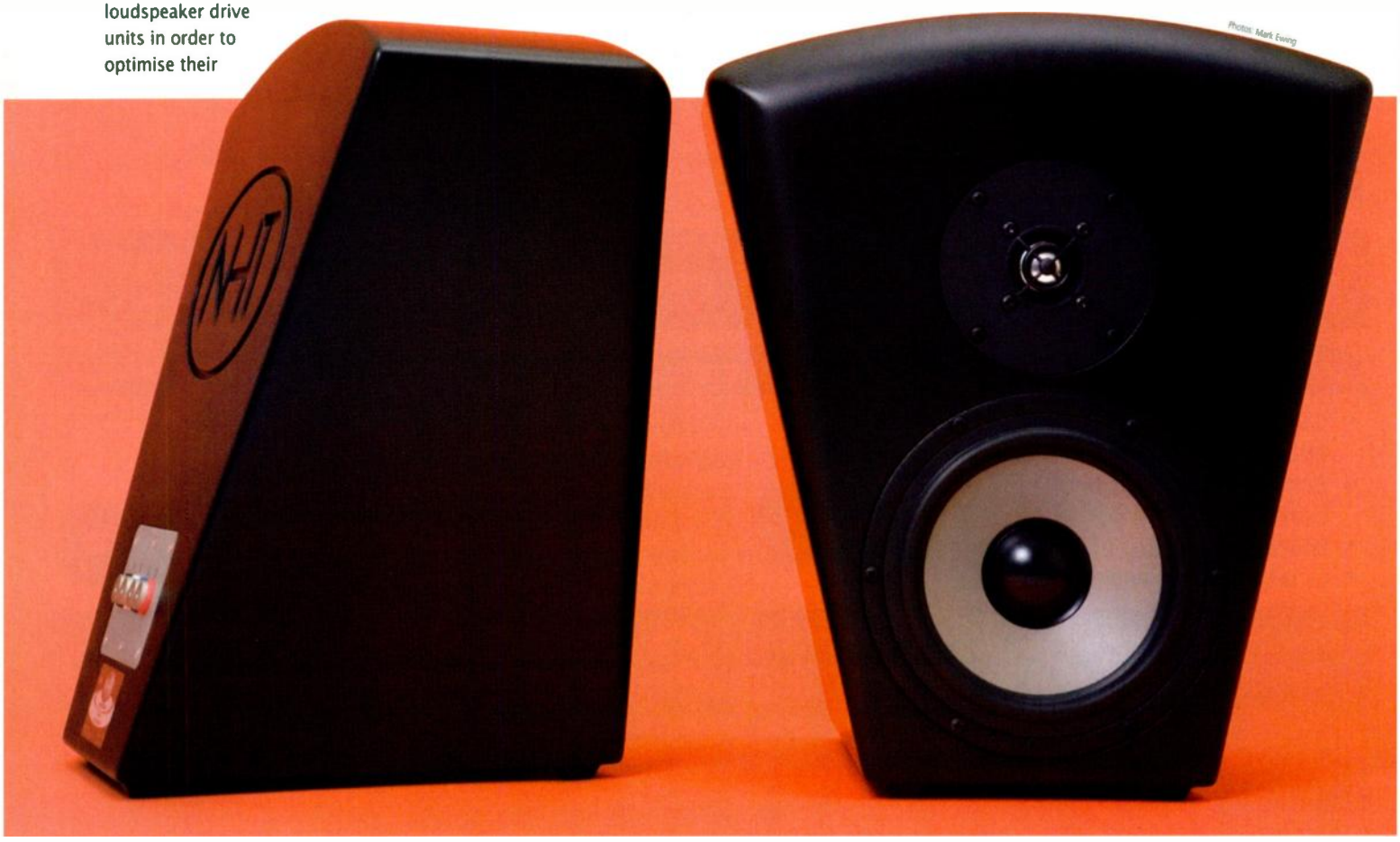
The Californian company NHT's latest offering is a two-way active nearfield monitoring system comprising a pair of passive speakers which are connected via a four-way, 4mm banana-socket connection panel to a separate unit containing four 150W Class-D amplifiers. The crossover functions are performed by an integral DSP system, which has the potential to be upgraded and modified via the built-in USB 2 interface.

NHT have used an OEM digital crossover design from DEQX in Australia. This technology is designed to integrate loudspeaker drive units in order to optimise their acoustic performance and minimise their mechanical deficiencies. As well as simple frequency-response correction, this system is also claimed to optimise the system's time-domain response — and all with a latency of under 10ms. Distortion is specified as below 0.5 percent across the entire 55Hz-to-20kHz bandwidth at 90dBSPL, and less than 0.3 percent above 100Hz. The on-axis frequency response is apparently flat within ±0.5dB over the full range.

Innovative Cabinet Design

The pictures of the speaker cabinets might look like someone has been playing with the skew transformation facilities in *Photoshop*, but they haven't — the speakers really do look like that! No two surfaces are parallel and everything is just madly wonky, with an egg-shape top tailing off into a narrow foot, and the rear panel sloping out towards the bottom. NHT's literature explains that the widened baffle around the tweeter is intended to promote an even power response, and keeping the box narrow at the base reduces mid-range reflections. The edges of the baffle are well radiused to avoid secondary radiation, and this is an unported cabinet.

The cabinet is machined from MDF panels of different thicknesses, and the absence of parallel surfaces helps to minimise the formation of internal standing waves, of course. The maximum cabinet dimensions are 15 x 13 x 10 inches, and each speaker weighs 17.5lbs. The drive units comprise a 6.5-inch woofer with a magnesium cone and a 'super linear motor', and a one-inch aluminium-dome



Photos: Mark Ewing

Four status indicators on the front of the amplifier show the settings of the internal DSP filter bank.

tweeter with ferrofluid cooling and a textile surround. Both units are magnetically shielded. The connections are all standard 4mm sockets, colour and letter coded (A-D), and the supplied four-way cables are fitted with a colour- and letter-coded four-way banana plug that simply slots straight in.

Class-D Amplification

The DSP processor/amp is surprisingly compact, but that's the beauty of Class-D amplifiers. Even the internal Audiomatch P500CE power supply is a switching type, so there are no heavy transformers and bulky smoothing capacitors to worry about, and it will accommodate mains supplies from 90-260V without any modifications. In addition to the eight speaker-output terminals, the processor also provides line-level outputs for up to two optional NHT S80 subwoofers, and the appropriate filtering characteristics can be recalled to the processor. In fact, the unit incorporates four different user-selectable crossover filter sets, and others may become available for uploading via the USB interface.

The processor/amp unit is equipped with both unbalanced phono and balanced XLR inputs for each channel, and they are wired in parallel. No input-sensitivity figures are provided, but I didn't have any problems hooking the unit up to either balanced console monitor outputs (nominal +4dBu) or unbalanced consumer preamp outputs. Needless to say, the analogue inputs are converted to digital, processed in the DSP crossover, and converted back to analogue for the amplifiers and outputs. The specifications give no information concerning the converters, but it would seem reasonable to assume a base sample rate is employed. A pair of XLR outputs (with associated unbalanced phono sockets) are provided to drive active (powered) subwoofers, and a slide switch above the XLRs determines whether the output is feeding one or two units.

The mains power switch is built into the IEC mains inlet, and there is a remote powering control facility with

SOUND ON SOUND

NHT Pro M60 & XDA £3779

pros

- Highly optimised crossover filters.
- Accurate time alignment.
- Very efficient Class-D bi-amplification.
- Stylish amp/crossover unit.

cons

- Technically perfect isn't always subjectively right!
- Would benefit from a subwoofer in mastering applications.
- Weird cabinet design aesthetics might not be to everyone's taste.

summary

A very competent monitoring speaker with bi-amped Class-D amplification and digital crossovers. Future upgrades promise to offer system in-room measurement and alignment, as well as alternative crossover algorithms.

a group of four terminals. Two accept a standard external 12V DC trigger signal input, while the other two provide a trigger output for other equipment. An adjacent slide switch sets the trigger mode to be permanently on (On), to power up when an external 12V DC signal is present on the interface (Ext), or to power up when the input audio signal exceeds a very low threshold (Audio). This facility certainly makes it easier to operate a multi-channel surround-sound system.

There are also some other rear-panel facilities not mentioned in the handbook. A coaxial socket next to the Trigger input is labelled Accessory Power. I presume this to be a DC power outlet for some sort of future optional add-on unit. A single XLR input labelled Mic is presumable for a test alignment mic so that in-room correction can be carried out by the DSP processing. Finally, there is the B-type USB 2 socket which is mentioned in the context of enabling future firmware upgrades. However, I imagine it may also in the future offer remote control and real-time system alignment from a computer too, although the handbook makes no mention of this.

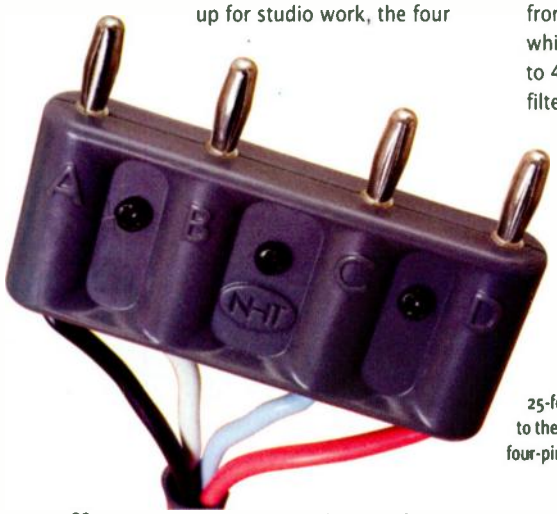
Alternatives

The M60 is an unusual design in many ways, making it difficult to find an alternative model that stands up for direct comparison. At roughly the same budgetary level are a lot of far more traditional but extremely capable designs. Favourite amongst those for me would be the ATC SCM20A Pro, or the PMC TB2A — both excellent two-way self-powered designs. However, the DSP crossover approach is increasingly common these days, with such contenders as the Dynaudio Air Series, as well as offerings from Genelec and Tannoy.

Operation

The front panel is very *Star Trek: The Next Generation*, with hidden-until-illuminated indicators on a dark glass panel. A single button for each channel cycles through the various filter sets, with simple illuminated graphics representing the mode. When set up for studio work, the four filter options are a gentle roll-off below 70Hz (indicated by the fully shaded ellipse); a steep roll-off below 45Hz (half-filled ellipse); a steep cutoff below 45Hz combined with a 1dB dip between 2kHz and 6kHz (quarter-filled ellipse); and a steep cutoff below 45Hz combined with a 1dB shelf above 2kHz (square box).

Once set, the filter-selection options are automatically disabled, and the front-panel lights are extinguished after a short time period. To regain access, both channels' selector buttons have to be pressed and held, after which the indicators illuminate and filter changes are allowed. Three further hidden-until-lit icons indicate when the unit is in standby mode, when the unit is operational, and when there is a fault condition.



25-foot cables are supplied to connect each speaker to the amplifier, and they are terminated in a special four-pin banana plug to make setting up easy.

Listening Tests

I experimented with a variety of speaker positionings and DEQX settings. In the end I favoured a freefield placement, well away from walls, and the second filter setting, which allowed the monitors to run flat down to 45Hz. I played with the two additional filter settings with mid-range and treble tweaks, but preferred the flat version overall. However, I have to say that I found the general balance a little forward and shouty compared with my PMC references. While this voicing certainly has the benefit of making it slightly easier to 'see' into mixes, I found it also a little wearing over extended listening.

The bottom end is well defined, clear, and tuneful. As this is a sealed-cabinet design, the bass extends lower than the cabinet size might suggest, and it is free of the port resonances that afflict so many monitors. With such small bass drivers, it is not surprising that the M60s don't move huge amounts of air, and when cranked up they can feel a little lightweight, although the deep notes are actually there. Fortunately, this problem can be addressed easily enough with a decent subwoofer.

Imaging seemed a little disappointing to me. The speakers proved able to produce impressively wide and stable images that often stretched beyond the physical spacing of the cabinets, but depth cues weren't portrayed as well as they are by some other monitors. There was some sense of depth, but it was not as deep and lifelike as I am used to. I suspect this is related to the ruler-flat frequency response of each speaker — gentle response tailoring in the mid-band affects this psychoacoustic aspect dramatically.

Overall, the M60XD is an impressive system. The amplification is competent and capable, the digital crossover is well designed and optimised to the drive units, and the overall presentation is to a very high standard indeed. These monitors are detailed and revealing, and are well suited to mastering and critical mixing applications. The voicing may not be to everyone's taste, but the future release of system alignment software will enable users to measure the system's in-room response and optimise the voicing as they see fit. I still can't quite come to terms with the shape of the cabinets, though... SOS

information

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Native Instruments Kore

Hardware Controller & Software Host For Instruments & Effects

Photos: Mark Ewing

NI's Kore system promises to unify your library of software instruments and effects, creating the mother of all workstation synths or multi-effects units.

Simon Price

Native Instruments' Kore is an integrated software and hardware system that acts as a universal 'front end' to other VST/Audio Units instruments and effects installed on your computer. Plug-ins are hosted within the *Kore* software 'shell', which appears as a plug-in in your sequencer, or runs as a stand-alone host. *Kore* plug-in instances, or the stand-alone host, are controlled with the hardware unit. In addition to providing a standardised hardware control system, and a means to construct patches from multiple plug-ins, Kore has a powerful patch and preset cataloguing system, creating a centralised database of all the sounds and effects at your disposal. This rather dry description may not blow your skirt up, but believe me it's this last feature that could have a profound impact on your desktop music production, and bring your computer much closer to feeling like a musical instrument.

Koncept

Kore's announcement was greeted with a cautious reaction in some quarters, and an excited one in others, probably because of the lack of agreement on what it was actually going to do. Let's start by taking a quick look at what it doesn't do. Kore is not a sequencing or recording package; it is more like an instrument that draws its sound sources and effects from other plug-ins. It does not provide any DSP hardware: your plug-ins still run using the host computer's processing power. The controller is not a generic MIDI control surface, and is purely meant to control the *Kore* software and plug-ins hosted in it. There is, however, a MIDI interface built into the controller allowing MIDI controllers and keyboards to be connected to the computer via the same USB cable. A two-in, four-out, 24-bit/96kHz audio interface is also built in.

Like many innovative new products (the iPod is a good example) Kore addresses needs that you may not have been fully aware you had. However, anyone who predominantly uses a computer to make music will have come across the same frustration: you have a number of soft

synths, samplers, drum machines, effects and sequencer/recording packages, and it can be a pain trying to integrate these into some kind of workflow. It's no wonder that keyboard workstations are still popular, because they allow you to compose in one place, with all your sounds together and controlled with the same knobs, laid out in the same way. Workstation sound sources are blended, have appropriate effects, and are available as presets. Native Instruments probably felt particularly close to the problems of computer-based composition and performance, as their flagship *Komplete 3* package installs no fewer than 13 separate instruments and effects. Not only does this mean 13 different user interfaces, but also 13 different places to look for the sound you want. If you want a warm pad, you could use *Reaktor 5*, *FM7*, *Pro 53*, *Kontakt 2*, *Kompakt* and more. In terms of hardware control, you would need to spend endless hours setting up templates to control all these instruments and effects from a traditional MIDI device. Kore confronts all these challenges in an attempt to combine the advantages of computers with those of hardware synths and effects.

One of the core ideas behind Kore is that you can choose patches on the basis of what they sound like, regardless of what instrument(s) made them.

As well as being close to the problem, Native also had the solution right under their nose with Guitar Rig. If you haven't seen it, Guitar Rig is a computer-based effects unit that uses a software host controlled by a hardware foot controller. The host software lets you build patches from a suite of effects modules in a virtual rack, then assign the hardware controls to any parameters on the modules. This patch is then used stand-alone, or inserted as a plug-in into any recording/sequencing package's mixer. Kore works in exactly the same way, except that instead of building patches from a selection of internal modules, Kore patches (called Kore Sounds) are constructed from other VST or AU plug-ins. NI realised they could integrate their entire *Komplete 3* package into a single environment, and third-party plug-ins could come too. What's more, this environment (the *Kore* plug-in) can be used in any software that supports VST, Audio Units, RTAS, or DXi plug-ins, creating a totally portable patch format. And yes, that does mean you can run VST and AU plug-ins in *Pro Tools*, VSTs in *Logic*, and so on. The final — and perhaps ultimately most significant — step was to add a patch library system that classifies sounds based on their sonic attributes instead of which plug-in made them.

Getting Started

My Kore came bundled with the *Komplete 3* package, which required installing first. This is because the Kore installer updates several of the plug-ins and libraries included with *Komplete* in order for them to work properly with Kore and the Kore Sound library. It's probably fair to say that to get close to seeing the full potential of Kore you need *Komplete*, or at least several NI plug-ins. As you'll see, much of the power of the system comes from the Kore Sound library, and the more NI plug-ins you have, the more pre-programmed sounds you can access out of the box.

The controller connects to the computer via a USB2 cable, which handles bi-directional communication of control and display data, and connects the integral MIDI and audio interfaces. The unit also takes its power from the USB port. In order to launch the *Kore* application (or insert a *Kore* plug-in) the controller must be connected. Although the software side of Kore would still be useful without the hardware, NI stress that the integration between the components is key. This is also, of course, an effective way of copy-protecting the software, and who can blame them for that? Firing up the stand-alone *Kore* application wakes the surprisingly small controller (it's 30cm wide) revealing its bright red display and button lights. You are presented with an empty rack-style or mixer-style display (depending on the selected view option), with a master parameters area above, and the sound browser below. The application has the standard NI Audio and MIDI Settings page for choosing a soundcard and MIDI connections. Although the controller has built-in audio and MIDI interfaces, you don't have to use these. I opted to use Kore's MIDI input for my keyboard, and use my Digidesign M Box for sound output, which ▶

SOUND ON SOUND

Native Instruments Kore £370

pros

- Excellent, highly programmable controller with endless encoders.
- Almost total control of software from the hardware.
- Bi-directional controller communication allows you to see values, button states, and assignments on the hardware.
- Simple, yet ingenious sound-based patch browsing.
- Provides a unified front end for the hugely powerful, but fragmented, *Komplete 3* suite.
- Huge potential for creating complex sound patches from multiple plug-ins.
- Lets you use VST and AU plug-ins in any sequencing/audio package.
- Great live tool.
- Doubles as MIDI and audio interface, with separate main and cue outputs.
- Great value.

cons

- Not all features available in first release.
- Library doesn't yet take full advantage of Kore's sound-design capabilities.
- You really need a few NI plug-ins to get the full experience.
- Benefits of running plug-ins in *Kore* in a host sequencer come at the expense of some CPU overhead.
- Live features not available when running as a plug-in.
- Proprietary instruments bundled with most of the big-name sequencer packages are not supported.

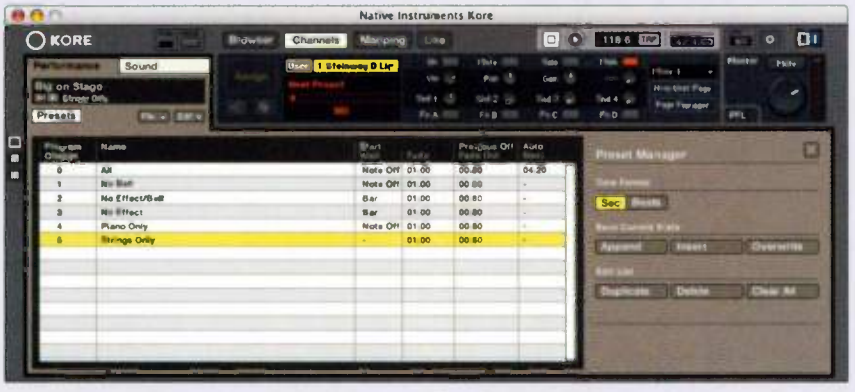
summary

A bold and ground-breaking product that solves several of the problems of computer-based music composition and performance in one stroke. A classy product from version 1.0, and bound to keep getting better. Grab one bundled with *Komplete 3* while the offers last!

Kore On Stage

The combination of multi-instrument hosting with flexible controller assignments means that many people are looking to Kore as a live system. Several Performance examples are provided to show Kore runninng as a keyboard rig, an electronic music and beats station, and a DJ tool. Used in stand-alone mode, you can set up a Performance with multiple Sounds, some of which may be drum machine or loop-based, and which can be triggered by the clock. Each channel in Kore also has a MIDI file player, allowing sequences to be played back, and even strung into patterns.

Kore has several features specifically aimed at live use: the Song List, Performance Presets, and the Live View. The Song List is a folder in your Kore directory that contains links to any Performances that you wish to use live. This list is accessible from the hardware controller, allowing you to load up Performances without resorting to the mouse/trackpad. However, with a little clever use of the Preset functionality, you can reduce or possibly even remove the need to load new Performances. Presets are snapshots of the configuration of a Performance (see screen below). They store the state of all panel controls including channel on/off switches, levels, mutes, sends, pans and effect controls, but not the fundamental architecture such as routing, or which channels are present. A great feature of Presets is that you can set up fades so that the Presets morph into each other. You can also specify automatically sequenced Preset changes, which you can use to prepare small, automated pattern changes, or even sequence whole tracks or sets. In the current release you can only set times in minutes and seconds, although you will soon be able to use bars and beats. Presets can be recalled from the controller, which you can arrange so that the Control button toggles between Presets and controller views. You can also change Presets using program changes, which works perfectly, although you have to remember to filter these messages so they don't get passed on to the plug-ins that *Kore* is hosting —— otherwise they all change their patches too! What makes Presets so useful is the ability to switch channels off instead of just muting them. This frees up the CPU power they were using, so you could build a large Preset with everything you need for a show, and use Presets to activate just the channels you need at any time. This means never having to stop and load new Performances. Finally, Live View (top) switches *Kore* to a simplified full-screen display, which only shows the most important things you would need when playing live.



An important consideration is that, because there is no Performance layer when *Kore* is used as a plug-in, these live tools (Song List, Presets and Live View) are only available in stand-alone mode. This could be a problem if you tend to use Ableton *Live* for live work. One option would be to use multiple *Kore* plug-ins instead of Presets, and use the *Live* mixer to handle the transitions, although this would lose the crossfading and sequencing. Personally, I would run both *Live* and the stand-alone version of *Kore*, sending MIDI Beat Clock from *Live* to *Kore*. I was able to set up an internal MIDI port between the two, and play *Kore* devices from MIDI sequences in *Live*, but was not able to run Beat Clock as this feature was not available in the initial release. However, by the time you read this, an update should have added the all-important External Sync button to *Kore*. Rewire would come in handy in this configuration so you could run *Kore*'s outputs back into *Live*. However, I found you can mix both *Live* and *Kore*'s outputs through the Kore controller audio outs. You can also send your *Live* cue mix feed out of the Kore controller's headphone outs, as well as pre-fade listen signals from *Kore*. Just to really push the boat out I ran *Reason* Rewired into *Live* at the same time, making for a seriously powerful live rig. Now, if Propellerhead would just release a plug-in version of *Reason* that I could run in *Kore*...

▶ all worked without a hitch. The stand-alone *Kore* application does not currently support Rewire, although I think this is on the cards. This would provide useful flexibility of operation, as *Kore* works differently in stand-alone and plug-in modes.

Thinking In Sounds

There are several ways to open up sounds and instruments in *Kore*, but I started with the main Kore Sound browser. NI describe one of the main concepts of Kore as 'thinking in sounds' and this patch browser is what they are talking about. As you can see in the screenshot on the previous page, the Sounds browser presents five columns with families of words to describe sounds, which, in Kore-speak, are called Attributes. Above this are options for choosing between Instruments and Effects patches (with different lists of attributes for each) and a text search. In the picture I'm searching for an Instrument patch. The first column lets you choose the kind of instrument you are looking for — piano, synth, and so on — while the second, Source, describes the method used to generate the sound. These attributes are clumped into mutually exclusive pairings or groups, such as Synthetic or Sample-based, Analogue or Digital. Further down this column are attributes that describe other properties of the patch, such as whether it is a Solo or

Ensemble/Kit instrument, Layered, Looped, and so on. In all but the first column you can select more than one attribute, so you could choose a Synth that is Digital and Layered for example.

The third column lets you choose the character of the sound, and is populated with words like Warm, Dissonant, Bright, and so on. Column four is Articulation, and describes the way the sound is played (for example, Chord or Arpeggiated), the way the sound evolves (Slow Attack, Pulsating) and more traditional patch categorisations (Pad, Lead). Finally, the fifth and most subjective column is Genre, with listings such as Film Music, Techno/Electro, Rock and Pop. Categorising sounds by a musical genre is of course problematic, although useful at times, but you don't have to choose attributes in every column to use the browser. Each time you add or remove attributes to your search, the list of Kore Sounds to the right updates to list the patches that have been tagged with all your selected items. Some useful preferences alter the way the browser displays this information, such as allowing you to see the number of patches in each category, and to hide attributes that don't appear in combination with those you've already selected. You can also reorder the patch list alphabetically by any of the columns (Vendor, name, and so on) and add or remove columns by right-clicking.

In the example I've opted for the Kore Sound called Velvet Pads, which, as is indicated in the Plug-ins column, is made with *FM7*, *Kontakt 2* and *Reaktor 5*. I've deliberately chosen a complex patch as it shows up several features of Kore. The majority of patches in the library are made with single instruments (for more about this, and other aspects of the Kore Sound library, see the 'Inside The Library' box). Double-clicking the patch opens it up into the first spare slot in the rack, and makes it immediately available for playing via MIDI, and controlling with the hardware.

Kore channels can be displayed horizontally or vertically.

Sounds And Performances

The screen above, top, shows the Kore Sound loaded into the rack, with the Browser closed to save some space. To the left of the rack are three buttons that alter the display of channels. Instead of this sideways view, you can display channels vertically like mixer channels (as in the screen above, for example) or a combination of the two. The channel represents the entire Velvet Pads Kore Sound, and appears as a generic device that gives no clue to what plug-ins are generating this sound. The channel has

Kore's Mapping functions allow you to split and layer multiple sounds for simultaneous control from one controller.

▶ an on/off switch, input/output assignments and gain controls, a MIDI File player section, four insert slots, and a central area showing assigned knobs and buttons. The master section can be switched to reveal four send controls, which are always visible in channel strip view (while the controller assignments are not). Right-clicking an insert slot lets you add or replace *Kore* effects.

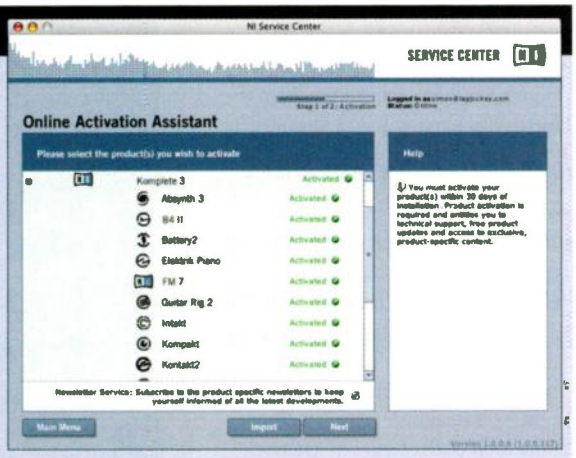
Double-clicking another Kore Sound in the Sounds browser will open up a new channel in the rack. The parameters currently being controlled by the hardware appear in the main section at the top of the window (more on this soon). By default, the keyboard will play both channels, but by opening the Mapping window, you can split keyboard ranges, map velocity ranges, and assign different MIDI channels to each Sound, which is obviously particularly useful for live work. As well as adding insert effects to any channel, you can create Group (submix) channels and buss several Sound channels to them. You can also add Send (effects return) channels and share reverbs, delays, and so on. Each Sound Channel, Group, Send and Effect has its own hardware controller assignments, often spread across two or more pages each. The device and page which the controller is addressing can be selected either from the controller itself, or on screen. Importantly, you can create extra User Pages, and assign parameters from different devices and channels to the same page, allowing you to control several channels with the controller at once without changing pages. This entire configuration of Sound channels, sends, groups, inserts and controller assignments is called a Performance. And it's only half the story.

Double-clicking the Kore icon in the middle of the Velvet Pads channel brings up the display below. You are now looking at how the Velvet Pad Kore Sound is constructed (note that the main display strip indicates you are now viewing a Sound instead of a Performance). The Sound has its own internal rack, which can accommodate individual instrument plug-ins. In this example there is one *FM7* and one *Kompakt 2*. The third channel is a Group channel, where both plug-ins are being mixed in order to go through a filter plug-in. Notice that the instruments also have their own FX insert slots too. A Kore Sound has its own routing, effects, sends and key/velocity maps, all of which collapse to a single device when viewed from the Performance layer. Channels at the Sound level can use any VST or AU plug-in in the FX inserts, as well as the built-in *Kore* effects. There are also different controller assignment pages at the Sound level, although some special pages called Easy Access (EA) pages appear in the Performance too. This dual-layer, 'nested' nature of Kore's environment makes very flexible configurations possible and easier to manage. It also contributes to Kore's fairly steep learning curve, and causes some confusion early on! You can move between the Performance and Sound ▶

The Sound layer displays all the elements that make up a Kore Sound.

NI Service Centre

The *Kore* installer includes the new NI Service Centre application, which comes as standard with all NI software from May 2006. This small application will be a welcome addition for all NI users, as it greatly simplifies the registration and activation of software. This was previously a fairly complicated and time-consuming procedure carried out at the NI web site. Now, Service Centre stores all your details, and connects to NI's server in the background. The utility scans for all installed products, and lets you activate or deactivate them in bulk. Updates are also taken care of by Service Centre, another major bonus as keeping a computer full of NI software up to date has always taken quite a bit of time and effort.



Test Spec

- *Kore* software version 1.01.
- Dual 2.7GHz Mac G5 with 1.5GB RAM and Digidesign M Box interface.
- Tested with NI *Komplete 3*, Korg *Legacy Collection*, Digidesign *Pro Tools LE* 7.1 and Ableton *Live* 5.2.

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layer from the hardware controller, although it takes some practice to be confident about where you are, and you will want to simplify things with User pages when playing live. An important footnote to this section is that when you use *Kore* as a plug-in in a sequencer, the Performance layer does not exist. In a host you are always looking at the Sound layer, so when you open a Kore Sound you see all its separate components. This is because you would tend to use the host's mixer environment to open multiple Kore Sounds instead of *Kore*'s Performance mixer.

Kore Kontrol

Kore's hardware controller is a sturdy wedge of metal that, if dropped, would probably come off considerably better than whatever it landed on. A strip of glossy black plastic across the middle houses eight rotary encoders, the display, cursor buttons, a data wheel, and several mode buttons. The cursor and mode buttons are large and inset into the surface, and their size means that they rock about a bit and occasionally fail to engage unless you press them in the middle. The demo unit had small, rubber, glued-on feet which all fell off during the review, but I'm assured that the more up-to-date production runs have larger, securely affixed feet. The rest is all good news. The all-important knobs feel great, moving with a smooth, light action. The touch-sensitivity is immediately apparent, as red lights ringing the bottom of each knob light up while you touch the knob. Whenever you are touching a knob, the display becomes dedicated to that parameter, showing the parameter's full name (all parameters can be given long and short names for different display purposes). Values are displayed in real units (ms, dB) when provided by the plug-in, or otherwise as a percentage, a value between –100 and +100, or between 0 and 1. A large horizontal display like a progress bar shows the position of the knob within its range. Cleverly, if you are using more than one knob at a time, the display changes to show an overview of all knobs and their current positions within their ranges. I'd have liked the option to show this display even when you are only using one knob, because when you are holding one knob you can no longer see what the other assignments are until you let go, or touch another knob.

Each channel within a Kore Sound can make use of any effects available on your machine, including *Kore*'s own.

The knobs are endless encoders as opposed to normal pots, so they don't have end stops. Parameters don't 'jump' to the position of the knob when moved, they pick up smoothly from the current value. What's more, there is an ingenious option in the Assignment Properties screen (see screenshot on next page) that lets you set the sensitivity. For example, mapping the parameter across 360 degrees sets the assignment so that one turn of the knob moves the parameter through its entire range. This relationship can be set between 30 and 3600 degrees, so a parameter can be set so that it takes 10 turns of the knob to move through the whole range, providing ridiculously fine tuning. The knobs are high-resolution (ie. not MIDI!), with 500 steps per revolution, which, combined with the sensitivity option, means it's not unusual to see the display showing values with six decimal places. Finally, using an approach that's similar to the assignment system in *Reason*'s Combinator, you can specify the maximum and minimum values that an assigned knob will act between, and you can create reversed polarity assignments. You can also set multiple assignments for the same knob. Put all these ideas together and you could, for example, have a knob dramatically increasing cutoff frequency, while subtly reducing gain to compensate. Assigning a control and range is easy: click the assignment and range buttons, select a knob, then turn the target parameter through the desired range. In short, the knobs, and assignment flexibility, are better ▶

The Kore controller's rotary knobs are infinite encoders, which always reflect the position of the assigned parameter.

Inside The Library

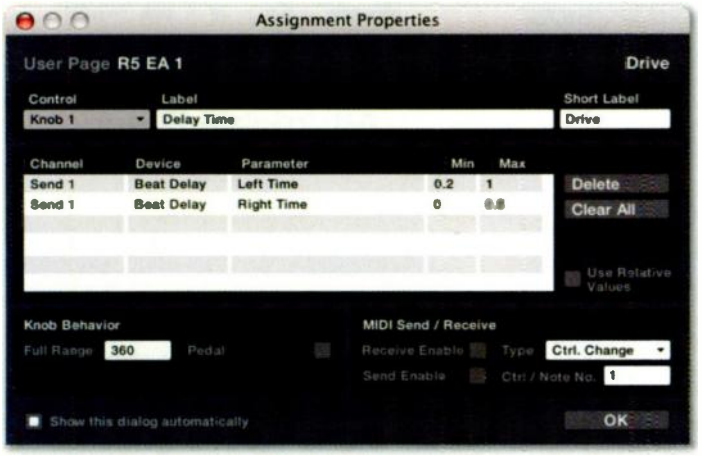
The Kore Sound library, as installed, consists exclusively of patches made with Native Instruments instrument and effects plug-in, and *Kore*'s built-in effects. When you first run *Kore*, it scans your system for all installed VST and AU plug-ins, which takes a few minutes. The Sounds browser then only displays patches that are made from plug-ins that you actually have on the system. Third-party plug-ins are also detected and listed in the Plug-ins browser, and can be used freely in *Kore*, but with no NI-created patches. This is partly due to licensing issues, and partly because it would obviously be impractical for NI to create Kore Sounds for every possible combination of plug-ins on a user's system. This is not to say that third-party plug-ins won't be supported by the controller, though: they are, and you can load and edit the plug-ins' own presets. Also, the installed library is only a starting point. You can save your own patches as Kore Sounds, and add Attribute tags so that they will be found at a later time. No doubt users, and maybe some manufacturers, will publish Kore Sound versions of the preset libraries of most popular plug-ins. You will probably also see many layered and processed combo patches and Performances appearing for download on the Web.

Other useful data can be added by the user (such as a Rating which you can list search results by), and there is also data generated by *Kore*, like the number of times you've used the patch, and the Load Time and CPU overhead on your computer.

If you have *Komplete 3* installed, the *Kore* library contains over 9000 Kore Sounds, of which about 7000 are instruments and 2000 are effects. These basically fall into two categories: basic, single-device patches that open up a factory preset for a *Komplete* plug-in, and more complex patches that take advantage of *Kore*'s layering, routing and effects. The vast majority of patches are of the first kind, which make all the presets in the *Komplete* bundle accessible to the Attributes library, and add Easy Access controller assignment pages.

At this point there are only 200 patches (plus 12 Performance examples) that really use the sound-design features of *Kore* such as layers, splits, insert and send effects, groups and stacked controller assignments. This is a bit disappointing, because there's so much more spectacular stuff that could be done with this huge armoury of plug-ins. However, it's early days, and I'm sure we'll see many more updates with more sounds. Up to the launch, Native Instruments have focused on getting the thousands of basic presets from the *Komplete* plug-ins running in Kore Sound, and most of these presets use effects within the plug-ins themselves, and sound good on their own. It could have been tempting to stick *Kore* effects or *Reaktor* effects on most patches, but this would be hugely wasteful in DSP resources. One thing that is true, though, is that until we see some Kore Sound libraries for third-party plug-ins, you'll be missing one of the best aspects of Kore if you aren't running it with NI plug-ins.

By clicking the Attributes button when viewing a Sound in *Kore*, you can see all the metadata that is stored with the patch, or will be stored if you save out as a Kore Sound (.ksd) file and add it to the library. The screen above shows the Velvet Pads patch's Attributes page. All the sound's characteristics have been selected, determining which searches will result in a 'hit' for this patch. Further information, such as the author, comments and which plug-ins are used in the patch, is listed to the right.



The Assignment screen allows you to set the 'gearing' of the Kore controller's knobs compared to the parameters they affect.

▶ than anything I've seen before.

Two main display modes are toggled between with the View button. One shows only the knob assignments with a miniature value display. The other shows all knob and button assignments, without the values. The buttons all have integral lights, which are unlit on unassigned buttons. Assigned buttons are lit brightly when the parameter they control is in the nominal On position, and are dim when Off. This is the kind of invaluable benefit of having bi-directional control between hardware and software, because when you switch pages you can instantly see the state of the parameters. Button assignments can be set to toggle a value at each press, or only act while they are held down. They can also be inverted. In another parallel with *Reason*'s Combinator, buttons can be assigned to change a continuous parameter (knob, fader, and so on) between two values, but they can't be set to step through multiple values. As well as the main eight controller buttons, the Kore controller has transport buttons that are used to start and stop the internal clock in the *Kore* software. These can't be used for your sequencer.

Real Life And The Kore Controller

It's obviously great to have hands-on control, but all too often, hardware control systems end up being more frustrating than they're worth because it's too difficult to follow what is mapped to what, and you get lost when trying to move between different groups of target parameters. Often controllers that look great on paper are relegated to writing the odd automation pass, or for use with a single page of assignments in a live setup. Kore has some distinct advantages up its sleeve, however. The first is that it hosts all your plug-ins in a standardised environment that has the same layout on screen as the knobs and buttons on the controller. NI have also tried to make their factory assignments conform to a standard model. Every Kore Sound starts with two Easy Access pages at the Performance level with volume, cutoff, resonance, tuning, and so on, in the same locations. Each channel has the same Mixer page with levels and sends, followed by pages for any effects. At the individual component (Sound) level, each plug-in starts with a mixer page, then two Easy Access pages, then pages specific to that plug-in. Finally, having a display with all the assignments in front of you, while not unique, is only usually seen on devices costing three times as much or more.

However, even with Kore you can't get away from the fact that you have to learn how to change pages to address more than eight knob and button assignments. Clearly, a lot of thought has gone into trying to make this as painless as possible, and NI have done quite well here. The truth is, though, that you'll need to put in a fair bit of practice to get fluent with moving around. Crucially, you need to understand thoroughly how Kore works, and be familiar with the patches you are working with.

I went through a painful learning curve with this (the manual is not brilliant, but makes sense if you try first, fail, then look at the manual). However, you do soon reach a breakthrough point. The frustration is put into perspective when you remember what it's like learning to get around in a new sequencer package, or learning the complexities of a powerful synth workstation. It does require some patience, and some determination to get away from the comfort zone of the mouse.

Here's how the navigation system works: when you're running *Kore* stand-alone, you start off in Performance mode. The left and right arrow buttons move the controller between the channels in your performance, plus a User page section at the leftmost 'position'. The up and down arrows (or data wheel) move you between the pages for the current channel (or the User pages). Pressing the Sound button switches you to the Sound layer, with access to the separate devices that make up each Performance channel. The same principle now applies, with the left and right buttons moving between the user page area, and the channels in the Sound. Up and down, or the data wheel, again step between the various pages in each channel.

Kore The Plug-In

Unless you are lucky enough to be able to devote a separate computer to it, most studio-based work will involve running *Kore* as a plug-in in one of the main audio and sequencer applications. During testing, I mainly ran *Kore* in *Pro Tools* as an RTAS plug-in, and in *Live* as an Audio Units plug-in. I also tested in *Logic Express*. One of the beauties of *Kore* is that it doesn't matter what host you run it in — all the same Kore ▶

One of the joys of Kore is that whatever host you run it in, all your VST and Audio Units plug-ins are available.

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▶ Sounds work, and you can still load up other VST and Audio Units plug-ins. In *Logic* and *Pro Tools* this gives you access to plug-in formats you couldn't normally use. However, *Kore* can't load RTAS plug-ins due to Digidesign's licensing rules, so you can't host the plug-ins that come with *Pro Tools*, or Digidesign's RTAS instruments like *Xpand!* or *Hybrid*. Similarly, *Logic*'s and *Live*'s bundled plug-ins are unavailable to *Kore*.

When running in a host, *Kore* only has one layer (the Sound layer) so the controller works slightly differently. Instead of toggling between the Sound and Performance layers, the controller's Sound button brings up a list of *Kore* instances in your mixer. Selecting an instance and pressing Enter switches the controller to address that instance. The lack of two layers certainly makes *Kore* easier to use in a host, but does slightly limit how easy it is to set up complex patches. This is because it's not currently possible to layer two Kore Sounds outside of a Performance (although this may change). If you load one sound from the Sounds browser, then try to load another, it replaces the first one. You can, however load additional instrument plug-ins into blank channels and layer these on top of the Kore Sound. You could also, of course, set up the same layering by using two tracks in the host application's mixer. Although the Kore Sound browser is one of the advantages of *Kore*, you don't have to use it. You can load plug-ins straight into *Kore*, and use them with the controller. The controller will map to automatable parameters on most plug-ins, and some third-party instruments have already had Easy Access pages added (like the Korg *Legacy* package). You can load a plug-in's presets from the *Kore* channel, or you can open up the plug-in's own interface. One problem, however, is that while the plug-in is open, you are locked out from doing anything else, so can't leave plug-in windows open. Another problem, visible in the screenshot on the previous page, is that any plug-in hosted by *Kore* appears as 'Kore' in the host application's mixer.

A further consideration when deciding to using *Kore* instead of just inserting the raw plug-ins is that there is some CPU overhead. *Kore* doesn't use much CPU power, but it adds up when you use several instances. I did some tests in *Live* and *Pro Tools* to see what the difference was. I set up a six-note chord MIDI sequence in *Pro Tools*, over a four-bar loop. I then inserted a *Pro 53* plug-in on several stereo instrument tracks, all playing the same patch, and kept adding tracks until I reached the point where *Pro Tools* couldn't play back reliably without CPU overload errors. With raw *Pro 53*s I got to 19 tracks (with a 512-sample buffer, which is about 12ms output latency). With *Kore* running AU instances of the same *Pro 53* patch I could run 14 instances. *Pro Tools* managed 15 *Kore* plug-ins when running VST versions of *Pro 53*.

I then tried running multiple instances of a complex *Guitar Rig 2* effect, with a drum loop bussed through each instance. *Pro Tools* managed 12 RTAS instances on its own, while running within *Kore*s it overloaded at seven. For comparison, I ran the *Pro 53* test in Ableton *Live*, with the same buffer settings and MIDI sequence, and running with the Core Audio driver via the Digidesign M Box. I got to 14 instances before slight popping crept in. Running via *Kore*, the *Pro 53*s started crackling at eight instances. The conclusion is that *Kore* does have a noticable impact on you CPU load, and on larger projects you'll probably want to ration your use of it for tracks where you want to use the controller or the browser. Native have said that there is still room for optimisation of *Kore*, with future updates promising performance increases.

The Kore controller is a USB2 device, and offers audio and MIDI I/O as well as its control functions.

Conclusions

Kore is a difficult product to sum up succinctly, because it does so many different things and covers a lot of new ground. There's really nothing to compare it to! There are a number of areas where Kore could make a considerable difference. I can see film or TV composers knocking out hours of music simply by loading up a few complex patches based on sonic descriptions. If you're a fan of Native Instruments' heavyweights like *Reaktor* or *Kontakt* but can't get to grips with using them, this is the front end you need. If you want to play live, or you're frustrated by the hands-on control of instruments offered by most MIDI controllers, you're unlikely to find a better option. Above all, the simple idea of Kore Sounds, and browsing for sounds instead of instruments, makes you approach music-making differently. It reveals whole areas of your sonic arsenal that you didn't even know were there. For example, *Kompakt* gets to stand up as a great tool when it's usually overshadowed in the *Komplete* bundle, and *Reaktor* becomes a much more versatile instrument. It's this creative angle that really gets you into Kore, and keeps you going through the learning process. Native Instruments have realised that what the world doesn't need is another new soft synth, but a way to make them behave more like musical instruments. Their solution, Kore, is a strong step in the right direction, and could well be a landmark. I hope it will continue to develop as it gets used and abused in various environments. It's certainly renewed my enthusiam for making computer-based music by putting me back more into the role of musician, and less that of a sound engineer or programmer. After spending some time with Kore, I'm developing that familiar feeling that came with iPods and mobile phones. The feeling you get from a product that you didn't really know you needed, but can't imagine how you got on without it. SOS

information

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Classic Tracks

Bryan Adams 'Run to You'

Artist: Bryan Adams
Track: 'Run to You'
Label: A&M
Released: 1984
Producers: Bryan Adams, Bob Clearmountain
Engineers: Bob Clearmountain
Studios: Little Mountain, Power Station

The *Reckless* album was a huge success for Bryan Adams, giving rise to six hit singles — but the first one, 'Run To You', was almost never even recorded.

Richard Buskin

One of the most recognisable names among the studio elite of the past 25 years, Bob Clearmountain has certainly earned his stripes as a producer and engineer. Indeed, since the 1980s, 'Mixed by Bob Clearmountain' has been an industry catchphrase. Suffice it to say, it would almost be easier to list the major artists whose records he hasn't worked on, such is the veritable Who's Who of his track record.

Influenced by his guitar-playing older brother, Clearmountain began playing bass as a teenager. However, it was his fascination with recording technology that led him to apply for a job at New York's Media Sound after a band that he played with had cut a demo there. The year was 1972, and although, with much persistence, he was initially hired as a delivery boy, after just a couple of deliveries he found himself assisting on a session for Duke Ellington. Not a bad start, and one that quickly led to several of the aforementioned engineering assignments, as well as productions during the second half of the decade for artists such as The Rezillos, Billy Cobham and Narada Michael Walden.

Nevertheless, while bigger things ensued at the start of the Eighties courtesy of projects with the Stones, Roxy Music, Bowie and Huey Lewis, 1984 was arguably Bob Clearmountain's halcyon year. Not only did he produce and engineer Hall & Oates' *Big Bam Boom* and mix Bruce Springsteen's *Born in the U.S.A.*, but he also co-produced and engineered Bryan Adams' smash-hit *Reckless* album, having previously fulfilled the same role on Adams' *You Want It, You Got It* (1981) and *Cuts Like a Knife* (1983).

Come Together

It was on the advice of A&M Records A&R exec David Kershenbaum that Adams initially hooked up with Clearmountain, and the latter duly helped the Vancouver native assemble a band in L.A. for the *You Want It, You Got It* sessions.

"He had been rehearsing with some musicians there and I guess he was really disappointed with their performance," Clearmountain explains.

"The Power Station had already been booked, and a couple of weeks before he said, 'Look, do you know any musicians? I've fired every one.' Well, I had worked with [drummer] Mickey Curry on a G.E. Smith album that I'd produced, and another guy named Brian Stanley who was the bass player, and then I also knew Tommy Mandel, a keyboard player who had worked with Ian Hunter, whom I'd previously engineered. So, I just called them all up, and that worked out really well for the *You Want It, You Got It* record. For the next album, Bryan brought in a different bass player, Dave Taylor from Vancouver, and these were the guys who also formed the rhythm section for the *Reckless* album."

Reckless captured Bryan Adams at the top of his game, even if it would subsequently be eclipsed in terms of sales by 1991's *Waking Up the Neighbors*. Boasting an idiosyncratic mid-Eighties sound and spirit, it spawned no less than six American Top 30 singles, and the first of these was 'Run To You', that paean to illicit love co-written by Adams and Jim Vallance, and largely built around a chorus that melds melodic hard rock with the singer's trademark raw-throated vocals. Still, while one of the track's distinguishing features is the obligatory heavy drum sound, this was crafted by Clearmountain in unconventional fashion amid fairly adverse conditions.

Simple Stuff...

While most of the *Reckless* overdubs would take place at New York's Power Station, where the album was also mixed, the basic tracks were recorded at Little Mountain in Vancouver, owned by Bruce Fairbairn and Bob Rock, where the setup included a Neve 8048 console, a Studer A80 and, according to Clearmountain, little else. Indeed, one of his fondest memories there is of assistant engineer Michael Fraser sitting cross-legged on the producer's desk in front of the patchbay and re-patching whenever there was a need for playback.

"That thing had an antiquated design, so it was a lot of work," Clearmountain recalls.

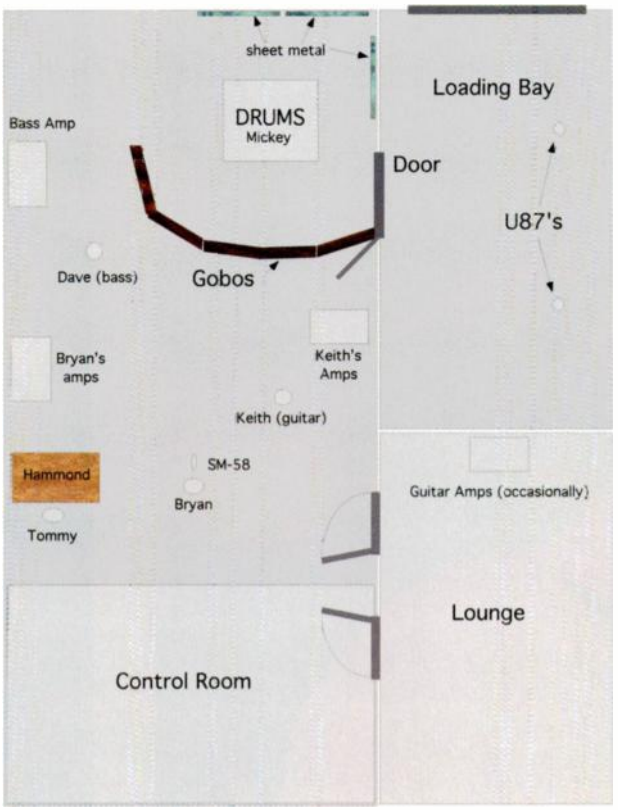
"I'd look at Mike like, 'I don't know what the hell you're doing,' but as long as I could hear what I wanted to hear, whatever he did was fine by me. He was amazing, and I'm not surprised that he went on to become a brilliant engineer. That studio was basically just a console and a tape recorder, which was also a problem because, ▶

Bryan Adams performing live around the the same period as *Reckless*.

RECORDING 'RUN TO YOU'

while the A80s were great-sounding machines, their motors were kind of under-powered for two-inch tape. I remember one song, maybe 'Summer of '69', where we had a bunch of edits — it came right at the end of the reel, and as it would hit the edits the tape would start to slow down a little bit.

"Although it was really well-known, Little Mountain was almost like a low-budget studio, with virtually no outboard gear, a smallish control room and these horrible speakers that were pretty much unusable. I can't remember what they were — I just listened to them one time and turned them off. The main room, meanwhile, was enormous — they recorded orchestras in there — but it was very dead. It was also used for jingles, and the walls were all thick with insulation and padding. If you had your eyes closed, you'd swear you were in a bedroom or a closet, but then you'd open your eyes and see this enormous high-ceilinged room.



The recording layout at Little Mountain.

"On the first day I thought, 'Man, how are we going to get a rock drum sound in here?' But then I walked around and found a door off to the side of the studio that led into a loading bay; a big concrete garage into which you could back a truck. They just used it for storage, they never really opened the garage door, but it had this incredible sound. I went in there and clapped my hands and said, 'Wow, can't we record the drums in here?' As it turned out, we decided it would be kind of awkward to have Mickey the drummer in a whole different room, so I set up the kit right in front of the door, got these gobos on which one side was a real hard wood surface, and made a big funnel-shaped device that focussed the sound through the door into the loading bay. I put a couple of room mics in there, and that's how we got our big rock drum sound.

"The funny thing is, someone apparently measured exactly how we'd set the drums up, and when Aerosmith's records and other rock records were done at Little Mountain they'd set everything up exactly the same way. So, if you listen to some of those Aerosmith records, the drums sound almost identical to the ones on the *Cuts Like a Knife* and *Reckless* albums."

To the best of Clearmountain's recollection, for the latter album Mickey Curry's kit was miked with an AKG D12 on the bass drum, Sennheiser 421s on the top and bottom of each tom-tom, AKG 451s on the hi-hat and cymbals, and a Shure SM57 on top of the snare with an AKG 451 on the rim in order to capture a little more attack. Room mics were Neumann U87s. It was, in essence, a straightforward setup that achieved an amazing sound, not least considering the environment in which the recording took place.

"Having such a dead-sounding room was quite an obstacle," Clearmountain remarks.

"Then I found these big 4' x 8' pieces of sheet metal out in the loading bay. I don't know what they were there for, but we put them up on the walls around where the drums were set up just to try to get some ambience. The drums were set up at a sort of right-angle to the door, and with the gobos in front of the kit the sound was bounced at a 45-degree angle into the loading bay. It was interesting, to say the least, and was further proof that you can pretty much achieve anything anywhere. I mean, I record drums now in my tiny little lounge, which is certainly not a studio, and that works really well, so you can work just about anyplace."

Altogether Now

Alongside the drums, the other musicians played together as a live rhythm section, scattered around the room.

"The guitar amps would always be way off to the side, because unfortunately that studio apparently had power mains that ran right down the middle of the floor," Clearmountain recalls.

"So, if you got a guitar amp anywhere near there, it would just hum like crazy... It really wasn't a terrific studio. But the room was big and a lot of people liked it. And it was also one of the only games in town at that time.

Uncertain Beginnings

As with all the Bryan Adams albums that Bob Clearmountain worked on, the MO was to rehearse for a couple of weeks before the start of recording. The material was all written beforehand and demo'd at the home of Jim Vallance, and while these demos often served as blueprints for what ended up on the finished record, Clearmountain used the rehearsal period to offer his own suggestions in terms of the musical arrangements.

"When I first heard 'Run to You' I thought it was pretty good," he recalls, "but Bryan was thinking about leaving it off the album. He was writing songs for other bands at the time, and there was some other band that he was going to give that to. I remember riding around town in his car when I first arrived and he was playing me the demos, and when we got to 'Run to You' he said, 'I'm not sure what I'm going to do with this one,' and I said, 'You're gonna put it on this album! It's a great song.'"

In truth, it was a song of simple and somewhat incomplete structure, looping around a hook without ever developing in the manner that might have been achieved by way of greater application during the compositional process.

"That could have been the cause of Bryan's uncertainty," Clearmountain agrees. "I don't think he considered it to be up to the standard of his other material. But it had such a great guitar hook, which was there right from the start, and everything was kind of based around that hook. Sure, it was real simple, with a nice and simple melody, but it just sounded like a hit song to me, as did a couple of others. It was 'Summer of '69' that I wasn't too sure about."

'Run to You' was very straightforward, comprising about a half-dozen takes out of which the best two or three were then chosen to edit between.

"Bryan was a good guitarist and he kept getting better as we went along, doing a couple of really good solos on the [1987] *Into the Fire* album," Clearmountain says. "He really worked on his guitar playing over the years and I now think he's a great guitarist — he was both meticulous in his approach and capable of letting it rip. He would let it rip and then we'd go back and fix bits. Both of us were pretty meticulous, and that's one of the reasons why we got along so well."

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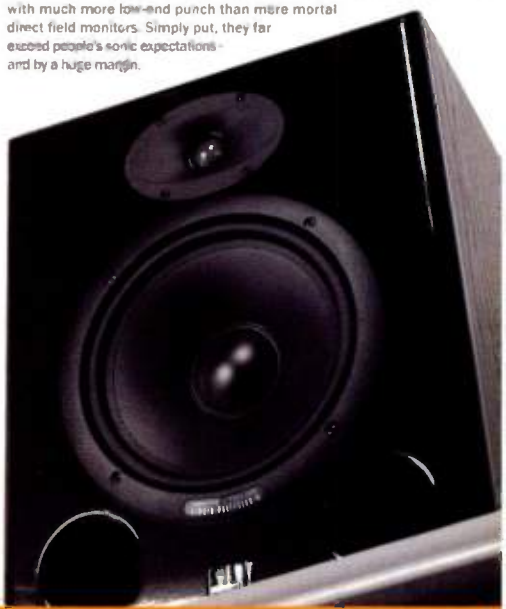

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"YOU WANT THE BEST? ALL THINGS CONSIDERED THIS ISN'T FAR OFF"
Guitarist - March 2006

RECORDING 'RUN TO YOU'

"[Guitarist] Keith Scott had a Marshall amp, recorded with an SM57 in front, but I'd also sometimes use a couple of mics, place a gobo nearby and face one of the mics away from the amp, towards the back of the gobo, to get this reflected sound. Then I'd mix that with the main mic, maybe put it out of phase, just trying to get a bit more. Sometimes we'd also combine amps — a Marshall with a Fender Twin, or something like that — just to get different sounds.

"In terms of the keyboards there was always a Hammond B3, so I'd usually have four mics on the Leslie — two 87s on the top and two on the bottom, recorded in stereo. Tommy [Mandel] also had this cheap little Casio keyboard which sounded really good. You can hear it on 'Run to You' — these little tinkling sounds, especially coming out of the solo section. The Casio was DI'd, and so was Dave Taylor's bass, which also used an amp — I think it was an Ampeg SVT.

"I would always put the bass player as close to the drums as I could get him, so they were as tight as possible, and Bryan would be standing somewhat in the middle of the room because he'd also be singing a rough vocal into an SM58 and directing the band. Keith Scott was off to the right and facing the control room, standing in front of the drums that were at the back and to the right."

An average of two backing tracks per day was the norm during the recording sessions — all of the pre-production evidently paid off — yet the Little Mountain session ended up lasting about three weeks due to some overdubbing of guitars, as well as the non-Pro Tools editing of each number.

Powering Up

The decision to relocate to Power Station was borne largely out of Clearmountain's desire to return to his home base.

"It was nice to be in New York, it was nice to get out of Vancouver for a bit," he says. "Then again, we couldn't really mix at Little Mountain. They didn't really have the facilities for mixing, and Power Station was great. I'd mixed tons of records there, so I was very comfortable, and Bryan also liked being in New York. Plus the fact that the keyboard player Tommy [Mandel] was from New York."

Before the mix took place on the SSL E-Series console in Power Station's Studio C, the facility's Neve 8068 came into use for more overdubs, as it had for all of the recordings on the preceding *You Want It, You Got It* album. Among the assignments this time around was to capture the lead vocals, with a number of different microphones being used.

"We would choose the mic for the song," Clearmountain says. "We actually used a Shure SM58 for a couple of the tracks because we wanted a real edgy sound, and then for other songs that we didn't want to be as edgy we used a U87 or an FET 47. That studio didn't have good vintage mics, and neither Bryan or I could afford expensive mics back then, whereas now I've got a few good microphones and he's got an amazing mic collection. So, back then we just used what was in the studio, and I remember at one point lining up one of every mic there and just getting him to try a verse and a chorus with each of them, before picking the one that we liked best. Usually it was a U87.

"Bryan was pretty confident about his

Photo: John Abbott

The control room in Studio C as it is today. The Power Station was renamed Avatar Studios in 1996 and continues to enjoy a reputation as one of the world's best recording studios.

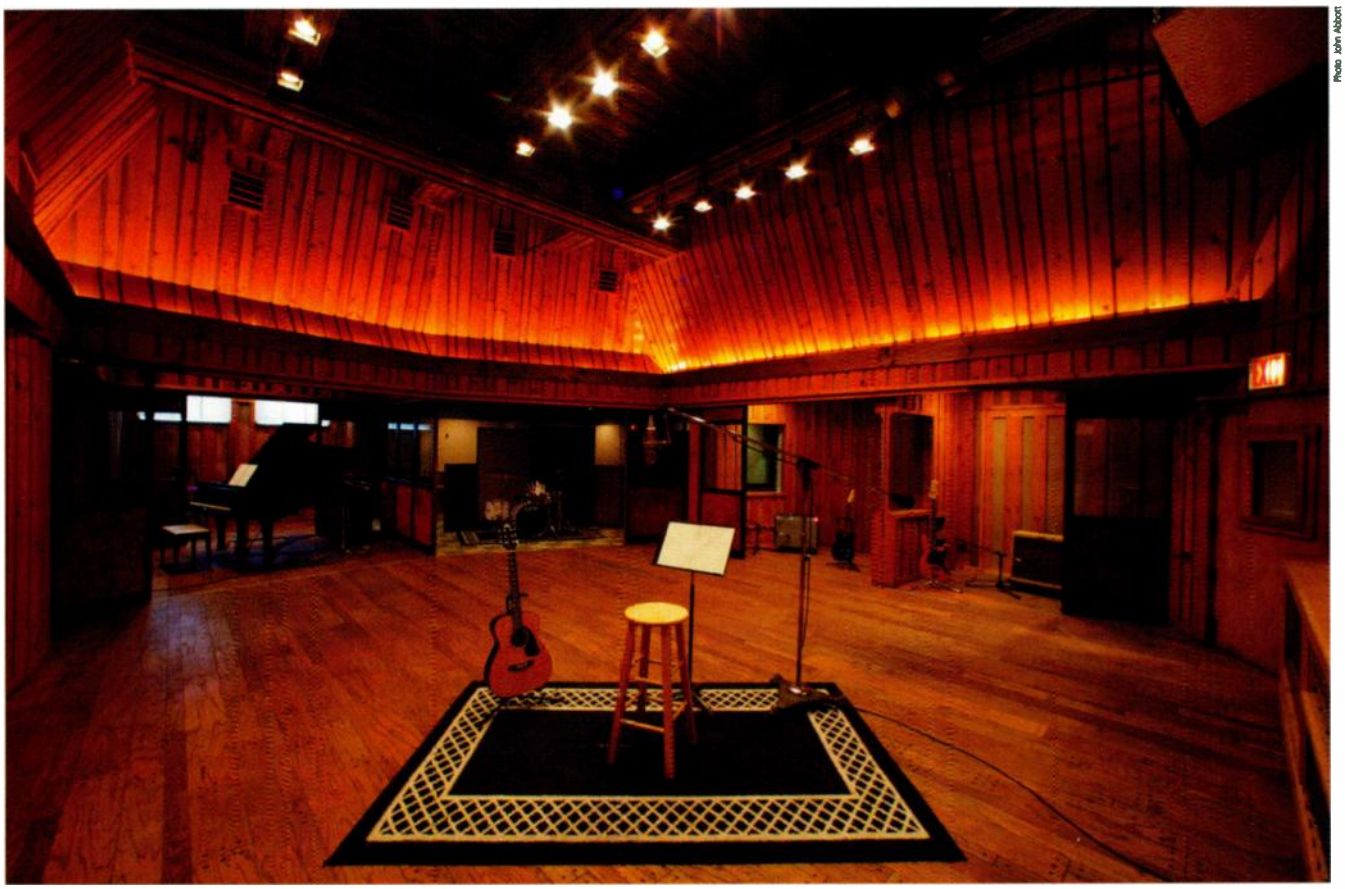


Photo: John Abbott

The live room in the Power Station's Studio C as it is today.

vocal abilities and also very objective. He's got an amazing ear. We'd be doing vocals and he'd go, 'Oh, I sang out of tune. Let's do that again'. You see, we wouldn't do comps in those days, because it was all 24-track. We'd have two tracks and keep punching-in on one track, and he was really good at that. He could punch word after word and it would sound like a performance. He was pretty amazing at that. He would perform a line and go, 'Okay, yeah, that was good,' and I would say, 'Well, let's try it again.' He'd go, 'No, no, no, that was good,' and I would persist: 'Let's try one more on the other track.' So, we'd give it a shot and then compare the two, and if it was better then I'd just bounce it over. That was the extent of our comps."

Adams' aforementioned ability to punch-in is all the more remarkable in light of the sustained high energy and rounded performance of a vocal such as that on 'Run To You'.

"He was unbelievable at that," Clearmountain confirms. "And he would never get worn out, because he'd only do a couple of lines at a time. Each line was like a burst of energy, so it wasn't a case of being a little bit tired by the time he'd get to the third verse. He'd just concentrate on each line, and he would use that technique to really get something amazing. In fact, he would often start off by doing a couple of complete passes to get a take that he felt really good about, and then we'd go back and listen to it and say, 'Oh, we can do that line better.' We'd go line by line, and he would always say, 'Yeah, I can do that better,' so we'd usually end up redoing the whole thing. It was a case of having a blueprint to work with and then just outdoing it.

Hurry Up & Wait

"Bryan always had a tendency to sing on top of the beat because he was so energetic, and then I'd actually have to pull him back, saying, 'Okay, you're too far ahead of the beat. Keep the energy and pull it back.' He would pull back so that he was still pushing it, and that's part of where the energy came from. In fact, that's why I was disappointed when he started to work with 'Mutt' Lange — Mutt would actually sample each line of his vocal and lay it back in so that it was exactly on the beat. Mutt had a mechanical approach where he wanted it to be perfectly in time, and to me that kind of overlooked Bryan's energy, where he was pushing the band, leading the band. I always thought that was an exciting thing about his voice, and the later albums don't really have that. Of course, most listeners aren't aware of it, but there's an immediacy to the way that he pushes everything."

Likewise, Clearmountain asserts that the energy of the thundering chorus on 'Run to You' was more down to the recording than to the mix.

"Most of it was the performance," he remarks. "We'd always push for really exciting performances. Like on *Into the Fire*, there's a song called 'Victim of Love' which has this long outro, and we had Mickey Curry just go out and fill up a whole tape with drum fills. He would play the end of the song and every four bars he'd do a different drum fill, and then we'd go through it, pick the ones that we really liked and place them in the outro. We did the same on *Reckless*, where a lot of the songs were kind of pieced together even though they sounded totally live. It was all about getting the most exciting bits, and once in a while we'd also get a great take. That was pretty rare, but occasionally we would actually have a full take.

"To be honest, it was a great band. Mickey Curry is an unbelievable drummer, besides being hysterically funny, and so we'd always have a great time cutting tracks. Everybody would be cracking jokes in between cuts and sometimes they would just start jamming on something. There was usually a Linn drum machine in the control room, which we'd use to provide a click track, and so Keith would do these little rap things, I'd start playing handclaps on the Linn, and we'd just crack ourselves up with these silly, stupid things and then go do a take. Everybody would be

pumped up, having a great time, and it was all about getting this vibe in the studio. I think that comes through on all those records, where it just sounds like there's this energy going on. Well, that was there. It was in the original recordings.

"We didn't have the boxes back then to create different room sounds, so we did it the hard way, and that was always the fun of recording for me. No matter what record I was producing, I would always insist on having everyone in the band play together, even if ultimately we weren't going to keep the tracks. It was about having everybody's vibe in there and the drummer hearing as much as possible coming through his headphones. That energy was so important, whereas a lot of the records nowadays are all done bit by bit and they don't have that thing that those Eighties records had; the excitement of a band playing together.

Forever Delayed

"Sonically, I didn't leave much to the mix. Especially recording on analogue, which is so different to digital. Nowadays you can record pretty much flat with digital and then do everything in the mix, but you couldn't do

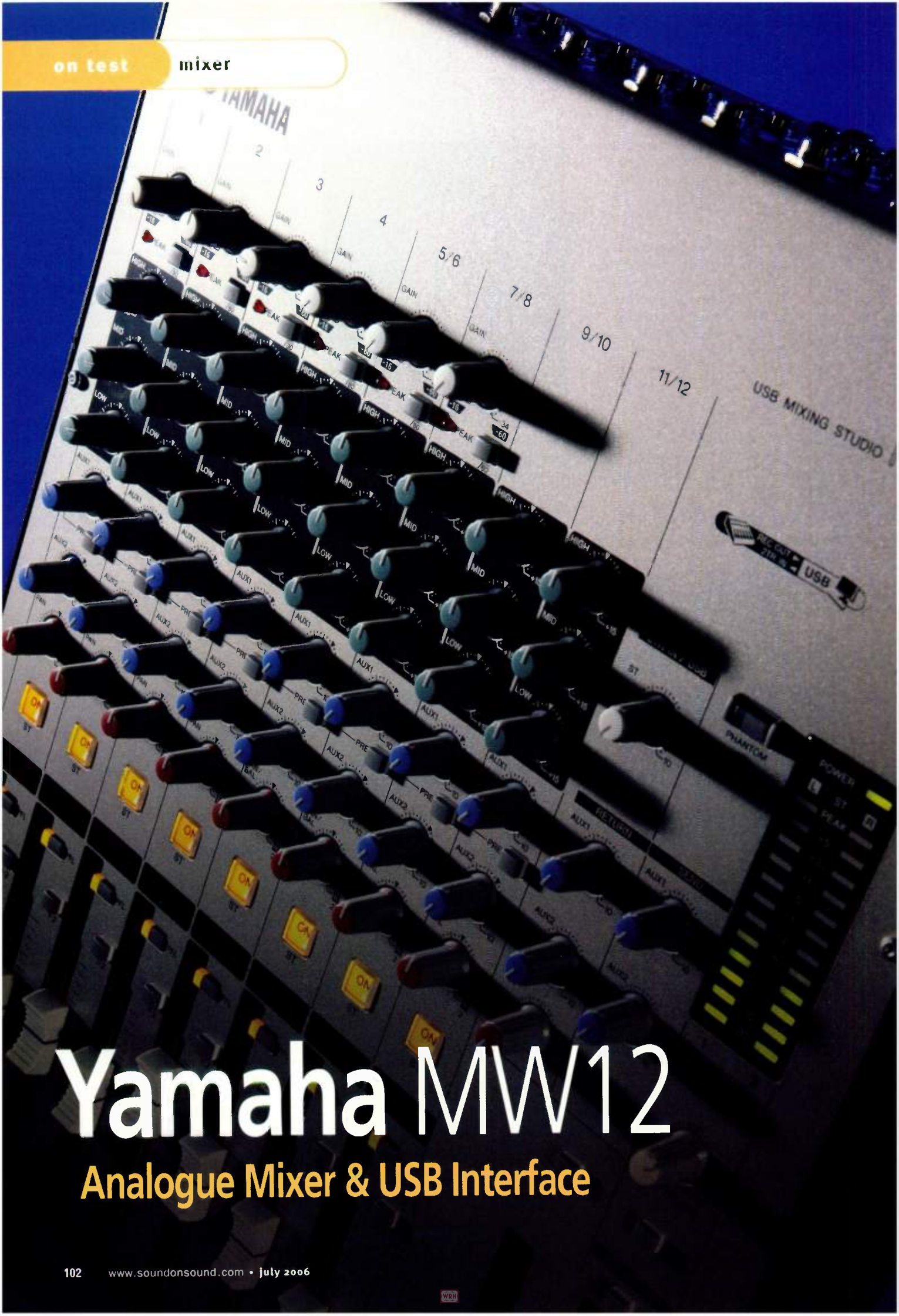
Bob Clearmountain today.

that with analogue because you'd just get a load of noise. You'd go to EQ and be bringing up tape hiss, and the more you'd play the tape the duller it would get. So, I would always put extra top-end when I EQ'd, trying to make something sound as if I was mixing it when I was recording it. You know, I'd really crank the treble, because the nature of analogue was that as soon as you played it back it would be missing something. That's why I was so glad when digital started to sound good!"

As for the mix of 'Run to You', Bob Clearmountain has never been completely happy with the effects that were applied to Bryan Adams' vocal... Too much delay for his liking.

"That's one song I've always wished I could have remixed," he admits. "Looking back now, I don't know why there was so much delay on the voice. It obviously seemed like it sounded good at the time, but when I listened to it later I thought, 'Jeez, I wish I hadn't put so much delay on the voice and I wish I hadn't put so much bass in the mix.' I thought there was too much bottom-end, and it never sounded right to me, but then other people seem to think it was fine.

"The song starts without any bass, just the guitar lick and a little cross-stick on the snare drum, and it sounds so great. But then the bass kicks in and hits radio compressors and the whole thing gets kinda quiet, and it always bugs me when that happens. At home it sound fine and in the car it sounds fine, but on the radio it's another matter. Still, that didn't keep it from being a hit, and nobody else has ever commented on the problem, only me. So, who knows?" SOS



Yamaha MW12

Analogue Mixer & USB Interface

A compact analogue mixer that also acts as a stereo USB audio interface.

Tom Flint

The MW12 is a 12-channel analogue mixer with a built-in 16-bit/44.1kHz USB interface, allowing a stereo mix to be sent directly to and from a Mac or PC. A copy of Steinberg's *Cubase LE* is bundled with the mixer, demonstrating just how keen Yamaha are to promote this part of its feature set. Without a computer attached, however, the MW12 functions just like any other hardware mixer of this size. First and foremost, the Yamaha MW12 is a studio-oriented product — it doesn't, for example, have any of the built-in effects that are often found in mixers of this size intended for live use. Nevertheless, its feature set includes a few output busses that could easily be routed to fold-back monitors, and the chassis, derived from Yamaha's MG-series live sound mixers, should withstand a few knocks.

Inputs & Outputs

Apart from the headphone jack output positioned on the top panel, all of the MW12's inputs and outputs are round the back of the unit (pictured above right). The mixer can be fitted with side brackets for mounting it on top of a rack, in which case having the I/O at the back would be ideal.

Most prominent are the six phantom-powered XLR sockets, four of which are routed to the mixer's mono channels, together with four balanced quarter-inch jack sockets. Each of the four mono channels also has its own unbalanced TRS jack insert point, allowing a processor such as a compressor or gate to be inserted into the signal path between the EQ and fader.

Stereo channels 5+6 and 7+8 are slightly different in that they offer three different input options. If just a mic is connected via the XLR, its signal is distributed to both the left and right channels. However, there is also a stereo pair of balanced jack sockets that take priority over the XLR if used. A mixture of the two input types can be achieved by plugging a jack into the right-channel input only. This automatically routes the XLR signal to the left channel and the jack signal to the right. A mono signal can also be fed to both sides by using just the left-channel jack input.

The remaining channels, pairs 9+10 and 11+12, offer slightly different options again. There are left and right jack sockets, this time unbalanced, and these are joined by a pair of RCA phono connectors. The manual suggests that all four inputs can't be used together, but this is not quite correct. In fact, the jack and phono inputs simply get mixed together, albeit at a reduced level. The mixer's remaining inputs consist of a stereo aux return on balanced jacks and a pair of RCA phono inputs labelled '2TR In' which share the same input path as the signal returned from the computer via USB.

As far as outputs are concerned, for the main stereo mix there is a choice of XLR or balanced TRS jack outs, plus a line-level stereo output on RCA phono sockets. The latter is labelled 'Record Out' and is intended for use when sending the mix to DAT, CD-R and other two-track recorders. The remaining outputs are all on balanced jacks — a pair for the Control Room output, a pair of outputs from the mixer's stereo group buss and two more carrying the send signals from aux busses 1 and 2.

Test Spec

- Steinberg *Cubase LE* v1.0.8
- 2.66GHz Pentium 4 PC with 256MB RAM running Windows XP Home.

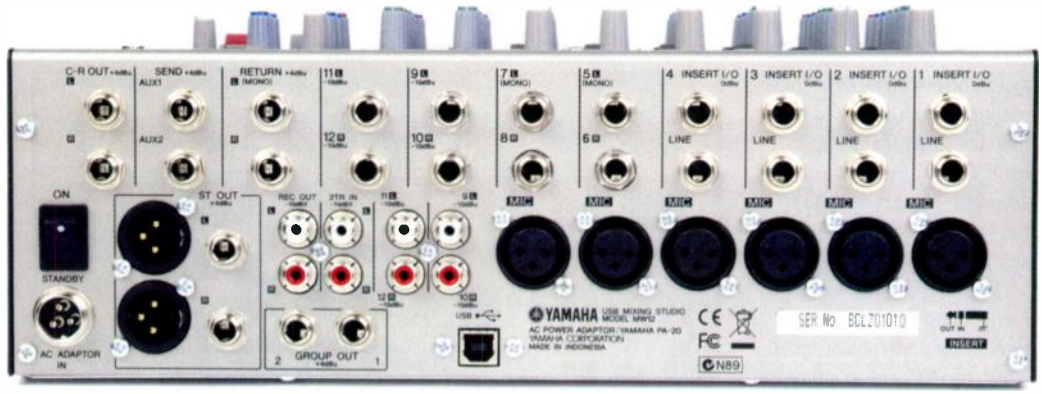
Features

The four mono channels have identical facilities, headed by a gain control offering a 44dB range, a peak indicator LED and a high-pass filter (80Hz, 12dB-per-octave). The EQ is a simple fixed three-band design offering +/-15dB, with high, mid and low bands centred at 10kHz, 2.5kHz and 100Hz. The high and low bands are shelving filters, whereas the mid band is of the peaking variety.

Next in the path are two aux send controls, the first of which has a pre-fade switch allowing you to take the send from either before or after the channel fader. A pan knob is followed by a button labelled 'ST' which turns the individual channel's routing to the main stereo buss on and off. A PFL (pre-fade listen) button feeds the channel's pre-fader signal to the headphones and Control Room outputs, while a button labelled '1-2' routes the channel output to the mixer's group buss, regardless of whether the ST button is on or not. The group has its own master fader and rear panel outputs so that it's possible to set up a custom stereo mix and output it separately.

The first two stereo channels — 5+6 and 7+8 — have identical controls to the mono channels, however 9+10 and 11+12 are missing the gain knob, peak LED and high-pass filter, and are best suited for receiving the returns from an effect processor connected to one of the Aux outputs, or for accepting the output of stereo sound modules with plenty of gain adjustment of their own.

In the master section is the ▶



SOUND ON SOUND

Yamaha MW12 £280

pros
- Simple but effective EQ.
- Four inserts.
- Good build quality.
- A few useful bussing and routing options.
- Convenient way to record to a computer.
- Free copy Of *Cubase LE* software.

cons
- No EQ bypass.
- Noisy monitoring via USB.
- Global phantom power switching.

summary
A really well-made mixer with only a few minor limitations. Proof that hardware and software each have their strengths, and a combination of the two can often provide the best solution.

YAMAHA MW12

Alternatives:

Yamaha are not the first manufacturer to build a mixer with built-in computer interfacing. A Firewire option card is available for Mackie's Onyx mixers and the Alesis Multimix Firewire range offer Firewire interfacing as standard. A closer comparison to the MW12 might be Alesis' 12-channel Multimix 12 USB mixer, which also ships with *Cubase LE* and even includes a digital effects processor. It's a little cheaper than the MW12 but can't quite compete with the Yamaha's routing and I/O options. The Behringer Xenyx 1222FX also costs less than the MW12 and comes with an external two-in, two-out USB audio interface.

aforementioned stereo group fader, plus the master fader. There's a 12-segment meter, a global phantom power on/off button, aux send and return controls and a level knob that controls both the headphone and Control Room outs. These outputs can be switched to monitor either the group or main stereo buss, and you can also route the group buss to the main stereo buss, should you wish to.

The final control worth highlighting determines the level of the USB return signal, which can be sent to either the stereo buss or the control room/monitor buss. All of the mixer's 60mm faders are scaled to provide the most control around 0dB — the area from -10dB to +10dB occupies more than half of the fader's travel.

Sound Tests

I tested the performance of the MW12 by feeding it and my Behringer Eurorack MX1602A mixer the same audio source. The Behringer is a cheaper product with a lacklustre EQ, but I'd previously tested its 'ultra-low noise' preamps against those of an Allen & Heath Mix Wizard and they'd fared quite well, so I consider it to be a good benchmark. Having matched the levels as closely as possible I concluded that the Yamaha behaved a little more evenly across the audio spectrum and seemed a touch less noisy than the Behringer, although the difference in gain scaling made it impossible to come to any definite conclusions.

The MW12's preamps do begin to get rather noisy when they are pushed to their maximum level (as most designs tend to) but you only have to back off the gain about 10 percent before it ceases to pose a problem, and I found that there was more than enough gain for me to DI my electric guitar without having to push either the preamp or channel fader to its maximum.

Having three bands of fixed EQ is limiting, as it pretty much rules out the possibility of making any really precise adjustment to an input signal. Nevertheless, Yamaha seem to have picked the right filter shapes and frequency points to ensure the EQ is effective as a general tone shaping tool on the majority of sound sources. What's more, even the most extreme boosting and cutting still sounds quite natural, which in my book is the mark of a good EQ. It would, however, be advantageous to have a bypass button as this makes it quick and easy to A/B any adjustments against the un-EQ'd signal.

USB Output

Recording the USB output into the supplied *Cubase LE* software is fairly straightforward, although it is necessary to follow some of the setup instructions in the manual in order to establish the correct signal routing. On my PC, the Device Wizard was summoned when I connected the mixer, adding the USB Audio CODEC to the *Cubase LE* ASIO Multimedia setup window. I was then able to select the MW12 as the default device and as the active I/O ports, as advised in the manual notes.

From then on recording is simple, even if establishing the optimum monitoring setup is less so. The ST switch of each mixer channel determines whether the signal is sent to the USB port, and this routing is made post channel fader, pre master fader. The output level from the computer is controlled by the mixer's 2TR In/USB knob, but it's necessary to route the incoming signal to the control room buss rather than the main stereo buss to avoid re-recording it during overdubbing. Overdub monitoring level can be set against the pre-recorded playback without affecting the recording level by using the master fader.

There were no problems with latency, but there was a considerable amount of noise present when monitoring the USB return, although this didn't affect the quality of the recorded audio.

Conclusion

The MW12 is designed with flexibility in mind so that the user can make the best possible use of its limited channels and I/O in whatever setup they have. If, for example, you are not using an external effects unit in the aux loop, the inputs double as a spare stereo input, while the group buss could be used to feed custom mixes to anything from stage monitors to effects modules. The insert points on the four mono channels are a definite plus point too, as they open up the possibility of using dynamics within the signal path.

Semi-pro mixers typically come with output options for feeding the stereo signal directly to DAT machines or tape recorders, but now the home computer can be added to the list, and getting a copy of *Cubase LE* bundled free is a definite bonus. There are a few minor complaints to make — globally switched phantom power is a touch restricting, there's no EQ bypass switch and the USB output is noisy — but nevertheless, for the price, Yamaha are offering a very well-designed mixer. SOS

information

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Marshall Regenerator, Echohead & Reflector

Effects pedals

Marshall have added a new range of digital guitar pedals to their portfolio, but rather than stick with the usual 'one trick per pedal' format, they've come up with three pedals, each of which features six different effect modes, selectable via a rotary switch. They've also given the pedals two outputs — one is fed from a true mechanical bypass and the other allows the decay of delay-based effects to trail off naturally when the effect is switched off. When both jacks are used together, you lose this latter feature but gain the advantage of stereo output. Each pedal also has an extra input jack, accepting either an expression pedal or a footswitch.

The pedals have solid cast-metal casings and can be powered from a 9V battery or an optional power adaptor. The battery compartment is underneath, secured by a large-headed screw that can be unfastened with a small coin. A non-slip rubber base keeps the pedals stable on the floor and the controls are set back out of the way of careless feet.

First up, the **Regenerator** pedal (£60) offers a choice of Vintage Chorus, Multi Chorus (based on Marshall's own Supervibe pedal), Vintage Flanger, Phaser, Step Phaser and Vintage Vibe modes. Besides the rotary mode switch, two further knobs control rate and depth. The final knob — labelled 'Regen' — feeds some of the output back into the input.

I found the Vintage Chorus mode a little less fluid than, say, the Boss CE-2, but it worked well enough. Multi Chorus produced a richer sound that I generally preferred. Flanger produced a very traditional flanging sound, but without a control to adjust the nominal delay time, it isn't quite as flexible as some units. Nevertheless, it's a very musical and useful sound. The standard Phaser setting sounded right to my ears — warm and smooth — while the Step Phaser creates a jumpy, synth filter-type effect. Finally, the Vintage Vibe recreates the classic Univibe-style sound that is half way between chorus and phasing, used extensively by Jimi Hendrix. I thought this sounded pretty authentic. An expression pedal may be plugged in to control the modulation rate of any of the effects.

The **Echohead** (£65) has a similar layout, with the knobs adjusting Mode, Delay Time, Feedback and Level. The dry level of the guitar always remains constant so you never end up with an echo-only output. The mode settings are Hi-fi, Analogue, Tape Echo, Multitap, which produces a more dense cluster of echoes, and Reverse, which delays and reverses the effected part of the sound and adds it to the normal dry sound. Finally, there's Mod Filter, which adds a filter sweep to the delayed sounds. An optional footswitch can be used to tap in the delay time.

I rather liked this pedal, particularly the Tape Echo and Analogue settings, which nailed the tonal character of the real thing pretty well. Tape Echo even introduced some gentle pitch variations to give the effect of worn transport components. Mod Filter was more subtle and musical than I'd expected and the Reverse setting provides a nice variation on normal delay, though you can't dial in 100 percent wet and play backwards solos as you can with some units. In stereo mode, the Multitap delay ping-pongs from side to side to add movement and interest to the sound.

Reflector (£60) is dedicated to reverb effects, with Hall, Plate, Room, Spring 1, Spring 2 and Reverse modes. The middle two knobs control decay time and high-frequency damping while the fourth controls the mix of reverb and dry signal. Most of the modes are self-explanatory, though Spring 2 is a blend of spring and electronic reverb while Reverse puts a reverse envelope on the reverb decay making it appear to build up rather than decay. The reverb may not be quite studio grade but it is still pretty good and certainly more than adequate for live performance. I liked the character of the Plate setting but also found the Room setting to be useful for adding space to a sound at short decay times. An expression pedal can be used to adjust the decay time and there's a nice stereo spread to the sound if you use stereo outputs to feed two amplifiers.

These pedals are tough, practical and sound very good without actually breaking any new ground. That's not such a bad thing, as most of their algorithms are designed to recreate vintage effects! Given that they cost around the same price as a decent single-function pedal, this range combines flexibility with value, and the build quality should ensure they stand up to life on the road. *Paul White*

SUMMARY: These rugged, versatile pedals offer authentic sounds and value for money.

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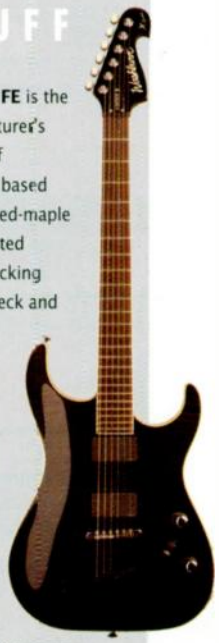
COOL STUFF



The new **Washburn X50 Pro FE** is the latest addition to the manufacturer's rock-oriented X Series range of solid-body electric guitars. It's based on the X50 Pro, but has a flamed-maple carved top (as opposed to quilted maple) and active EMG humbucking pickups — an EMG 81 at the neck and an EMG 85 at the bridge. The X50 Pro FE also features a plain rosewood fingerboard and Grover 18:1-ratio tuners, and it uses the patented Buzz Feiten tempered tuning system, designed to give better intonation. It's available now in transparent black or transparent red gloss finishes, priced £599.

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TECHNIQUE Tuners and tuning stability

Guitars can suffer from tuning stability problems for a number of reasons. Most stability problems can be traced back to what might be called the friction points — the bridge and the nut — but strings can also bind on string trees (where fitted) and slippage around the tuning machines can also be significant, especially if you don't fit the strings correctly.

Everyone has their own way of fitting strings, but I tend to pass the string through the hole in the tuning machine post, and then put a right-angle bend in it to prevent it slipping back through. As I tighten the machine, I'll take one turn around the post above the string hole before guiding the string below the hole and then winding two more turns around the post. On tuner posts that taper inwards towards the centre, this helps grip the protruding string end between the upper turn and the lower ones. By limiting the number of turns to around three in total, and by winding the string neatly rather than having turns crossing each other, there's less risk of the string slipping. After fitting new strings I'll give them a good hard pull to stretch them out, then check the tuning with an electronic tuner, and keep repeating the process until the tuning remains stable. By doing this you can usually play a gig ten minutes after fitting new strings with very little in the way of tuning problems.

Having said all that, I prefer to fit locking machine heads to any of my guitars that have tremolos as they really do eliminate string slippage at the tuning end of the string. Some manufacturers, such US company Sperzel, produce sets where post heights get progressively lower the further you go from the nut, so on many Strats and similar instruments, you can remove the string tree (or pass the string above it) without risk of the strings popping out of their slots or losing tone through insufficient break angle over the nut.

Players who haven't used locking tuning machines before sometimes make the mistake of thinking that it is the rotation of

the string peg that is locked, but this isn't the case. The way the Sperzel tuner works is that there's a hole down the centre of the shaft that intersects with the string fixing hole, and inside this centre hole is a metal pin that can be driven towards the string hole by means of a knurled knob on the rear of the machine (pictured left). Once you've passed the string through the hole in the peg, you simply tighten the knurled knob to clamp the end of the string in place. Not only does this prevent slippage but it also enables you to take only half to three-quarters of a turn around the peg, which means there's less chance of the string windings sliding around on the peg and affecting the tuning.

When fitting new strings, I pull the string through the hole just far enough to take up the slack, and then clamp it before tuning normally. This seems to leave the right amount of string around the post once the string is up to pitch and also leaves enough tuning range for those players who like to re-tune to a drop-D or other alternate tuning. One tip when fitting strings to Sperzel locking tuners is that if the string doesn't seem to want to poke through the hole in the post, and you've checked that the knurled knob is loose, just give the tuning shaft a gentle tap on the top as the pin sometimes sticks. *Paul White*

COOL STUFF

T-Rex Engineering have produced a pair of new pedals. The **Room Mate** is a valve-equipped reverb pedal offering four types of effect — Classic Plate, Warm Hall, Bright Hall and Chorus with Warm Reverb. It has a mono input and stereo out and has Mode, Mix, Level and High Cut controls. The **Bloody Mary** (pictured near right), on the other hand, is an all-out distortion pedal with Gain and Level controls and High, Mid and Low EQ knobs. A Body switch boosts the bass and low mid-frequencies. ASAP Europe +44 (0)20 7231 9661.
www.t-rex-engineering.com

Vox have added two new tube-driven pedals to their Cooltron range. The **Duel Overdrive** pedal (pictured far right) is based on Vox's Big Ben overdrive and provides two separate drive effects with independent Gain, Volume and Tone controls. A bass boost can be added to one or both overdrives. The **Vibravox** combines tremolo and vibrato effects. In addition to Depth, Ratio, Skew and Volume knobs, there are two Speed controls that govern both effects. The Vibravox can be set to switch between two speeds of tremolo, two speeds of vibrato or between a tremolo effect controlled by the first Speed control and a vibrato effect set by the second. Both pedals run off four AA batteries or a 9V mains adaptor (not included). The new Cooltron effects will be out in June, with pricing to be confirmed.
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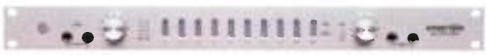
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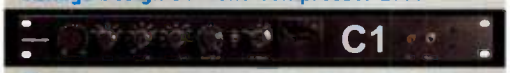

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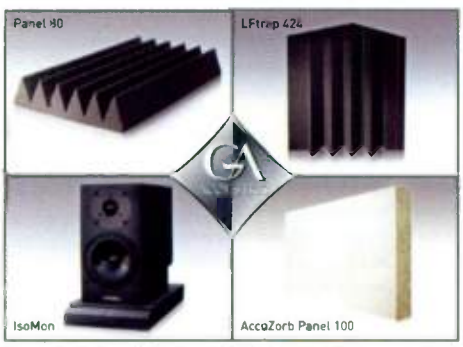

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Sound & Vision

Making A Living From Music For Picture: Part 8

Bill Larey

In music-for-picture work, there's never enough time to do what you'd ideally like to. We look at what it's like producing music under pressure.

When scoring music for picture, it's always worth bearing in mind that time is one luxury that you will rarely have in sufficient quantities. Deadlines must be met; 'writers block' is unacceptable, and a sharp sense of focus is essential (fresh coffee and snacks can be helpful as well). Successful completion of a project will usually entail long hours and working late into the night. You'll need to get along well with clients, and learn to accept criticism and make changes regardless of your personal opinion. Be careful what you wish for, as making a living doing what you love will demand much from you.

Last month, I introduced the trailer to the film *The Narrow Gate*. You've had a chance to rise to the challenge and have a go yourself. Now we'll look at what I did and how I did it despite not having enough time...

Time Is (Not) On My Side

The circumstances surrounding the scoring of the trailer for *The Narrow Gate* were a little different than the other films we've discussed. Shortly after completing the score and mix, producer Heather MacAllister informed me that she would be travelling to the Sundance Film Festival, and needed to quickly put together some promotional materials for the film that could be handed out. We agreed that a trailer had to be cut for the web site, and that a DVD with the trailer on it would be helpful as well. However, the film editor was not available, so I tackled the editing job myself. This was just as well, as Heather only had one day available to edit the trailer, have the music scored, *and* create the DVD!

Having previously given *Final Cut Pro* seminars for Apple Computer, I'm pretty well-versed in the art of video editing, and I'm usually able to move quickly enough to get a job done. However, first I had to digitise the film and select appropriate source clips for the trailer, which took a few hours in itself. Once the trailer edit was finished, I created a Quicktime movie of it and imported it into my sequencer. By this point, half the day was already gone and I had to move fast to create the score. As I discussed last month, this trailer needed to be very dramatic, to generate excitement and interest in the film (you can download

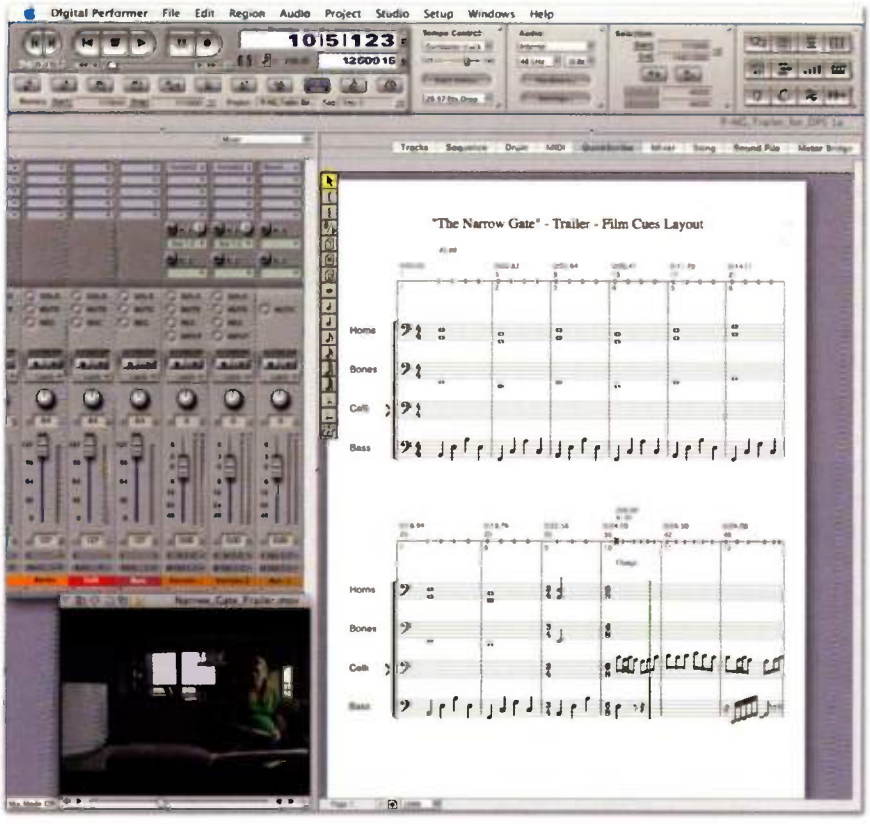
Contracts

One area that we have not yet touched on in this series is work contracts. I'm not a lawyer, and would prefer that you seek legal advice regarding contracts from appropriate professionals, but a few of my thoughts on the subject might prove helpful.

Every job opportunity often requires a different approach. For much of my television writing, where most of the clients have come about as a result of long-term relationships, contracts are rarely written up. The same is true with music houses that I've worked with for many years. Verbal agreements regarding terms and budgets are common. I think it is a general rule that the more money is involved, the greater the need for some contractual agreement in writing. When the project you're working on is going to be broadcast on major networks, contracts are required for musicians as well. Some independent films I've scored, on the other hand, have proceeded on a handshake. For others, a simple contract was drawn up after speaking with the client. And for a few others, it was necessary to have a lawyer review the terms. Use common sense when proceeding in this area. It certainly doesn't make sense to get all worked up and hire a lawyer for a low-budget job with a first-time filmmaker.

In the case of *Dinka Diaries* (see overleaf), producer Filmon Mebrahtu presented me with a fairly detailed contract stipulating ownership, delivery dates and compensation. When dealing with films that are funded, especially by private or non-profit organisations, it is necessary for *everything* to be done in writing. Don't hesitate to ask questions or request revisions; after all, contracts are designed to protect both parties.

On the whole, I have found that contracts are a good thing. As you get higher up the food chain, they can get fairly complicated and deal with royalties, soundtrack recordings, cue sheets, session costs, licensing issues, and so on. Read them carefully, and be sure to specify in writing exactly when you are to be paid. If you don't, you can be sure that it will be much later than you hoped!



Part of the finished score for the *Narrow Gate* trailer in MOTU's *Digital Performer*. Note that the score is written in on a now-completed version of the Film Cues layout introduced last month.

the clip with the finished music at www.soundonsound.com/sos/soundandvision/narrowgatetrailerafter.mov).

To get started, I selected a palette of instruments. For dramatic percussion, I loaded the 'Taiko Earthquake' and 'Thunder Ensemble' patches from East West's *Stormdrum* virtual instrument. Strings were covered by the VSL (Vienna Symphonic Library) *Horizon Opus 1* library loaded into Native Instruments' *Kontakt*. I wanted an ostinato bass line, so I loaded 'KB-6_PZ', a pizzicato bass patch. As I had to move very fast, I loaded a string Multi patch which covered a large enough range; 'VI+VC+KB_Stacc' combines violins, cellos and basses in a single patch with staccato articulation. I knew I would not have time to load the individual patches with more samples layered per note, and I would have to sacrifice the violas for the sake of speed.

To give the basses more weight on some low sustaining notes, I also grabbed a patch from my East West/Quantum Leap *Symphonic Orchestra Gold* library, 'F CBS Big Sus'; I often layer sample libraries together when I need a fuller, richer sound than any one can provide. For French horns and trombones, I went with the Project SAM libraries, loading forte patches '23 HN Sustain F' and '30 TBN Sustain F'. Finally, I added a tam tam and, from the VSL *Opus 2* library, a sustain flute ensemble patch 'FL-3L_Sus'.

As I discussed last month, composers usually prefer to work in private, and present only finished pieces for review. In this case, though, as with last month's *Ghost Soldier*, the client was sitting right next to me in the studio. The clock was ticking and there was no time to be shy; I just had to jump in and start scoring. Since we needed to open the trailer with something dark and ominous, I scored the French horns and trombones at the lower end of their ranges (see the screengrab of the arrangement on the previous page), playing sustained whole notes against a quarter-note bass ostinato line. Astute readers will notice a similarity between the bass line here and the one used in *The Destruction Of Civilization*, the film I scored and used as an example in parts 5 and 6 of this series; in fact, it's almost identical. Scandalous plagiarism ...not! No one ever accused the Beatles of ripping themselves off by reusing a I-IV-V progression on the guitar, and the bass line is, after all, just a supporting element here. When you have little time to compose, you'll often have to draw upon whatever works to get the job done. End of story!

To add to the ominous character of the opening bars, I added a soft taiko drum hit on each dissolve. The placement from scene to scene was a little inconsistent, as I wanted them to land on a beat and in the black for maximum impact. It almost sounds like two simultaneous time signatures happening, which is a musical effect I like. At bar 10, the pace quickens, as the shots move inside the apartment of the central character and the viewer witnesses her growing terrorist paranoia. Staccato cello lines take over here, and are overlapped with slightly confused flutes to further emphasise the characters' state of mind. A low bass line also comes in to set us up for our next section. Notice that despite the changes, I don't have more than three musical parts playing at once — I'm observing the 'Rule Of Three' that I discussed in part 6 of this series.

At bar 18, the pace picks up even further, and the dramatic build to the finish line begins. The French horns make a dramatic entrance, while the basses play a fast moving staccato line that is doubled and tripled by the cellos and violins. The string textures thicken, and at the point where the picture goes to black, the tam tam and taiko drum sound and ring out. To push this a little further, I added a horn glissando from VSL's *Epic Horns* library, 'Ho-8_Gliss_12_Up' to lead into the percussion sting. It's a bit of a cliché, but very effective!

Dinka Diaries producer Filmon Mebrahtu.

Photo Reel Voices/Vera Vidtz-Ward

As I've mentioned previously, I'm really a guitar player with dreadful keyboard technique, and playing this last section in real time without quantisation was a challenge for me. It didn't help that the client was sitting next to me, hearing one embarrassing attempt after the next. Time was getting short, and we needed to wrap up so that we could move on to the DVD. So I politely explained to the client that I would prefer to put some headphones on and turn the speakers off. Once that was done, I proceeded to slow the tempo down until I could play it just right. That went very well, and when finished, I put the tempo back up to the original speed and played it for the client, who was extremely pleased. So whatever works to get you to the finish line is fair game!

I bounced the mix and assembled the DVD in Apple's *DVD Studio Pro*. Once that was accomplished, I created a smaller-sized Quicktime movie for the web site. It was a long day, but by the end of it, we had a suitable film trailer, a dramatic music track, and a happy client. All in a day's work!

Until now, I've focused on dramatic films destined for cinemas, and now it's time to change gears a bit, and turn to the world of television, and to a documentary in particular. It is not uncommon for documentaries to require a degree of authenticity that is not always called for in the cinema; there can be less flexibility for artistic licence with the score, for example. Documentaries often require the music to be representative of a particular locale, or even a specific period in history, in addition to accommodating the subject matter. In the case of the next film, the director had some very specific musical requirements.

Meet The Director

The documentary film *Dinka Diaries* was written and produced by Filmon Mebrahtu. It tells the stories of African immigrants living in Philadelphia and their experiences dealing with American culture (for more about the film, see the box overleaf). Filmon was born in Eritrea, located to the East of the Sudan in Africa. In the 1970s, when he was six, his family left to escape the political and civil strife there. In 1984, after attending schools in India and the United Arab Emirates, he arrived in the United States to study electrical engineering in college. After working for a while in wireless communications, he developed an interest in filmmaking, acquired a digital video camera, and began to learn the art of filmmaking. From 2002 to 2004, he participated in the Independent Television Service (ITVS) Mentorship Program, working with award-winning filmmaker Louis Massiah. His most recent films include *Stop Killing Taxi Drivers* (2001) and *Rencontrer (To Meet)* (2004), which have been screened at various museums and festivals in the US. His work has also been broadcast on Philadelphia's WHYY and WYBE television channels. In 2005, Filmon was a recipient of the prestigious Pew Fellowship Award for the research and development of his new film project *Émigré*, which chronicles the experiences of two

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Senegalese Muslim immigrant families over the period of one year — one family in Paris, the other in Philadelphia. It is through documentary filmmaking that Filmon seeks to understand his own immigrant experience, while exploring the experiences of others.

Filmon contacted me after listening to some samples of music on my web site. When I first spoke with him by telephone, we discussed the subject matter of the film and some specific issues that were of importance regarding the music. We also needed to determine the feasibility or working together, as we were some 130 miles away from one another. We agreed to meet at my studio and discuss the particulars. Distance is less of a concern these days with the advent of high-speed Internet access (more on this in a moment), and we were able to put that issue to rest. The bigger hurdle to overcome was the temp music.

Filmon was influenced by the music of African blues guitarist Ali Farka Touré; he was attracted to the blues element of Touré's playing, and felt it would work well to represent the universal experience that the immigrant teenagers in the film were experiencing. The good news was that I play the guitar. The bad news was that I had never heard of Ali Farka Touré before. Luckily, Filmon brought along a few CDs of Touré's music as a guide. While he did not edit the film with this music, he had a few specific tracks that he felt were appropriate, and he asked that I listen to them.

So began my education in the music of Ali Farka Touré! But then one of the benefits of working on a documentary film is being introduced to music that you might previously have been unaware of. I grew up admiring the great blues rock guitarists of my generation, but African blues guitar was something a little different. We agreed that a solo acoustic guitar playing in the style of Ali Farka Touré would be the best solution for certain scenes in the film, although it was not important to exactly mimic his style. For other scenes, a string orchestra would be suitable, along with African percussion, the idea being to make the music a significant hybrid of American and African styles.

As I mentioned in Part 5 of this series, I like to take on jobs only if I feel that I can handle them. Had Filmon wanted a virtuoso violin to be the focal point of the film, I would have passed on this project, as such a task is beyond my skills, and the budget did not allow for additional musicians. So being a guitar player was the deciding factor for me in accepting the job. What's interesting is that there were few examples of my guitar playing on my web site! Yet Filmon liked what he heard despite that, and in the end, it was a bonus that I was a guitar player. Filmon is something of a guitarist himself, so it was fairly easy to communicate about the specific parts that would be played.

Email, Web Or FTP?

The Internet has certainly changed our lives. Faster speeds and cheaper prices continue to improve our ability to communicate and exchange files with clients. For years we've been able to exchange small MP3 audio files via email, but it's only recently that it's started to be practical to send video files around electronically, as the file sizes are so much larger. However, clients usually prefer to hear the music in context, with the visuals, and there are now a few solutions

About This Month's Film

Dinka Diaries is a documentary film that tells the story of three Southern Sudanese teenagers who are uprooted from their homes in the Sudan and find their way to America. In November 2000, Mike Kuch, Abraham Kuol and Joseph Deng, all under 18, arrived in the city of Philadelphia. Two years later, they collaborated with Filmon Mebrahtu to document their experiences as they assimilate into American society. The goal of the film was to allow immigrant communities to participate in the creation of the media that tells their story. Through a combination of video diaries shot by the young men themselves and scenes shot by their teenage peers and filmmaker Mebrahtu, the film provides an honest and thought-provoking account of the complexities of acculturation. The film was shot in Mini DV format, and edited by Mebrahtu on a PC using Adobe *Premiere Pro*. *Dinka Diaries* was produced in association with various organisations in the US, including ITVS (the Independent Television Service) and NBPC (the National Black Programming Consortium), was co-produced with Philedelphia Public Broadcast channel WYBE, and was funded by the US Corporation for Public Broadcasting.

Scenes from *Dinka Diaries* (clockwise from top left): the film explains the background to the conflict in the Sudan which leads to Dinka like Joseph and Abraham coming to the States. Partly guided by audio cassettes of advice from their village elders in their former homes, they begin to assimilate, both into the community of other Dinka based in the USA, and also into wider American society.

available to us to make this work.

The first thing I do when I'm ready to present a piece for review is bounce the mix with the picture to a small Quicktime movie file, usually sized at 320x240 using the MP4 codec. This gives me reasonably good-quality audio and video for review purposes. This size may be small enough to email, especially if the scene is under a minute in length. However, some mail servers reject attachments larger than 5MB, and almost all email hosts limit the storage space available. Another method of delivery is to post your clip on your web page, and email your client a link to it. This avoids email-related problems, but requires you to *have* a web page, plus the knowledge of how to use web-design software, so it's not always the best alternative, particularly if you have lots of files to send, and scant time to send them in. What's more, this method won't help if your client needs to send *you* a revised video file quickly; some degree of two-way communication is desirable here. You can connect with your client via *iChat* or instant messaging and send the file that way, but that assumes your client is available to be on line. And I have found that the latter solution, while it gets around the file-size limits, is very slow.

For me, the best solution has been to use an FTP (File Transfer Protocol) server to exchange files. There are Internet file-transfer services that host these, or you can get one yourself. I have found Yahoo's Web Hosting to be a cost-effective solution, although there are many others as well. For US$12 a month I get five Gigabytes of storage, a web domain, a web site and email. I create a directory that the client is given password access to, allowing them to upload and download files at their convenience.

One extremely important thing to remember when using FTP transfers is to name your files appropriately. You can't use any illegal characters, such as dashes and commas, and you mustn't leave spaces in the name, although it's OK to use an underscore ('_'). Be sure to use the standard 'dot-and-three-letter' file extension after the name. For instance, don't just name your movie file 'my_masterpiece' and post it to the FTP; instead, if it's a Quicktime Movie, name it 'my_masterpiece.mov' to be sure the file won't be garbled by the server. And be sure to add '.mp3', '.aif' or '.wav' to your audio files as well.

Using this method, I've been able to upload very large audio files for clients to review or for post-production houses to download for mixing. And at times when a critical last-minute change is made to picture, the client can upload the fix to the FTP for *me* to download, so that I can make suitable changes to the score. It's a cost-effective solution that allows you to work from home and with clients anywhere in the world where there is high-speed Internet access.

For *Dinka Diaries*, I would bounce each scene to a Quicktime movie, post it to my FTP server, and then send Filmon an email advising that the files were ready to download. He would review them, then call me on the phone to give me his thoughts. This allowed us to work remotely for the majority of the work. It also facilitated his sending me a revised cut for one scene, as well. Most of my clients prefer this method, as it means they don't have to attend sessions, and yet they are able to communicate feedback and receive revisions almost immediately. ▶

Possible Mission 3

I'm going to cover three scenes from this film. The first contains the background information that introduces the three boys from the Sudan (you can download this scene from www.soundonsound.com/sos/soundandvision/dinkaphotosbefore.mov). As is common in documentaries, this scene consists of a number of still photographs that dissolve into one another, accompanied by narration. Filmon and I decided that I should score this scene using guitar and string orchestra, and in particular a nylon string guitar. The idea was to create a piece that was sympathetic yet not overly sentimental. As there is dialogue, it is important not to score a piece that would be too distracting.

The second scene follows one of the youths walking around the city of Philadelphia while listening on headphones to an audio cassette made by the elders of their former village, which imparts wisdom and advice to help them as they make a start in this new land (download this scene at: www.soundonsound.com/sos/soundandvision/villageelderbefore.mov). We hear the voice of the village elder speaking in Sudanese, accompanied by subtitles. It is here that Filmon wanted the blues guitar influenced by Ali Farka Touré (if, like me, you've not previously heard of him, try searching the Internet for samples, or look at your local library for his recordings). Filmon and I agreed that I would use an acoustic guitar, and I would play two parts in a conversational style, with the two guitars symbolically representing the youth and the village elder. Having the subtitles gave us a little extra breathing room for the music here, as viewers normally focus on the subtitles when hearing a foreign language. Despite this, though, you should be careful not to step on the voice.

The third scene involves a map, and can be downloaded for you to watch from www.soundonsound.com/sos/soundandvision/dinkamapbefore.mov. Maps are frequently found in documentaries, providing background and perspective to the story being told. In this case, it is through maps that the filmmaker explains the background to the conflict in the Sudan, the reason why the three boys in the film emigrated to the USA. Filmon and I agreed on strings and percussion here, both pitched and non-pitched. While the maps take us back to Africa, the music is not required nor intended to be of any particular African style. There is a voiceover, so you should be restrained in your scoring. Also, the maps represent the passage of time. The music should not plod along, but have some rhythm and movement to reflect that. You'll notice that the clip begins before we see the map. It is very common to overlap your score with the scenes before and after, allowing for a smoother transition. In this scene, the three young men are travelling by train from Philadelphia with their friends to meet with the Sudanese ambassador in Washington DC. The music should begin on the cut to the window looking outside the train, and continue to the scene where they arrive and get into a taxi.

Photo: Reel Voices/Howard Kellogg

Filmon Mebrahtu (with camera) working on an interview for *Dinka Diaries*.

If you're up for the challenge, download the movie clips, do a little research, and brush up on your guitar skills. And if you don't play guitar, consider collaborating with a friend who does. After all, if you're planning a career in music for picture, you'll have to handle a situation similar to this at some point.

Next Month

Next month, in the final part of this series, we'll take a look at how I scored each of these three scenes. Appropriately, we'll also talk about scoring the closing credits to a film — that is, if the producer hasn't already inserted a pop song written by his nephew! SOS

Preparing Mix Files

This box just goes to show how opinions can vary in the world of music production. Hilgrove Kenrick, who wrote the first half of this series based on his mainly UK-centric, television-oriented experience of music for picture, advised you to compress and limit all of your finished music to within an inch of its life before finally submitting it to your clients. Whilst I agree that this is reasonable practice when you're one of several people pitching for work, and you're trying to make your material stand out (by dint of it being louder), it's not something I'd advise for completed submissions.

So why the difference of opinion? After all, when preparing music for compact disc release, most composers nowadays mix their music as loud as possible, and levels are then often maximised, leaving virtually no headroom. Why should music for picture be different? Well, in film and television an entirely different set of circumstances is at play. In addition to the headroom needed to accommodate broadcast transmission (broadcast levels in the States do not usually rise above -10dB on a peak meter, to take one example of this), audio mixers must accommodate dialogue, sound effects and location sound as well as music. Consequently, I think it's best not to normalise your mixes, and avoid heavy compression or slamming your tracks with a peak limiter unless it is an essential part of your track's 'sound'. Post-production mixers will only have to lower the level of your track, so you might as well save yourself the trouble.

There's another reason why I think you should go easy on the dynamics processing with your music-for-picture work. I do lots of film and television mixing myself, and when I'm performing this role for clients, I know that I prefer to have the option to add any compression or limiting *myself* in order to make the track fit better in the overall audio mix. A track that is over-compressed or heavily limited might have to be pulled further back in the mix than was originally intended; remember, in music for picture, dialogue is king. And that's why I think that budding composers should let mix engineers do what they do best, while the composers stick to composition.

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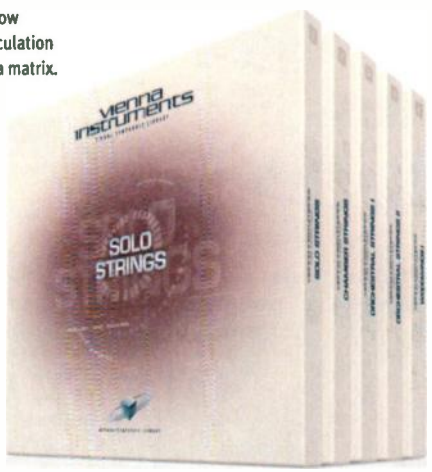
VSL's innovative interface for the *Vienna Instruments*. The patch list is displayed on the right (this space can be used to show various parameters via its top tabs), while the currently selected patch is shown at the top left. Below this is the 12x12 articulation matrix, in the centre is the selector ring, and at the bottom left above the virtual keyboard is a keyswitch note assigned to a matrix.

VSL Vienna Instruments



Virtual Orchestral Instruments For Mac & PC

The Vienna Symphonic Library are well known for their amazingly detailed collections of orchestral sounds, and now they're providing them as virtual instruments. Will this take the VSL concept to a new audience, or is it a step too far?

Dave Stewart

It's been a long ride. Since the first sample was recorded in December 2000, it's taken the Vienna Symphonic Library team over five years to release their definitive orchestral library. The unveiling of VSL's *First Edition* in

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VSL *Vienna Instruments*

pros

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cons

- The patches can't be edited.
- It's not possible to set up keyboard splits or alter Instrument playing ranges.
- Pitch-bend is limited to a realistic but limiting two semitones.

summary

Taken together as the *Symphonic Cube*, the *Vienna Instruments* look set to form a definitive lexicon of orchestral instrumental timbre in sampled form. Although their sophisticated, refined nature won't suit all tastes, they have applications beyond the strictly orchestral, and the sheer beauty of some sounds is a compositional inspiration.

2003 galvanised the orchestral sampling scene, and the gigantic 236GB *Pro Edition* which followed was even more stunning. Both of these enormous libraries combined exquisite musicianship with the pristine sound quality of VSL's specially constructed recording venue, the Silent Stage. Although lacking the reverberant 'glory trails' of a concert hall, this acoustic space has consistently yielded recordings with an incredibly low noise floor.

If one could always depend on VSL for quality, their release policy has been less predictable: rather than issuing a conclusive mega-edition as expected, the company began hiving off chunks of their sample database in a set of 15 themed titles called the *Horizon Series*. The Horizon *Opus 1* compilation offered an affordable entry point into VSL's lush orchestral world, while *Solo Strings*, *Epic Horns* and *Woodwind Ensembles* augmented the *Pro Edition*'s instrumentation. But this piecemeal approach was never going to satisfy VSL power users — like the believers huddled on the mountain side in *Close Encounters*, they knew something bigger was coming. Now, the long wait is over. Announcing its presence with a blast on its contrabass tuba, the *Symphonic Cube* has landed.

Even this, however, hasn't gone quite according to expectations — it would be more accurate to say that the *Cube* is in the *process* of landing, being beamed down to us in the form of 10 themed sets, the so-called *Vienna Instruments*, which will together comprise the 545GB *Symphonic Cube* — and at the time of writing, only the first five of the 10 are available, with the other five scheduled to be released by the time you read this. But, I hear you cry, what happened to the 'final hard disk edition' trumpeted by the Viennese baton-wavers since 2002? The short answer is that the company dropped the idea (if not the baton) and decided to follow the current trend towards virtual instruments, resulting in the creation of the new *Vienna Instruments Player* (compatible with Windows XP and OS X for Macs, and available in stand-alone, VST and AU plug-in formats).

When questioned on the change of direction, VSL say that by integrating performance control software and sample management, their new player can operate 'with more sophistication and intelligence than any sampler currently on the market'. I suspect another factor is that virtual instruments are harder to pirate than an unprotected sample library, but VSL prefer to accentuate the positive: according to them, the *Vienna Instruments* have a 'zero learning curve' and can even read your mind! We'll soon see about that...

Previous VSL Reviews In *SOS*

As each of the five currently released *Vienna Instruments* features a substantial amount of musical content from earlier VSL releases, you may want to read the reviews of these titles on the *SOS* web site.

VSL *FIRST EDITION*
W www.soundonsound.com/sos/may03/articles/viennasl.asp

VSL *PRO EDITION*
W www.soundonsound.com/sos/mar04/articles/vienna.htm

VSL *SOLO STRINGS*
W www.soundonsound.com/sos/sep04/articles/sampleshop.htm

VSL *CHAMBER STRINGS*
W www.soundonsound.com/sos/jan05/articles/sampleshop.htm

VSL *FRENCH OBOE*
W www.soundonsound.com/sos/jan05/articles/sampleshop.htm

VSL *WOODWIND ENSEMBLES*
W www.soundonsound.com/sos/dec04/articles/sampleshop.htm

Standard Attention

The first five *Vienna Instruments* (or *VIs*) to be released are *Solo Strings*, *Chamber Strings*, *Orchestral Strings I* and *Orchestral Strings II*, and *Woodwinds I*. The remaining five collections (*Woodwinds II*, *Brass I* and *Brass II*, *Harps* and *Percussion*) should be out by the time you read this. The *Vienna Instrument Player* is identical for each collection, so if you buy multiple titles you'll only have to install the *Player* once.

The installation procedure involves registering your collection's serial number, receiving an activation code from VSL, downloading a Syncrosoft licence, inserting a Vienna Key (a USB Syncrosoft dongle like the one used by *Cubase*), installing the *Vienna Instruments Player* and sample data and running a *Directory Manager* utility so the instrument can find its samples. Finally you're ready to rock, but after jumping through those hoops you'll probably feel more like a cup of cocoa and a nice lie-down!

In another break with tradition, VSL have not supplied individual printed manuals for the *Vienna Instruments*; each one ships with an identical slim booklet explaining the player's structure and installation procedure. However, registered owners can download PDF files containing complete patch and articulation listings for each collection from www.vsl.co.at/en-us/68/375/241.vsl. A less definitive listing for the non-registered curious amongst you can be found at www.vsl.co.at/static/vi_pop/shop_info_symphonic_cube.asp.

Editing controller assignments to switch between the articulations loaded into each cell of the matrix is easily done from the Control Edit screen in the right-hand display. Switches between up to 12 cells in any horizontal row are set in the 'H-Span' window, while switches between up to 12 vertically aligned cells are set in the 'V-Span' window. The 'A/B' keyswitch at the bottom can be set to change the up/down direction of grace notes, scales and glissandi. Shown at the top right of this screenshot is a list of possible controllers for articulation switching.

As well as the software instrument, each of the 10 *Vienna Instruments* collections includes a 'standard' and an 'extended' library — the extended one is twice the size of the standard and boasts a more elaborate set of articulations. The extended library is available to all purchasers for 30 days, but you can only gain permanent access to it by paying another fee on top of that required for the standard library (for

VI Second Opinions

When VSL announced the new *Vienna Instruments* and the *Symphonic Cube*, in common with many of my contemporaries, I phoned them up, told them to rob my bank, and left them to it. Was this wise? Well, the *VSL Pro Edition* had taken the film and TV market by storm; those that could afford it snapped it up, and those that couldn't tried to find a way to remortgage their grannies. What's more, there was the prospect of improvements. The *Pro Edition*'s stand-alone *Performance Tool* might have been powerful, but it always felt something of a half-baked afterthought, an awkward necessity. Thus the idea that the library would finally appear in its originally intended 24-bit format (the *Pro Edition* was dithered to 16-bit), with all the complex performance features handled by a new interface... it was simply too much to resist!

After the inevitable delays, the first collections (as reviewed here) arrived on my desk, and I duly ignored them. The *Pro Edition* had taken about a day and a half to fully install and decompress, so I was in no rush to tackle it all over again. But much to my surprise (and in contrast to Dave Stewart's experiences), when I did get around to it, I found installation to be swift and fairly painless compared to installing many other orchestral libraries. Whilst the DVD drive was chuntering away, I had enough time to load up the supplied Syncrosoft dongle with all the necessary licences. If you have the extended library installed (yes, I ended up ordering the lot...), there are two for each instrument.

Most of my first impressions are unprintable. I simply couldn't believe that a handful of people beavering away had come up with such a slick user interface that even a software dunce such as myself hardly needed to glance at the manual. What's more, the clever matrices, performance controllers and so on finally meant I could have a solo line on one track, switching articulations at will. Previously I'd ended up with several tracks for the different articulations and had to comp them down to one.

Moreover, there are so many different options for switching articulations — the speed you play at, different keyswitches, and momentary or linear controllers... Finding that virtually all are assignable is the icing on the cake. If you can't reach (or be bothered to program) your control surface, you just right-click on whatever you want to alter, to enable MIDI Learn mode, and wiggle the controller nearest to hand. Hey presto, for this instance of the instrument, that controller is mapped.

I feared that the *Vienna Instruments* would sorely test the resources of a single machine, especially with all the new articulations and matrices at my disposal, so I had already set up a spare PC with Steinberg's *Vstack* as a front end so I could load as many instances as resources would allow. However, when the *VIs* arrived, I was already running headfirst into a commission deadline, and I didn't have time to sort out the licensing for another computer — yes, *each computer* you use with the *VIs* requires its own Syncrosoft dongle loaded with all the necessary licences. Fortunately, VSL's ingenious new RAM-handling concept, which allows you to unload unused samples, came to the rescue. You program your track including as many articulations, dynamic layers and so on as you see fit, then put the instrument into Learn mode, play back the track, and hit the Optimise button. Every note, articulation, layer or dynamic that isn't actually used is removed from RAM forthwith. A window in the Perform page shows you the amount of RAM free and used. With the maximum possible 2GB of RAM assigned to the *VIs* (Mac users can apparently assign up to 4GB, for some reason...), and eight active instruments running, I had a paltry 32MB of RAM free, but running the Learn and Optimise functions immediately cleared out over a Gigabyte of data, leaving me spare capacity to load up yet more samples. It's quite astonishing.

In a nutshell, the *VIs* take VSL's trademark crisp sound to a new level. However, to my mind (and as strange as it may sound), with these instruments the beautiful sounds are almost secondary to the interface itself. Rarely have I encountered such a complex, powerful tool that's so instantly accessible, transparent and quick to use. Where previously you would forever be loading up yet another articulation (or would simply fake it if you were pressed for time), everything is now available at the prod of a controller. For once, an instrument interface feels like something that wasn't designed by a committee, but by a group of like-minded musicians and composers who understand what we want. Here, they've given us exactly that, with few compromises in the design or functionality.

This latest commission was the first time I let the *VIs* loose. Amusingly, the completed cue also featured VSL's *Pro Edition* (for parts as yet unavailable as *VIs*, like brass instruments) and East West/Quantum Leap's *Symphonic Orchestra Platinum*. In one fell swoop, VSL have managed to make their own *Pro Edition* feel clunky and unmanageable, and *Kontakt Player*-based libraries just a little, well... tame. I cannot remember when I was almost begging for new software or a library, but after using VSL *VIs*, I resent having to go back to the *Pro Edition* for my remaining palette. The rest of the collection simply cannot come soon enough. *Hilgrove Kenrick*

vienna instruments

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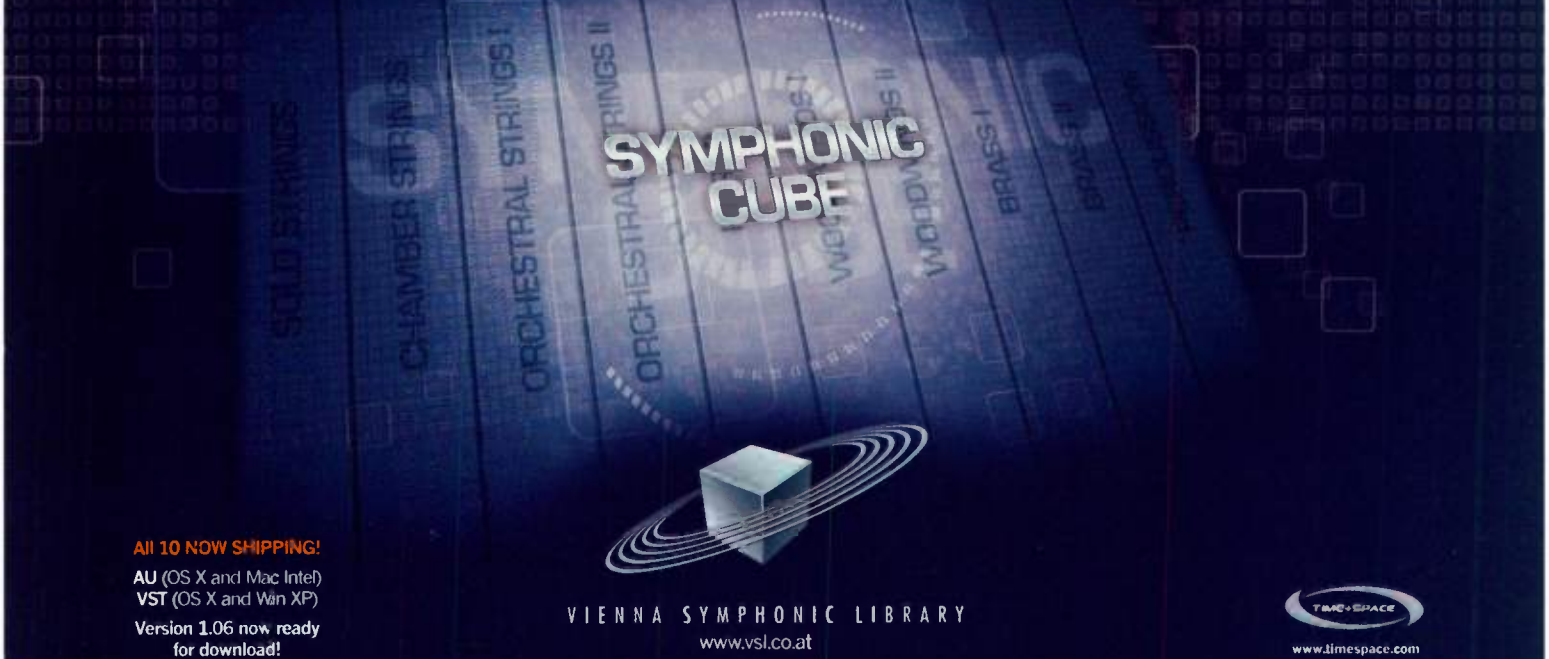

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more on the complex pricing structure of the *Vienna Instruments*, see the 'Pricing' box on the last page of this review).

Although the samples in the 10 *Vienna Instruments* have been reprogrammed from scratch and offer 24-bit resolution throughout, at least 50 percent of their musical material has already appeared in previously released VSL libraries. However, VSL are very open about this, explaining the provenance of the content in some detail at www.vsl.co.at/static/vi_pop/vi_overview_pop.htm. The 'Vienna Instruments' section of their web site also separates the articulations in each instrument into 'new' and 'existing' categories, and shows the contents of the standard and extended libraries.

The Interface

The interface used by all 10 *Vienna Instruments* (shown at the head of this review) has a somewhat nautical appearance which put me in mind of a smart Adriatic cruise liner. The main action takes place around the edge of the central 'porthole', called the 'selector ring'. Patches are loaded simply by dragging them over from the right to the left-hand window. Strangely, there's no 'MIDI channel select' option: the *VIs* operate exclusively in MIDI omni mode, something I've not seen since 1983! This is not a problem if you plan to run the *VIs* as plug-ins in your sequencer, but if you want to use multiple stand-alone players to play different parts, you'll have to use a virtual-instrument host program — VSL suggest Steinberg's *Vstack*.

The patches in the *VIs* can't be changed (although they do have programmable performance settings), so it's impossible to create keyboard splits or alter playing ranges. This posed an immediate problem: for example, I found the attack of the 'Strings sus vib' patch (a four-way split of double basses, cellos, violas and violins) too slow, and although its four components each have their own fast attack option, there was no way to map them into a playable combination, which seems like a backward step to me. Another limitation is that the pitch bend is restricted to two semitones, so you can't create extreme bend effects. Not an issue for those who stick to traditional orchestral sonorities, but I don't think pop/rock producers, sound designers and keyboardists who habitually set their instruments' pitch bend to more than a tone will be be too pleased.

Two patches layered within one cell.

Matrix Convolutions

Once selected, a patch occupies a 'cell' which has adustable volume, delay and envelope settings. The *VIs* can hold up to 144 such cells, arranged in a 12 x 12 grid which VSL call a 'matrix'. Only 18 cells can be seen on screen at one time, but the view area can be shifted. A highly flexible system allows you to swtich between articulations in horizontal and vertically-aligned cells by a variety of means: keyswitches, your keyboard's pitch wheel, a user-defined MIDI controller (mod wheel, footswitch, and so on), velocity (users can create velocity splits between two or more patches, although the patches' internal split points can't be changed) and Speed (of which more in a moment). The whole setup can be saved as a user matrix.

More complex setups are possible: a cell may contain two patches (as shown on the left) which can be layered or crossfaded, and a third 'parallel' cell can be added as a universal layer, which is a good way of adding a global attack (such as a sfz performance) to the matrix. The *Vienna Instrument Player* is not choosy about what type of instrument it plays (although it will not import sounds from other sampler formats), so you can switch between strings and woodwinds within a matrix, or even layer brass and percussion in a cell.

Up to 12 matrices can be assembled in a preset and selected via keyswitches. VSL gleefully point out that this system makes 1728 (12 x 12 x 12) articulations available to a single MIDI track — and while that's 1720 more than most people will ever need, it demonstrates that the *VIs* incorporate very powerful tools for dealing with the large number of performance styles that occur within an orchestral piece. Happily, as the *VIs* all contain countless factory matrices and presets, you can start to make music without having to do any programming if you prefer.

Mind-Reading Act

One aspect of the design of the *Vienna Instruments* is both innovative and musically powerful: we're used to the idea that key velocity controls volume, but in the *VIs* 'speed' refers not to the downward motion of the keys, but the rate at which successive notes

The Bringer Of Jollity

The *Planets Suite* is a popular work which requires an unusually large orchestra. Composer Andrew Blaney explains how he used VSL's massive library to create a sequenced version of *Jupiter — The Bringer Of Jollity* from Gustav Holst's classic score:

"My setup is a dual 2.5GHz Mac G5, two Mac G4s and two Pentium 4 PCs. Each computer runs Native Instruments *Kontakt* and the PCs also run *Gigastudio*. I like to have all instruments running in real time, which explains the need for five machines. I have the VSL *Pro Edition* complete package in *Gigastudio* format, plus VSL's *Horizon Series* orchestral titles.

"I copied and pasted zones from VSL *Gigastudio* patches into one big *Kontakt* patch per instrument/ensemble. I created large patches with up to 29 keyswitch groups for strings, woodwinds and brass (broken down into ensemble and solo instruments) so I could change articulations on the fly as I was recording. So during a take I was able to try out different combinations within a phrase."

UK composer Andrew Blaney in his home studio.

The arrangement uses 102 MIDI tracks, many of which show numerous keyswitched changes of articulation — for example, one piccolo part switches between staccato, sfz and vibrato short notes within the space of a few bars. It's clear that the extensive switching facilities of the *Vienna Instruments* could substantially reduce the number of separate tracks required for such a big score.

On the subject of VSL samples, Andrew says: "It was the sheer number of options available that made a lot of the *Jupiter* arrangement possible. It's written for a large orchestra, including two harps — unusually, the Vienna library has two — and I used just about everything VSL had to offer. That's the beauty of the VSL libraries — choice. Every time I hit a difficult passage that felt impossible to pull off with samples, a bit of digging around amongst the articulations nearly always delivered the goods. The demo took about 18 to 20 days."

Andrew's excellent arrangement can be heard at www.vsl.co.at/en-us/67/3920/4697.vsl.

are played. The Speed control can be adjusted so that slow playing will access (say) slow-paced legato samples, while fast playing will trigger quicker detaché notes. This unique facility produces excellent musical results, although I'd still stop short of VSL's claim that their instrument can read your mind!

Another great improvement is that the functions of VSL's *Performance Tool* MIDI utility are now seamlessly integrated into the player: so you can just load 'performance legato' and note-repetition patches and play without having to think about technicalities. This is a huge relief, as loading a template for each instrument was always a bit of a chore. And the performance functions now work multi-dynamically — previously, the *Performance Tool* could only work on one dynamic layer at a time.

Orchestral Strings I & II

On to the musical content. The *Orchestral Strings I* instrument (33.9GB) contains the 14-violin and 10-viola ensembles from VSL's *Pro Edition*, while the 27.6GB *Orchestral Strings II* features the accompanying eight cellos and six double basses. As well as updating the *Pro Edition* patches, both *VIs* include a substantial set of new articulations. Played by all four ensembles, the new 'short staccato' notes are considerably more brisk and businesslike than the longer staccatos from the *Pro Edition*, and impart a nice urgent 'zing' to rhythmic passages. VSL previously supplied staccatos in separate 'down bow' and 'up bow' versions, but the *VIs*' staccato, detaché, pizzicato and col legno articulations are programmed 'round-robin' style with four different samples per note. There are also new eight-way pizzicato repetition programs.

A new 'performance trills' style allows users to play their own trills, using intervals from a minor second to a fourth. Although reasonably lifelike, this style doesn't sound quite as fluid as a real-life trill, and unsurprisingly, it can't come close to reproducing the blurring effect of a string ensemble performing their trills at different rates. However, it's good to gain control over the speed and volume of a trill.

The violas' and cellos' '300bpm furioso runs' are quick, frenzied chromatic octave scales, fierce and bristling with bow noise, which will instantly energise arrangements, and the string-ensemble 'upbeats', which were previously only available for brass, are repeated fast notes tacked onto the front of a short note. The fast double upbeats sound like the galloping strings in the opening of the *William Tell Overture*, yet another useful and exciting rhythmic effect.

As mentioned earlier, VSL provide a set of basic 'strings orchestra' articulations which combine all four string ensembles (the patches are on *Orchestral Strings I*, but to hear the cello and bass samples you'll have to buy *Orchestral Strings II*). Since users can't create their own patches, it would be nice if VSL provided some more of these useful keyboard-split combinations — fast attack sustains and short staccatos are top of my wish list.

Solo Strings

VSL's admirable *Horizon Series* title *Solo Strings* duplicated the *Pro Edition*'s solo violin and solo cello samples and added solo viola and double bass. The new 82GB *Vienna Instrument* of the same name adds a fresh set of articulations to this material. I used the *Solo Strings* violin to put the *Vienna Instrument* through its paces, and its 'performance legatos' (now featuring two dynamic layers) sounded as smooth and convincing as ever. A new 'performance legato fast' style injects more zip into legato deliveries — using the Speed control to automatically switch between the regular and fast-bowed legatos in response to my rate of playing created an ▶

▶ even more lifelike violin legato performance.

New 'Zigane' (gipsy) legatos feature built-in slides between vibrato notes. When used on their own, these beautifully played samples produce a plaintive, singing vibrato style evocative of the Chinese erhu fiddle. I used the mod wheel to throw in these slides occasionally over the regular and fast performance legatos; the result was a very organic-sounding and enjoyable violin patch which could handle Irish jigs and reels as well as orchestral pieces. The same was true of the solo viola and cello.

Zigane-style grace notes are also provided, ranging from one to four semitones — these pitch-slide gracings are more slow and deliberate than the nippy conventional grace notes. All grace notes, scale runs and octave-glissandi patches have a built-in 'A/B' keyswitch to select 'up' or 'down' versions.

Chamber Strings

Like *Solo Strings*, the *Chamber Strings VI* (42GB) borrows its name from a VSL *Horizon Series* title, reworks its samples and adds new content. VSL orchestral titles are not known for their rock & roll attitude, but the chamber strings' 'harsh' note repetitions introduced here could change that — their attack is so vicious that you can almost see the horsehair flying off the bow. The only drawback is that the heavy, drawn-out bow attacks tend to make the samples sound a bit late, so you'll have to slide their MIDI notes back in your sequencer to make them play in time.

The six violins portamento (pitch-slide) legatos now have an extra quiet layer, which transforms an already emotive effect into something absolutely lovely. Once again, the pitch slides reminded me of an Asian stringed instrument, in this case the fabulous Indian sarangi. To see whether the *VIs* could cope with multiple layers, I layered the chamber violins, violas and cellos portamentos. They came up trumps: I ended up with three chamber string ensembles gloriously sliding between their notes in unison!

If you're looking for an astringent, hair-shirt type of delivery, all the chamber string sections now have non-vibrato sustains. By way of contrast, the 'espressivo vibratos' let it all hang out — the violins' and violas' vibrato is completely over the top, but the cellos show more restraint. Once again, I found the attack of VSL's four-way 'chamber strings sus vib' patch too slow for comfort, but the wide range and elegant, dignified delivery of the cellos' vibrato sustains makes them a good writing tool.

Many of the new articulations mentioned above are implemented in all four *VI* strings titles. A new, monophonic 'fast attack auto' option has been added to many sustained note styles; this allows separately articulated notes (including initial notes) to keep their original full attack, but legato playing triggers samples with shorter, fast attacks. The solo, chamber and orchestral strings now all boast a full contingent of natural and artificial harmonics (played on open and stopped strings respectively), the latter making a lovely gaseous, slightly wheezy atmospheric sound.

Into The Woods

Woodwinds I contains all the main ingredients for a woodwind arrangement: three of its four solo instruments (flute, clarinet and bassoon) starred in the *Pro Edition* and the fourth ('French oboe') has a *Horizon Series* title named after it. Each solo instrument has

Pricing

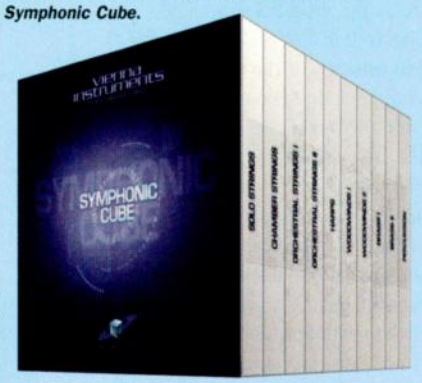
Costing the *Vienna Instruments* is a tricky affair; they're available separately, and when the remaining five are released shortly, they'll also be available separately, or in a package of all 10 *VIs* together as the colossal *Symphonic Cube*. To complicate matters further, the cost of buying the instruments depends on what VSL products (if any) you already own; as you might expect, given that much of the sonic content of the new *Vienna Instruments* has been released before, existing VSL customers who've supported the Library's efforts in the past are being rewarded with discounts. However, the cost of buying the standard *VIs* is the same for everyone, which has left some formerly staunch VSL supporters gnashing and wailing; the discounts only come into play if you purchase the extended versions of the libraries.

Fortunately, if you have access to the Internet, you can work out just how much your libraries and any extended versions you buy will cost you personally, including any discounts you may be due, with VSL's neat on-line Discount Calculator (see www.vsl.co.at/en-us/211/297/167.vsl). Here, it will have to suffice to give you an idea of the maximum these libraries could cost you; below are the basic prices for the standard versions of the individual *Vienna Instruments*, the basic (undiscounted) prices for the extended versions, and the undiscounted price of the entire *Symphonic Cube*.

A scene from a session at VSL's Silent Stage for the new *Horns I* and *Horns II* virtual instruments.

LIBRARY NAME	STANDARD LIBRARY	EXTENDED LIBRARY	FULL LIBRARY
Solo Strings	£230	£297	£527
Chamber Strings	£330	£397	£727
Orchestral Strings I	£330	£397	£727
Orchestral Strings II	£297	£364	£661
Harps	£117	£130	£247
Woodwinds I	£330	£397	£727
Woodwinds II	£284	£364	£648
Brass I	£330	£397	£727
Brass II	£330	£430	£760
Percussion	£264	£364	£628
Symphonic Cube	£2600	£3537	£6137

All prices include VAT.



The *Symphonic Cube*, which should be released by the time you read this.

a corresponding three-player section, all of which appear on the *Horizon* title *Woodwind Ensembles*.

The gimmick which won the French oboe its solo album deal is simple: unlike the *Pro Edition*'s Viennese oboe, this little fellow plays with vibrato. In this collection, the oboe shows its less endearing side with some raucous flutter-tongue samples reminiscent of an oboe played through a fuzz box with a flat battery. However, the oboe's new fast interval legatos are a great asset, allowing the creation of beautifully smooth, quicksilver reedy runs.

All instruments and ensembles in this 59GB collection have been fitted with the new 'performance trills' style. These looped deliveries are very versatile: using the Speed control to switch between performance legatos and the faster-speaking performance trills created a fluid-sounding solo flute patch which was great fun to play. The same approach also worked well for the bassoon and clarinet, though as I remarked when reviewing VSL's *First Edition*, some of the loud clarinet samples sound clipped to me.

Of all the *VIs*' new performance styles, I enjoyed the arpeggios (four-note broken chords played by solo flute and clarinet) the most. As well as demonstrating VSL's manic attention to detail (they're played in a choice of two speeds, up and down across the instruments' full range in all keys, legato and staccato, in major, minor and diminished scales!), these superbly played arpeggios have an irresistibly attractive, lively quality — once you hear them, you'll be hooked. The same is true of the solo flute's 'mordents', a wonderfully adaptable set of double grace notes.

Woodwinds I is an unmissable experience, but for the more luxurious shadings of alto flute, English horn, bass clarinet and the piercing tones of the piccolo, you'll also have to buy the sequel, *Woodwinds II*.

Grand Finale

It took me a while to get used to the switching capabilities of the *VIs*, so to say they have a 'zero learning curve' is a bit disingenuous. Having said that, the thinking behind their design is smart and logical, and the superbly nuanced, flowing performances they can produce are worth a little head-scratching. In terms of sheer volume of musical choice, the five *Vienna Instruments* released so far form a landmark collection by which all future orchestral products will be judged. I'll have to reserve judgement on the content of the remaining five *Vienna Instruments*, which I'll be reviewing individually over the coming months, but all the indications are good.

Newcomers to the field might find the *VIs*' performance options overwhelming, and the undiscounted prices of the full libraries are likely to strike fear into the bravest of hearts. But if you're committed to working with samples to reproduce the infinitely varied and subtle timbres, textures, performance styles and dynamics of a symphony orchestra, and you have time to plumb their depths, these instruments will yield fantastic musical results. Congratulations to Herb Tucmandl and the Vienna team — by sticking to their task so assiduously, they have created a wonderful musical resource. SOS

information

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Paul White

I recall being favourably impressed by the original IK *Amplitube* guitar amp modelling plug-in, and the new version 2 is far more than an update — it's almost a new product, with a fresh interface and new technology under the hood. A new modelling approach (DSM or Dynamic Saturation Modelling) has been developed to more accurately capture the way non-linear systems (such as amps that distort) affect a real-world dynamic signal, and the range of devices being modelled has also been extended. Plug-in formats supported extend to VST, Audio Units and Digidesign's RTAS, at sample rates up to 192kHz, and *Amplitube 2* requires Mac OS 10.3 or later or Windows XP.

Amplitube 2 allows you to mix and match preamp sections, EQ sections and power stages to create combinations that aren't available in the hardware world. Once past the rather nice guitar tuner, the signal path comprises four sections: stomp effects, the amplifier itself, the speaker cabinet and mic arrangement to which the cabinet is connected, and any studio rack equipment used for post-processing. A further advance is the ability to use two chains of components at the same time with a choice of series or parallel routing options, making it possible to create layered sounds or expansive stereo spreads. All this is accessed from a surprisingly intuitive user interface. What're more, if you're a little timid when it comes to sound editing, you get a comprehensive library of amps and amp/FX combinations with the program that covers all styles.

An optional hardware floor controller, the Stomp I/O, will soon be available but no additional hardware beyond your existing DAW setup is actually necessary to use the program, other than having an audio interface with a high-impedance instrument input for your guitar or a separate high-impedance DI box. (As ever, if you plug a guitar directly into a line input, the tone will suffer.)

Endless Possibilities

Amplitube 2 provides the user with a choice of 14 preamps, 14 amp EQs and seven power amps, which means a total of 1372 amp permutations before you even begin to tweak the controls, change the speaker cabinets or apply effects. If you count the effects, cabs and mic permutations, the number is astronomical — and you still haven't fiddled with any of the controls! The amp can be hooked up to 16 different speaker cabinets and miked via six different mic models (each with four placement options), though I was sorry to find no ribbon mic emulation as I've grown rather fond of those for guitar miking.

Prior to the amplifier you have a choice of 21 beautifully rendered pedal effects that sit on an equally beautifully rendered wooden floor, and if you need more processing, there are 11 rack-style effects that come at the end of the chain. Over 80 guitar amp and effect emulations are packed into *Amplitube 2*, with amp models based on the popular Fender, Vox and Marshall brands as well as a few boutique amps — as with most such products, though, there's no link or endorsement between the manufacturers of the amps being modelled and IK Multimedia. The same is true of the stompboxes, where the hardware modelled includes such classics as the Arbiter Fuzz Face, the Ibanez Tube Screamer, the MXR Dynacomp and Phase 100 and the Electro-Harmonix Memory Man delay.

Down The Tube

Amplitube II installs from a single DVD-ROM and comes with a USB dongle. This needs to be authorised and registered at the IK Multimedia web site, but if your studio computer is connected to the Internet, the whole procedure is fast and largely automatic. If you use a separate computer it takes a little longer but still is not complicated. The way *Amplitube 2* opens up is as a single window, with buttons that switch the middle section of the display between Tuner, Stomp, Amp Cab or Rack section views. In the upper strips are the usual Save and Load buttons, as well as

SOUND ON SOUND

IK Multimedia
Amplitube 2 **£224**

pros
- Good playable emulations.
- Good variety of amplifiers and effects.
- Easy to use.
- Supports most plug-in formats.

cons
- No tape echo emulation.

summary
Amplitube 2 is a very flexible and friendly guitar amp and effect modelling package capable of giving some great results. As with real hardware, though, you have to be prepared to fine-tune the sounds to get the best out of it.

arrow keys for cruising the presets, and of course there's a display showing the patch title. Eight rather small numbered buttons near the centre select from eight preset routing options, with the routing block diagram shown to the right, immediately above the section selection buttons. Bypass and Mute are to the right and do exactly what you'd expect.

At the bottom of the screen there are two more buttons that allow you to step through the various sections (Tuner, Stomp, Amp and so on) if you don't want to use the more direct buttons at the top of the screen. There's a section to access the plug-in's preferences, which contain oversampling and resolution settings that can be used to trade off quality against the amount of CPU overhead required; other utility sections include an input level control and meter, and a simple noise gate that sits at the front of the signal chain and has only threshold and release time controls. To the right of this is a window showing the value of the currently selected parameter, and there's also a simplified version of the tuner always on display so that you can check your tuning without having to call up the dedicated tuner window. There are also Pan and Level controls for the currently selected module, as well as a master output level control and meter. All the knobs and sliders can be automated from within the host sequencer to the extent that the host sequencer supports automation, with the caveat that if you use the same module twice in one of the signal paths, you can only automate the first one.

In the Amplifier window, there are separate menus for the preamp, EQ and power amp elements, and by default these are all linked so that when you change one, the others change appropriately for your choice of amplifier. However, you can turn off the Match switches to mix and match sections from different amplifiers.

When it comes to cab miking, you can have your choice of virtual mic on or off axis at either near or distant positions. These options are fixed, though, so you can't gradually vary the mic position. There's also a slider for room ambience that adds a nice sense of place and space to the sound. In the rack section, you can use up to four rack effects at the same time, and there's a good choice of modulation, delay/reverb, pitch, EQ, dynamics and enhancer effects from which to choose. Most of the effects are fairly familiar, but the pitch-shifting section includes the ability to create harmonies according to musical key and scale, providing you only feed in single-note lines. ▶

Alternatives

At a lower cost there is the Line 6 Tone Port, which also includes an audio interface and can produce excellent results on a par with the Pod XT. There's also the more straightforward *Guitar Amp* from Waves, which includes a special guitar DI box, and the incredibly flexible *Guitar Rig 2* from Native Instruments, which also has a hardware control option.

The Driving Experience

Using *Amplitube 2* proved to be extremely straightforward, and most of what you need to to do can be sussed without ever needing to read the manual. However, it is worth reading through the manual at least once, as it tells you important information such as the significance of using oversampling (or not) in the stomp and amp sections to avoid aliasing artifacts. There's also a lot of useful information about the various modelled amplifers and how their controls work, because as with any rigorously designed modelling software of this kind, the operation of the controls varies to match the device being emulated. I like the way the program is set out just as a stage setup would be — pedals first, then amps and speakers, and finally the rack processors that would be on the mixing console — as this makes the operating environment very familiar. The only slight departure from normality is the flexibility of the two-channel routing, but you don't have to use this if you don't want to.

As I so often find with programs of this kind, the factory patches are great for showing off the range of sounds that can be coaxed from the plug-in, but relatively few are things you'd normally use yourself, so you do have to take the time to set up your own patches. Furthermore, as any guitarist knows, even great amps can sound bad — it all comes down to finding the sweet spot on the amp that suits the instrument being used with it. I have several Strats and they all sound different, so when I change guitars, I also need to readjust the amp. It's the same with a program like *Amplitube* — time spent fine-tuning the amp settings to match the guitar really pays off.

A huge range of virtual stompboxes is available.

On the whole I found the amp sounds nicely responsive and satisfying to play, though I know there will always be contention, especially when it comes to emulating those really sweet, low-powered tube amps played just on the edge. Even so, I got some very nice quasi-country sounds by using the Brit Class A TB emulation with a compressor stompbox patched in front of it. The sounds sit well in a mix and the more overdriven sounds give off a nice sense of energy. Some of the preset names allude to well-known artists or tracks, and though I'd probably fine-tune them some more, they get pretty close providing you play in the same style.

I tried to recreate some of my personal favourite guitar sounds and was surprised not only by how close (and how quickly) I could get them, but also by the general playablity of the end result. These sounds record beautifully and sit well in a mix with no need for additional processing and the only slight cause for disappointment on my behalf was the lack of a tape-echo emulation. You get conventional analogue delays and digital delays, both of which sound great, but although there is a degree of tonal control on the digital echo which can get close to the sound of tape, it doesn't have that gentle wow and flutter.

Having played with *Amplitube 2*, I can see why IK Multimedia thought it would be a good idea to offer an optional floor controller so that guitarists could take it out live. Indeed, I'm almost tempted to try that myself. This being the case, it seems odd that there is no stand-alone mode — *Amplitube 2* always needs to run as a plug-in within a host program, and if you want a stand-alone amp simulator you'll need IK's *Amplitube Live*, which has yet to be updated to version 2.

Currently there are several software plug-ins that purport to model that elusive guitar amp sound, and I think *Amplitube 2* gets as close as any, striking a good balance between ease of use and flexibility. It's not the cheapest option around, but in my view it is well worth the asking price, and should meet the recording needs of most pop and rock players. SOS

There's a choice of speakers and mic positions, though the mic distance is not continuously variable.

information

- £ £269; upgrade from version 1 or *Amplitube Live/SE* £159. Prices include VAT.
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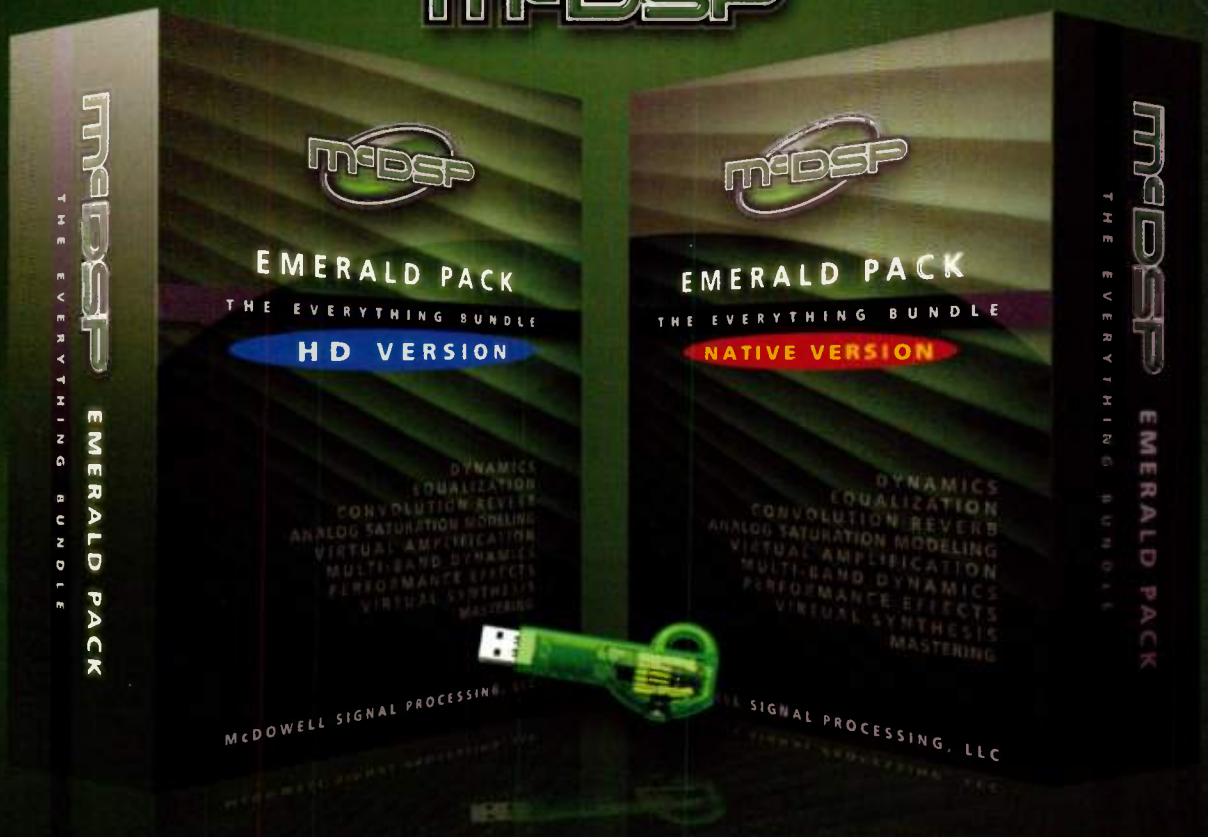

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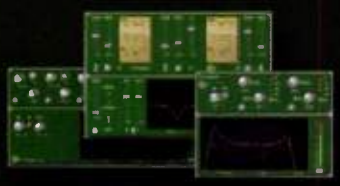

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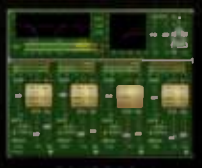

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Mix Rescue

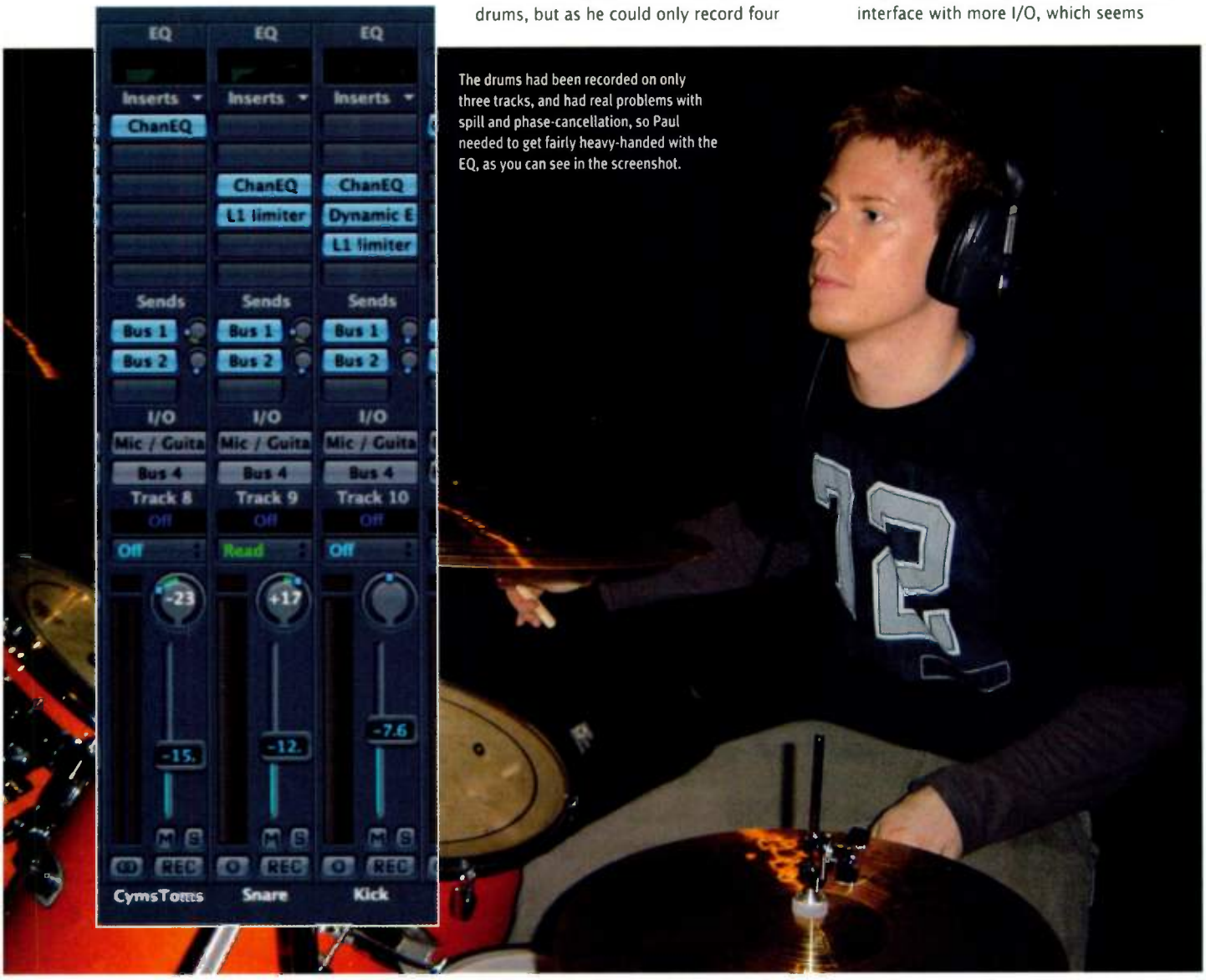
Ian McMillan's band recording presents some challenging drum parts this month, and there are also tips for better mixing of bass, guitars, and backing vocals.

Paul White

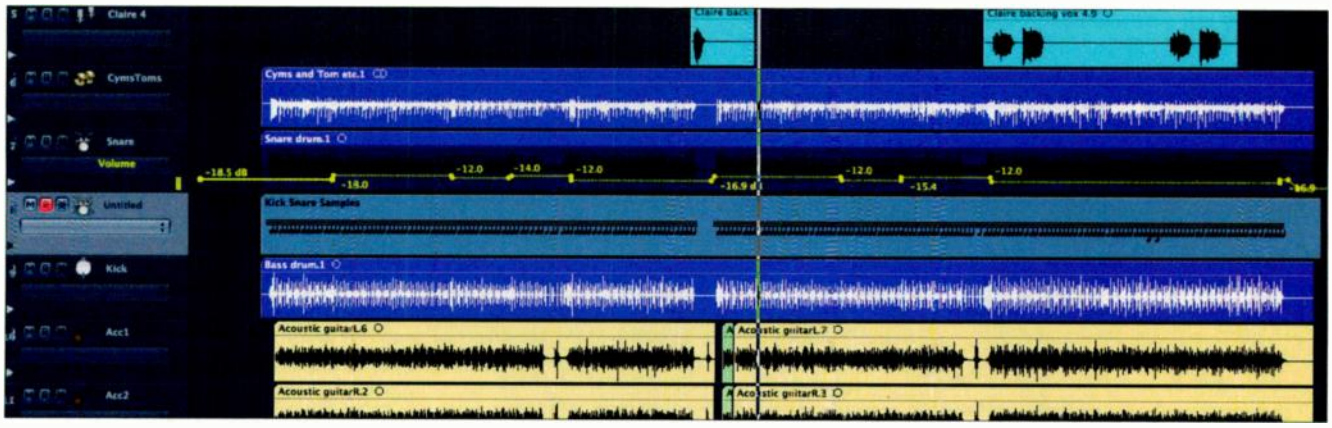
Steve Morano set up his band so he could start playing his own songs around the Oxfordshire and Berkshire area, and they have already had some interest from a few promoters, including one in New York. The track 'Believe' sent to Mix Rescue was made in Ian McMillan's studio (everyone knows him as Mac), called Groutfinger (don't ask!), which is mainly used by acoustic singer-songwriters from around the area.

Original Recording Setup

This song turned out to be from Mac's first band recording session involving acoustic drums, but as he could only record four audio streams at a time into his PC, he pre-mixed some of the drum mics, which gave me a few headaches at the remixing stage. He's currently looking for an audio interface with more I/O, which seems



The drums had been recorded on only three tracks, and had real problems with spill and phase-cancellation, so Paul needed to get fairly heavy-handed with the EQ, as you can see in the screenshot.



In order to get better-defined kick and snare sounds, Paul triggered samples from Toontrack's *EZ Drummer* in time with the live performance.

sensible for this kind of work. According to Mac, he used a single AKG Solidtube as a drum overhead, augmented by a Shure SM58 hung above the hi-hat. The toms were recorded via a stereo pair of SE Electronics SE3s, while the snare was recorded using a Shure SM57. The bass drum was picked up using a borrowed AKG mic, which was probably a D112. Dynamic mics are less than ideal for cymbals and hi-hats, because of their limited high-frequency sensitivity, but sometimes you just have to use what you have. For vocals, Mac used two SE3 mics routed via a Behringer DDX digital desk.

When the track was recorded, there were several vocal overdubs (all recorded using the AKG Solidtube) and extra electric-guitar parts, some of which the band didn't seem to have used in their own mix — they sent me everything anyway. The two electric guitars were courtesy of a Gibson Les Paul and a Gibson SG, both through Vox amps that were subsequently miked, though I have no details on exactly how. There's also an acoustic guitar part that forms the backbone of the song, and from the sound of the audio files I received, this was both miked and DI'd. Of the two tracks, I preferred the sound of the miked guitar, and so used mainly that for my remix. Given that the recording system could only handle four audio streams at a time, I assume a lot of overdubbing took place in making this recording, something which may account for some parts of the performance not being as tight as they might have been had the band been playing all together.

The track itself has an American pop feel to it and, given the source material and the recording I/O limitations, Ian had done a pretty good job on the mix, but I identified a few problems with the individual tracks once I auditioned them and felt I could make the mix a bit cleaner and tighter sounding. Firstly, some tracks were recorded at incredibly low levels, so I had to normalise them before I could start work, and where noise became a problem, I cleaned up the tracks manually by destructively silencing all pauses and fading out any decaying notes prior to pauses, so that any background noise also faded rather than stopping abruptly.

The acoustic guitar had a slight tuning problem that wasn't really evident when everything else was playing, but in the intro to the song, where nothing else was happening, it was very obvious, so I simply cut it out. I suggested to Ian that he get Steve to replay just the intro and send it to me to edit onto the song if he felt strongly about keeping it, but in the end he decided not to.

The bass guitar part seemed to suffer from a lack of picking confidence, so the sound wasn't quite as solid as I would have liked, and there were also some fluffed phrases and slight timing errors. Fortunately I managed to find good bits elsewhere and copy them in place of the dodgy bits, so getting a bass part free from obvious mistakes wasn't a problem.

As mentioned earlier, the acoustic guitar came as two audio files, one DI'd and one miked. The miked version suffered a bit of spill and noise from other sources in the room, but it sounded rather sweeter than the DI'd version from a musical viewpoint, so I used that with just a little of the DI'd version added in underneath, after processing the DI track through a clean guitar-amp model to give it a bit more focus.

For the drums, I was given three tracks called Kick, Snare, and Cymbals Plus Toms. As it turned out, the kick track had an awful lot of spill on it, and sounded almost like a single-mic recording of a complete kit, so I'm not sure where the mic was positioned. Similarly, the snare channel was almost overwhelmed by ride cymbal when the ride cymbal came in. The remaining track was also a bit of a mixture of the kit sounds, but it had a hard, filtered sound to it as well, as though it was the result of two or more mics at different distances being mixed to mono, causing some phase cancellation.

All the electric guitar parts were recorded with varying degrees of distortion and, from my viewpoint, the amount and type of distortion wasn't really ideal for this kind of song. I felt it was a bit thick and impenetrable, and perhaps a thinner-sounding guitar such as a Strat or Telecaster might have suited the song better, as it would have left more space in the mix for the other parts. However, the least-distorted guitar part also seemed to be ▶

Rescued This Month...

The Steve Morano Band is based around Steve Morano, a singer-songwriter who also plays acoustic guitar, electric guitar, and harmonica. Although originally based in Oxford and London, he has toured and performed extensively in Australia and now takes his 'country punk' songs out to audiences around the South of the UK, playing in the Reading, Brighton, and Aylesbury areas. The rest of the band comprise Ian 'Mac' McMillan (acoustic/electric guitars, backing vocals, percussion), Ross Nelson (bass), and Mark Acres (drums), but their friend Claire also helped out with backing vocals on this project. The band have just released a seven-track EP, and you can see them on the Guilfest Ents24 stage on 15th July.

W www.stevemoranoband.com

Need Help With *Your* Mix?

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Mono Or Stereo?

In his email to me, Ian said that he wasn't sure whether to record in stereo or in mono, or what the benefits were. In my view, instruments in a multi-layered mix like this (other than drums and stereo keyboards where used) are best recorded in mono, as it eliminates the risk of phase problems that can occur when you use two or more mics on the same source. I suspect that the odd sound of one of the drum tracks was caused by mixing two mic signals to mono — possibly the overhead and the hi-hat mic.

Acoustic guitars are essentially mono sound sources made stereo by their interaction with reflective surfaces in the performance area, so in the studio it makes perfect sense to record them in mono and then add stereo reverb or ambience when you mix. It is possible to use two mics to make a stereo recording, but the results are rarely natural, and you can end up with mono-compatibility problems unless you record using a coincident pair. The same is true of electric guitar, but because the electric-guitar sound is essentially artificial anyway, it is quite common to record using two mics at the same time, adjusting the relative mic positions so that the phase differences between the two contribute to the sound in some useful way. Once you add stereo reverb or ambience and pan the sound within the mix, it takes on an adequate sense of space without losing its focus.

Where you do record in stereo, as you would normally do in the case of a drum kit, it helps to limit the left-right panning to avoid the drum kit sounding as wide as the entire band. I tend to keep the pans between the 10-o'clock and two-o'clock positions for this reason. This argument also holds true for stereo keyboards or pianos miked in stereo, though you may want to offset the pans slightly to place the instrument to the left or right of centre.

▶ the main one, so I was fairly confident I could make that fit with a little tweaking.

In addition to Steve's main vocal, there were also four tracks of backing vocals by his friend Claire (who'd apparently never sung before), more done by Steve, and one track of Claire and Steve singing together. To my ears, Steve's main vocal and Claire's backing vocals were enough to create the right feel, and on scrutinising Claire's four vocal parts I felt that some sections worked really well while others sounded a bit weak, so I simply muted the bits I felt didn't work.

Drums Overhaul

While EQ could help me improve the drum sounds, I couldn't achieve the balance between the drums and Cymbals that I was looking for, due to the way the drum mics had been mixed prior to recording. In the end I resorted to loading in some kick and snare samples from Toontrack's new *EZ Drummer*, painstakingly lining these up, one beat at a time, with the drum hits I could see in the audio waveform display. Initially I used *Logic*'s Audio To Score facility to find the beats, but, because of the amount of spill, I had to add and remove a lot of beats manually as well as change the timing of individual hits.

I brought the *EZ Drummer* kick and snare up beneath the existing sounds rather than tying to replace or drown them, and this sufficed to restore some semblance of balance to the kit sound. I simply followed the timing of the original drum part rather than trying to improve anything. The original kick track then formed the main basis for the rest of the drum mix, with the snare track and, to a lesser extent, the phasey tom/cymbal track brought up to a level where the cymbals sounded reasonably well balanced. A gentle application of an Audio Ease *Altiverb* short ambience made the original drums sit more believably in the mix without making them sound over-treated. The addition of the sampled kick and snare left the toms sounding a little weak, but as these weren't available separately, there was little I could do about it and the song didn't seem to suffer much because of it.

Bass & Guitars: Reamping, Processing & Effects

To firm up the bass track, I reamped it via IK Multimedia's new *Amplitube 2* plug-in using the Bass Amp model, but switched the EQ section for a British EQ. I'm pretty sure the ▶

Some over-prominent notes in the bass track were brought under control with Apple *Logic*'s *Compressor*, and the signal was then reamped using the Bass Amp model from IK Multimedia's *Amplitube 2* to firm up the overall sound.

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Taylor Guitars K4 Equaliser

Dave Lockwood

This Rupert Neve-designed three-band EQ and preamp is intended for Taylor acoustic guitars equipped with the Expression System magnetic pickup array.

Taylor's K4 Equaliser is designed primarily to operate as a companion to the Expression System (ES) acoustic guitar pickup that is built into many of their instruments. The K4 provides a single channel, three-band EQ in a distinctive stand-alone, chrome-and-hardwood housing and, like the Expression System's on-board preamp/EQ, it was designed in collaboration with legendary console designer Rupert Neve.

A transformer-coupled XLR/TRS combi-jack accepts nominally mic- or line-level signals, but there's no gain control as such — the system's preset gain is set to suit the balanced, buffered, low-impedance output of a Taylor ES guitar. Actually, there's another 10dB of gain available if you turn up the output volume beyond its centre detent, making about 16dB available when using the TRS line input and about 50dB via the XLR, so you could use it as a mic amp on louder sources, although there's no provision for phantom power.

The output is via XLR at mic level and TRS jack at line level. Both outputs are balanced and may be used simultaneously, allowing the K4 to be used as an on-stage split, feeding both PA and an amp, for example, or PA and a direct recording feed. A dedicated Tuner output allows you to keep your tuner out of the signal path and picks up its signal right at the front end, remaining active whatever else you do to the controls.

Separate Send and Return TRS jack sockets for the switchable effects loop allow interfacing to remain fully balanced, avoiding the forced unbalancing of the more common

TRS, send-and-return insert point. By switching the EQ to post-loop and plugging into the insert return, you can bypass the gain stage and use the K4 just as an equaliser with line-level sources.

A phase-invert switch is there to assist in reducing feedback, or to match the DI signal's polarity to that of an additional mic on the guitar, and there's a ground-lift switch to eliminate ground loops, where necessary. A front-panel headphone jack with its own level control completes the connections. Power (3-15V DC) is via an external DC transformer or, in an emergency, two 1.5V 'C' cells that will run the unit for up to 10 hours.

Three-band EQ

The K4 has +/-10dB low- and high-frequency shelving filters, with 125Hz and 8kHz corner frequencies respectively, and a single, fully-parametric +/-10dB mid-frequency band, sweepable from 80 to 800Hz, with a x10 range multiplier. All of the gain controls in the EQ section are centre-detented with adequate legending, but only the extremes of the mid-band frequency control (80Hz and 800Hz) are marked. I, for one, would have found a few more markings helpful. The K4's configuration of two fixed bands with a wide-ranging parametric mid-band is a common enough design choice and perfectly effective in most applications, but given that the Expression System itself already has built-in high- and low-frequency shelving EQ, there is surely a strong argument to be made for equipping its companion equaliser with two fully parametric bands instead.

As it stands, the K4's EQ configuration is fine for recording applications, where the majority of your tonal shaping is likely to be of the creative kind, sculpting the response into the precise tonality you want. But in live performance, the EQ challenge is entirely different. Taylor's Expression System is one of the most inherently feedback-immune pickup systems out there, but at performance levels it can still feed back, even if only via the strings 'taking off' on certain notes. Notching out the guitar's primary resonant frequency will yield a few more dBs of additional level or a useful safety margin, but with only one sweepable band you are sometimes faced with a choice between creative or corrective applications when you really want both.

In Use

That aside, the K4 is a beautiful-sounding equaliser, with a warm, organic character that means the signal never sounds harsh or over-EQ'd, even at quite radical settings. With an ES-equipped Taylor 514 as the source, most of the time I found I only wanted to add or subtract a tiny amount of the body of the tone with the low-frequency band (depending on whether I was picking or strumming), add some gloss to the top end with a couple of dBs of high-frequency boost, and dip out a tiny bit of mid-range. The Expression System's magnetic string-sensor and body-sensor combination produces a fairly warm-sounding output anyway and the K4 seems to perfectly complement that. Using a guitar with a Sunrise magnetic pickup (via its essential SB1 buffer interface) produced similarly pleasing results at a range of settings.

There's a lot of overlap between bands — both high- and low-frequency bands extend well into the mid-range and the mid-frequency band spans more than the entire range of both. This generally makes it very quick and easy to get what you want, although guitars with under-saddle pickups — I also tested the K4 with an L R Baggs Element and a Fishman Matrix — fared less well, to my ears, than the Expression System guitar. The K4's fixed high-frequency shelving filter starts to rise at about 1kHz, reaching full boost at 8kHz. Consequently, when enough boost is applied to add some extra high-frequency sheen, the filter is also creating a bit of lift in the upper mid-frequency 'quack' zone. Of course, you can counteract this using the parametric mid-range band, but not if you want to be using it to drop out some of the mid-range!

However, equalising an under-saddle pickup is inherently compromised anyway, as you are essentially always chasing a moving target — the spectral content of the output varies with level sufficiently for your EQ setting for loud strumming to be completely wrong for fingerpicking, and vice versa. The 'four bands plus feedback notch' configuration of something like the ubiquitous L R Baggs Para DI — though that unit is undeniably not as sweet sounding as the K4 — seems perhaps to be a better tool for this job.

Conclusions

Although you can get fine results out of the K4 in other applications too, such as bass and electric guitars, I think it, unsurprisingly, does its best work with the source for which it was designed — an ES-equipped Taylor. However, the asking price would buy you a very nice, fully-featured channel strip with an on-board power supply, a variable gain control and dynamics processing, which would undoubtedly be far more useful in a variety of other applications. And, in the absence of a high-impedance input (1MΩ or so) suitable for passive electric guitars, and neither variable gain nor a variable high-pass filter, it is difficult to raise an argument for the K4 as the guitar-optimised recording system front-end that it might have been.

If you are a Taylor ES user, the attraction of the K4 lies in knowing that you will be getting the absolute best out of the whole system. It offers optimal impedance and level matching, high headroom and very low noise, in a format that lives up to the quality image of the brand, whilst also being robust enough to sit on the floor amongst the other DI boxes at a gig. The EQ delivers warmth and detail with pristine sound quality, albeit at a premium price. But if you are opting for a high-end, ES-equipped Taylor guitar anyway, why not go the whole way? SOS

SOUND ON SOUND

Taylor K4 Equaliser £749

pros
- Optimised for Expression System guitars.
- Smooth, easy-to-use EQ.
- High build quality.

cons
- No high-impedance input for passive instruments.
- Only one parametric stage.
- External power supply.

summary
A premium-quality preamp/EQ for Taylor's Expression System guitars, at a premium price. The absence of a few key facilities may limit its appeal in broader applications.

information

£	£749 including VAT.
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Unitonic Aurora 2

TDM Soft Synth For Pro Tools

Jem Godfrey

The first product from developers Unitonic is an unusual TDM soft synth with plenty of hidden depths.

I like to think there's not much that passes me by with regard to soft synths that are available for Pro Tools in one form or another, but I must confess that the original Unitonic *Aurora* did just that. So it was with great interest that I got my hands on the new updated version of this mystery soft synth, *Aurora* 2.0.

According to the press release, *Aurora* is a piece of software that fuses pure additive synthesis with frequency, pulse and noise modulation and non-linear wave distortion. Personally, I'm always a bit suspicious of synths that hide behind synthesis jargon as I'm not sure many people either (a) know what it means, or (b) really care. I'd much rather be stuck in a traffic jam than a linear vehicular anti-motion scenario generator, but that might just be me turning into a grumpy old man.

Currently, *Aurora* is only available for Pro Tools HD Accel systems running *Pro Tools* 6.0 or later under OS 10.2.2 and above, although Unitonic say that a PC-compatible version should be available by the time you read this. Unitonic also say there will be RTAS and AU versions in the not-too-distant future too, so pretty much everyone will be able to have a go. A quick check at the Digidesign web site also confirms that *Aurora* is compatible with *Pro Tools 7* too. Installation is simple, and *Aurora* is authorised to an iLok key, so

obviously you'll need an iLok and an iLok account before you can get cracking.

A Little (Northern) Light Music

Once summoned into existence, *Aurora*'s user interface is large and easy to read. It's quite a change from version 1's *Star Trek* look, opting for a more friendly and inviting grey-and-pastel colour scheme instead. The main page is split up into three sections. The upper section is the main edit area, where you'll find a selection of tabs running along the top that switch the panel beneath to the various control parameter pages — tone, waveform, envelope, key scaler, modulation matrix, sequencer, tuning and global control. To the right of this is a graphical display that provides a snapshot of each of your chosen waveforms and how they're being affected by the parameter changes happening on the left.

The middle section is a row of buttons that select the various parameters for editing in the upper and lower sections. This also enables the faders in the lower section to have access to one parameter, but applied to all the eight waveforms or, 'partials' in Unitonic-speak, that make up an *Aurora* patch.

The lower section is a context-sensitive mixer that directly relates to the middle section. You can Option-select multiple partials and perform group edits here, as well as muting and soloing. However, any changes you perform don't happen in real time; you have to keep re-triggering the key to hear the effects of your tweaking. It's not something I've encountered in a soft synth for some time, and although it's not exactly the end of the world, I did find it a little irritating after the hundredth key press.

Each sound in *Aurora* can be made up of as many as eight partials. Each of these partials is, in fact, comprised of two separate sound generators: a carrier wave (confusingly, also called a partial) and a frequency/phase modulator called a 'timbre'. Each can be assigned to a multitude of waveforms, and the carrier can then be duly modulated by the timbre. This can result in the creation of some very complex waveforms indeed. There is also the ability to load in your own custom waveforms to add further spice to the mix; once you've loaded in your audio, you can scrub along it until you find a suitably exciting moment, then *Aurora* can provide you with a handy spectrum editor that filters down the sound to remove any unwanted harmonics and distils the results into something rather more musical. In my case, I took a backing vocal I had knocking around and managed to mangle it into something almost resembling a granular lead sound akin to Propellerhead's *Maelstrom*. However, you do have to work at it, and I didn't find it a particularly quick process. On the plus side, I recorded the output of *Aurora* into *Pro Tools* as I was tinkering with the waveform, and that alone generated enough interesting audio to keep a sci-fi sound designer inspired for a week! Theoretically, you'll be able to get 128 voices out of *Aurora* using a single DSP chip, although the more partials you use, the greater the reduction in polyphony. I set up a whopper patch and still managed to get 16 or so voices to sound at once, which is more than enough for most *Aurora* applications.

Once you have your basic sound, you can the mangle it further with an impressive 12-stage looping envelope generator that works on each partial within the patch, should you so choose. You can have up to eight different parameters per partial controlled by their own envelope generators, including pitch, pan and amplitude. That means that with a full eight-partial patch utilising all eight envelopes per partial, you'll end up with a mind-boggling 64 independent envelopes

Alternatives

There are innumerable soft synths around, but surprisingly few that run on Digidesign's TDM cards. Of these, only DUY's *Synthspider* modular synth (reviewed in *SOS* December 2002) offers similar depth and programming potential to *Aurora*, though its architecture is very different. In the world of native synths, both Camel Audio's *Cameleon 5000* (April 2004) and Virsyn's *Cube* (September 2003) provide sophisticated additive synthesis capabilities, including the ability to resynthesize audio from WAV and AIFF files. Native Instruments' *Absynth 3* (April 2005) can also manipulate audio in interesting ways and is worth a look if you're into serious programming, but in terms of the sound it makes, NI's *FM7* is *Aurora*'s closest comparison.

SOUND ON SOUND

Unitonic *Aurora 2* $699

pros
- Not just another sample-based soft synth with all the same old sounds.
- Great experimental potential.
- Good graphical interface.

cons
- Buggy.
- Can sound a bit thin and cold.

summary
A synthesizer in the truest sense of the word with plenty of scope for development.

thundering along at the press of a key!

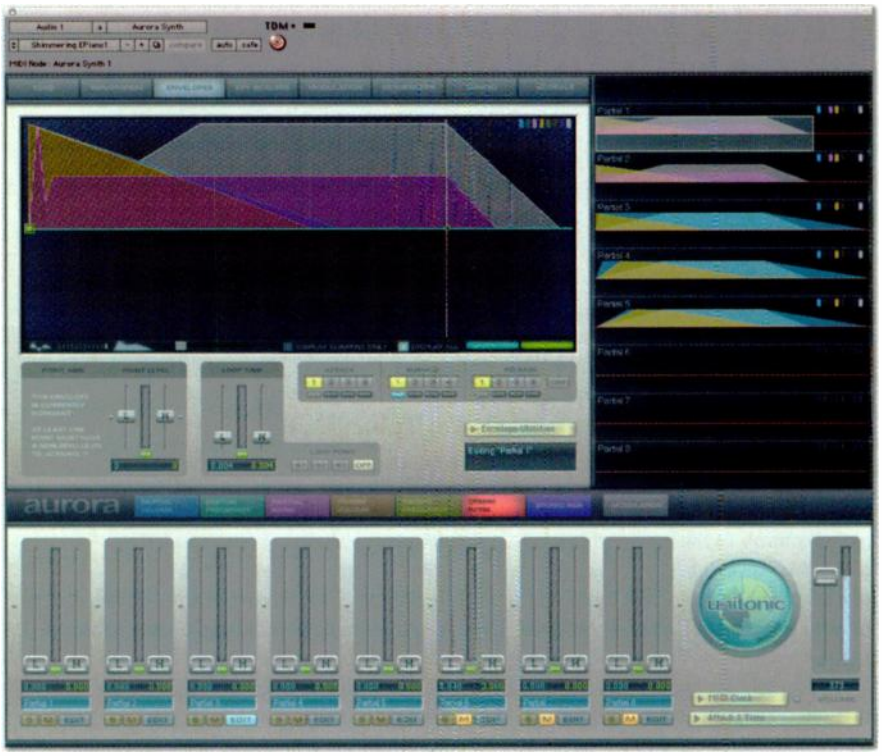
Aurora also sports a pretty comprehensive modulation matrix. Again the graphical interface is a real plus here: you just select the parameter you want and drag it to the desired level. In no time, I was sending one partial over to the left whilst another jumped a fifth and panned hard right, just by using the D-beam controller on my Roland V-Synth. Plenty of scope for fun and games here.

Each patch also has a 32-step sequencer that you can use to manipulate elements within the partials. Anything from filter sweeps to pitch information can be specified to give a patch yet more movement. Irritatingly, though, the MIDI clock switches back to internal rather than external sync every time you change presets, so if you want a sequenced patch to sync to *Pro Tools*, you have to reset every one that you load. In the end I gave up auditioning them and moved onto some other sounds, as this was doing my head in!

In fact, this brings me onto my main complaint with *Aurora*. In its current form, it's pretty buggy. For example, there's one bug where the volume levels can cause the synth to overload its own output; the bells section is particularly bad for this. In addition, changing the pitch of one partial to the strangely abstract increment of 999 (which equates to one octave up) when using the step sequencer caused that note to stop sounding. Also, if I held down some notes whilst changing patches, the resultant sounds could sometimes wildly differ from previous times I'd selected the same sound. I also found that volume levels differed considerably between some of the patches. The factory default sound is also set extremely high, so much so that the TDM plug-in overload indicator came on straight away when I played a note. Reducing the volume control rectified this, but I nearly leapt through the ceiling when I first tried it! Unitonic took my comments on board when I pointed them out and have assured me these problems this will be rectified when v2.02 is released shortly. There will also be a whole new array of patches included too.

Patching Things Up

Sonically, *Aurora* is capable of a great many different things. It can range anywhere from Minimoog-style lead and bass sounds through to some very digital DX-style noises and right the way on to some almost PPG-esque tones as well. Interestingly, the DX-style tones reminded me of my old DX21, the FM having more of a four-operator type sound than the six-operator character of the DX7. I also found that quite a few of the sounds benefited from reverb or chorus which I added separately, as they could be a little thin and harsh on their own.



Each *Aurora* partial can contain up to eight separate envelopes, and a sound can be made of up to eight partials.

You can load audio from *Pro Tools* into *Aurora* to serve as waveforms for the synthesis algorithms.

I couldn't honestly say that *Aurora* set me on fire, and at its recommended retail price of $699, you can pretty much have your pick of the soft-synth universe — although if you need one that can run on Pro Tools DSP cards, the choice is not wide. It might be a grower, though: the presets are a good starting point, but I feel *Aurora* really comes into its own once you start getting your hands dirty under the bonnet. This is a synth that will reward experimentation and produce some never-before-heard sounds, as long as you have some time to spare. SOS

information

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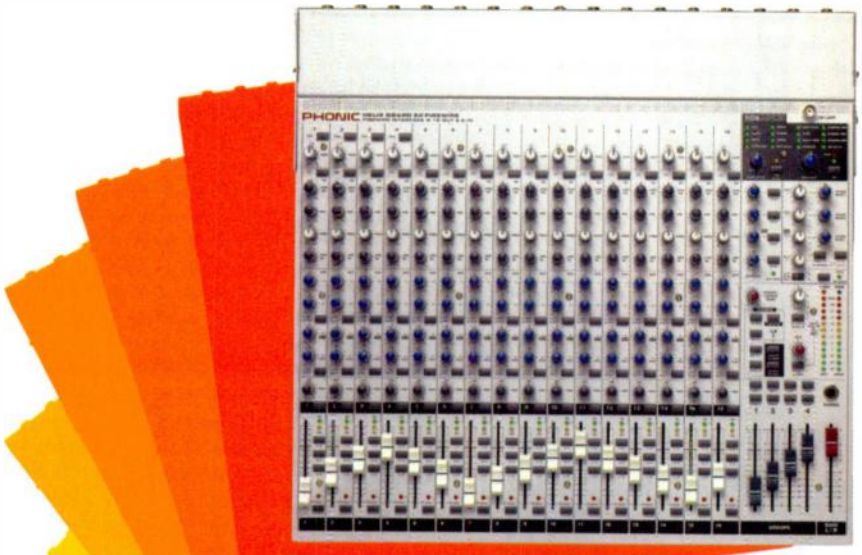

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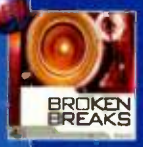

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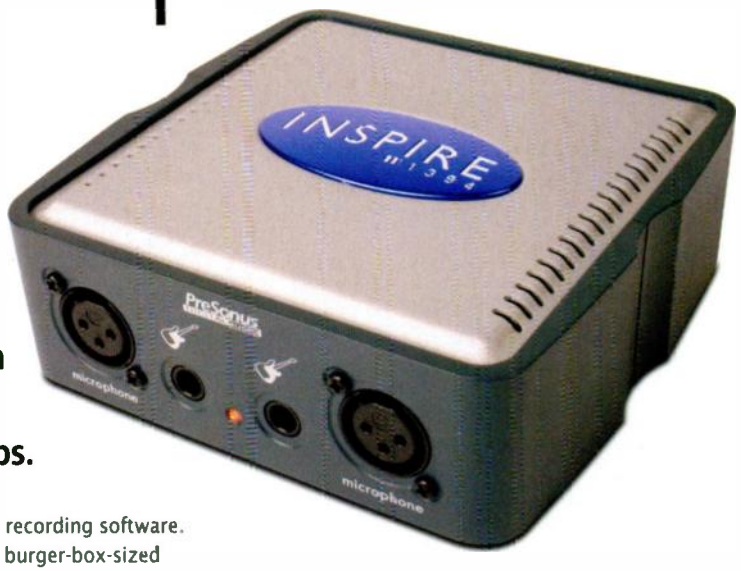

With their latest Firewire interface, Presonus promise high audio quality at an affordable price, thanks to the use of software controls instead of physical knobs.

Paul White

The Presonus Inspire 1394 is a compact Firewire recording interface that works at up to 24-bit/96kHz and provides four simultaneous inputs with a stereo output. Two of those inputs are balanced mic preamps with switchable phantom power and peak limiting, while the other two are on RCA phonos that can be switched from line level to RIAA phono for direct connection of a record deck. The Inspire 1394 works under Mac OS X or Windows XP, and as current versions of those operating systems support multiple audio interfaces, you can use up to four Inspire 1394s at the same time to provide more I/O where needed. Perhaps the most unusual thing about the Inspire 1394, though, is that while it has all the usual connections, it has no visible controls. That's become Presonus have managed to add more features while keeping the hardware manufacturing costs down by moving all the necessary controls to a software control panel. One distinct advantage of this approach is that the control setup can be saved so that when you return to a project, you can be sure all the settings are exactly as they were first time around. The Inspire 1394 can be powered from an external adaptor or parasitically from a six-pin Firewire socket, and works with all the common recording software.

Cosmetically, the burger-box-sized Inspire 1394 exudes desirability with its chunky but clean lines. The front of the unit houses the mic inputs plus two high-impedance instrument jacks for the direct connection of guitars or basses; the microphone and instrument preamplifiers draw on Presonus's analogue design expertise and are claimed to offer low noise and low distortion. Other than an LED that shows you the unit is alive and has digital sync, that's it for the front panel! Around the back are unbalanced RCA phonos for inputs 3/4 and for the stereo output, but there's also a stereo mini-jack (3.5mm) output and a further 3.5mm headphone socket. Other than that, there are just two six-pin Firewire sockets and a power input, so you can plug in the supplied adaptor if your computer only has a four-pin Firewire socket and thus doesn't provide power.

As I mentioned, the Inspire's uncluttered appearance has been made possible by putting all the necessary knobs and switches into a software control panel. Like most interface control panels, this allows you to set up the sample rate and buffer size (helpfully shown on the Windows version in terms of milliseconds of latency — on the Mac you set this up in the usual Core Audio way), but you can also set the input gain, switch the phantom power and limiter on and off, engage a 10dB preamp boost, and adjust headphone volume and main output level. There is also zero-latency input monitoring, with adjustable balance of the input source against the DAW output. All settings can be saved for reloading at a later date. This is, on the whole, a really good idea, but I still feel more comfortable with a physical analogue master level control when using active monitors, so that I can turn them down in a hurry if anything goes wrong or if the software crashes in a mode that feeds peak-to-peak white noise to the output! To be fair, modern operating systems are pretty robust in this respect but nevertheless...

Given the very attractive price of this unit, the bundled software that comes with ▶

Hardware And OS Requirements

Windows
PC with Firewire port, 900MHz Pentium or Athlon CPU (1.5GHz recommended), 256MB RAM (512MB recommended), Windows XP.

Mac OS X
Mac with 800MHz G4 (dual 1GHz recommended), 256MB RAM (512MB recommended), Mac OS 10.3.7 or later (10.4.2 or later recommended).

SOUND ON SOUND

Presonus Inspire £199

pros
- Compact and easy to use.
- Affordable.
- Includes a pair of good mic amps.
- Good selection of bundled Mac and PC music software.

cons
- May throw up a background monitor whine with some Mac G5 machines.

summary
The Inspire is a very neat combined preamp and interface for small systems, and can be used in multiples of up to four units where more I/O is needed. I particularly like the software control panel and the generous library of bundled software.

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it seems all the more enticing. The inclusion of Steinberg's *Cubase LE* is no surprise, and provides a perfectly competent sequencing platform for Mac and PC, while there's also Sony's PC-only *Acid XMC* for those who are more into loops. You also get Minnetonka's *Discwelder Bronze* for CD and DVD authoring, some *Discrete Drums* recordings of live acoustic drum parts for use in your own songs and 25 effects plug-ins. On top of this there's also around 3GB of drum loops and other samples on the included DVD. You really could just add a mic and a pair of headphones and start recording an album — though a MIDI keyboard would also be a distinct advantage.

The Inspire can be used to connect a turntable to your computer, thanks to its RIAA phono inputs.

Inspiration Or Perspiration?

I tested the Inspire on my Mac system running OS 10.4.3, where the unit was recognised by Core Audio without fuss. The control-panel software installed first time, after which I was in business. In most respects, the Inspire worked flawlessly and intuitively, but there was a problem with my particular Mac G5 — albeit one which has also tripped up a few other interfaces and at least one USB microphone. This manifests itself as a low-level digital whine, which can be heard over the monitors but isn't recorded to your sequencer. Isolating the Inspire's output using a stereo DI box with ground lift immediately solved the problem, so it seems to be due to a ground loop between the Firewire cable and the audio ground, caused by the way the G5 is wired internally. My MOTU 828 MkII interface works well enough in the same setup, but it is common knowledge that some G5s cause monitoring noise issues with some audio interfaces, whether Firewire or USB. Often the problem can be solved by switching off Processor Idling, for which you need to download the CHUD utility from the Apple web site. CHUD is part of a free Developer Tools download, but unless you're unlucky enough to have one of these awkward G5s, you shouldn't need it.

Having the controls on screen is in many ways more friendly than having them on the box, as you can put your interface wherever it is convenient to do so, and don't need to adjust anything. What's more, the settings are remembered by the software control panel so you can call them up later. The mic preamps are gratifyingly clean, and though not esoteric, they perform as well as the preamps on a decent mid-price mixer and gave me no cause for concern. I liked the software-controlled internal mixer for zero-latency source monitoring, which again is very easy to use, while the free software and samples provide a thick layer of icing on the cake! The *Discrete Drums* samples are particularly useful if you need the sound of real drums, as they are multitrack versions of real performances in various styles, and very good they sound too.

Thumbs Up

With the exception of the background monitoring noise problem that my Mac G5 seems so keen on bestowing on selected hardware, the Inspire worked perfectly, installed easily and sounded very clean indeed. The software control panel is a great idea and it really works, while the bundled software is very welcome, even if not quite all of it works on both the Mac and PC platforms. It's also good to know that you can use multiple units to get more I/O if necessary, though if you know in advance that you're likely to need more I/O, there may be better solutions available to you than stacking up four Inspires.

Over the past few years I've come to really respect Presonus gear and the very affordable Inspire lives up to the design standards they've set with their more expensive products. The interface market is pretty keen at the moment so I won't say that this is the only contender for your money, but if you need a compact four-in, two-out solution with a pair of decent mic preamps built in, the Inspire has to be on your shortlist. SOS

Almost all of the Inspire's features and options are set up via the software control panel.

Specifications

- Up to 24-bit, 96kHz recording.
- Driver formats supported: Core Audio (Mac), WDM and ASIO 2.0 (PC).
- Dual microphone/instrument inputs.
- Dual line-level (-10dB) inputs via stereo RCA phonos, with optional RIAA turntable setting.
- Stereo line-level outputs on RCA phonos or 3.5mm stereo TRS jack.
- Stereo headphone output on stereo 3.5mm TRS jack.
- Two IEEE 1394 (Firewire 400) ports.
- Auxiliary power input jack.
- THD+Noise: <0.008 percent.
- Signal-to-noise: >95dB.

Alternatives

The market for small recording interfaces is a crowded one, especially if you consider USB as well as Firewire devices. Among the many devices on offer, M-Audio's Firewire Solo is a four-in/four-out interface with one mic preamp and one high-impedance instrument jack; it also boasts balanced, full-sized jack outputs and S/PDIF I/O, but lacks the Inspire's soft limiting and RIAA turntable input. The same company's Fast Track Pro delivers up to four-in/four-out functionality via USB, with two mic preamps that double as instrument inputs, S/PDIF and MIDI I/O and balanced outputs, but again, no limiting or RIAA input. Edirol's UA25 is a USB interface with two mic preamps which, like the Inspire's, have phantom power and built-in soft limiting; it also has MIDI and S/PDIF digital I/O and full-sized output jacks, but no turntable input, and is only a two-in/two-out device. Finally, Terratec's Phase X24FW is a little more expensive, but offers two-in/four-out recording via Firewire with two phantom-powered mic/instrument preamps, analogue insert points, S/PDIF and MIDI I/O and support for 192kHz, although again there's no limiter or RIAA input. One feature of the Presonus Inspire that appears to be unique in this bracket is the ability to use up to four of them simultaneously.

information

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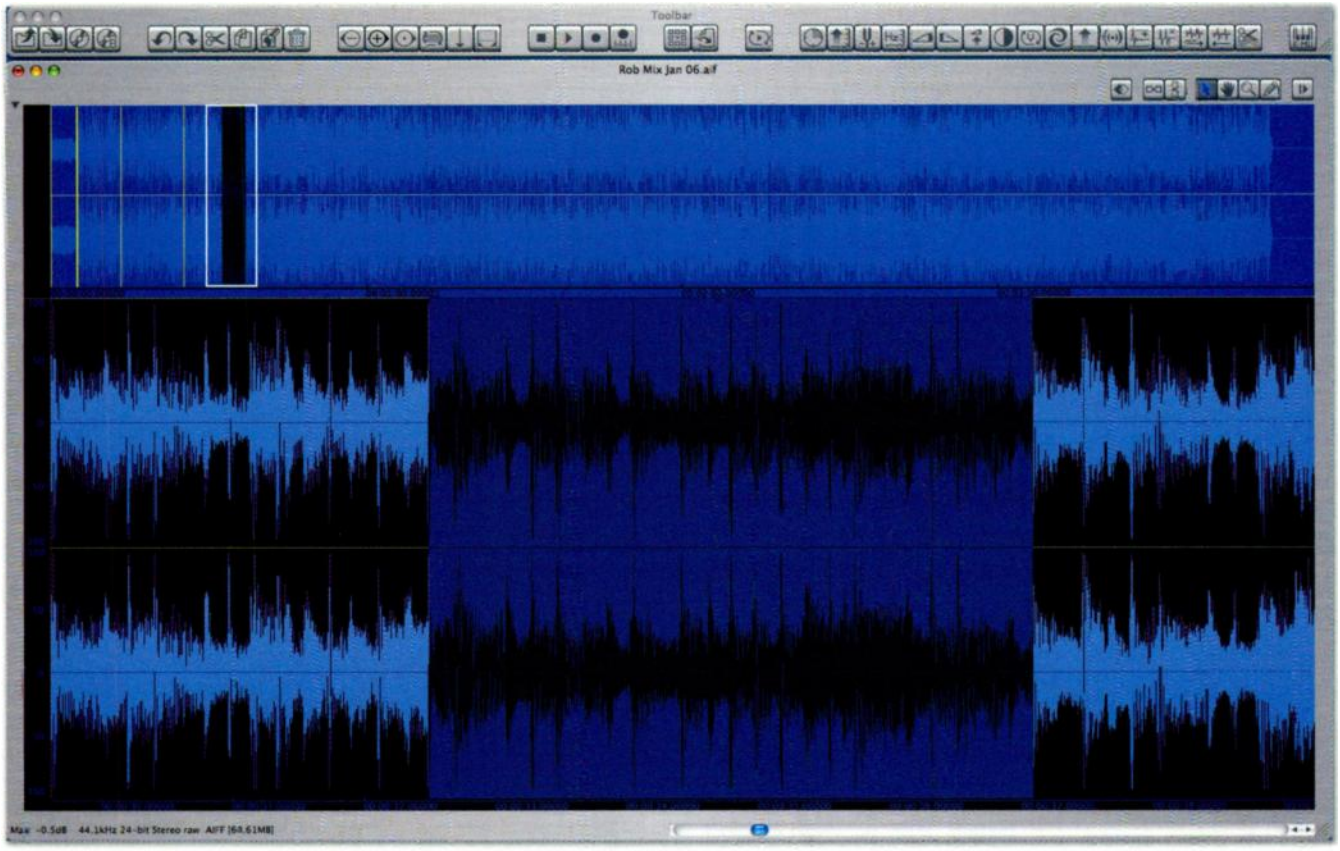


The main waveform-editing screen, where regions are selected and markers placed. The complete file is shown at the top of the display, while selected sections can be zoomed in the lower area.

BIAS Peak Pro 5

Audio Editing, Processing & Mastering Software for Mac OS X

This major update to *Peak* includes a new Playlist editor, full AU plug-in and virtual instrument support and a heavy-duty suite of mastering plug-ins.

Paul White

I often describe BIAS' *Peak* as a 'Swiss Army Knife' of a program, as it goes far beyond the stereo audio-file editing capabilities that are at its heart. Certainly, if you simply want to trim or edit audio files and then produce a pre-master for CD duplication it will do the job perfectly, but it is also a powerful sound-design tool popular with sample developers and film sound-effects creators. Numerous sophisticated DSP sound-processing functions (which seemingly increase in number with every incarnation of *Peak*) are built into the program, but third party plug-ins, including virtual instruments with the latest version, may also be used. *Peak* can also handle numerous batch-conversion operations, which can be a real time-saver in a commercial situation, and there are several tools for improving or rescuing damaged audio files. These features, along with a comprehensive graphical interface, have helped to keep *Peak* at the forefront of audio editing on the Mac platform. However, far from being complacent, BIAS (Berkeley Integrated Audio Systems) have radically overhauled the program, adding enhancements that include comprehensive Audio Unit (AU) plug-in support and a brand-new Playlist section. In the *Peak Pro XT* version they've also bundled the six heavy-duty mastering plug-ins that comprise the *Master Perfection Suite*. (*Peak Pro 5* is essentially the same program but without the *Master Perfection Suite*.) As with the earlier version of the software, audio files of up to 32-bit, at whatever sample rate your hardware can deliver, are supported. All edits are non-destructive until a project is saved, and there's unlimited Undo/Redo, with a full Undo history. I don't propose to cover all of *Peak*'s many attributes again during this review of the update, so for more info on what *Peak* is already capable of, check out our review of *Peak 4* in *SOS* May 2004 (http://www.soundonsound.com/sos/may04/articles/biaspeak4.htm).

If you don't need the *Pro* version of the sotware, the budget *Peak LE* offers similar editing features and, like the flagship version, has been updated to include movie support (more details overleaf) and *Peak*'s

newly designed Red Book-compliant Playlist, complete with all the necessary PQ subcode editing, ISRC (International Standard Recording Code) entry and CD Text features for CD burning. Missing from *Peak LE* is the new advanced waveform view in the Playlist — a key feature of the professional edition.

On the plus side, *Peak LE* features the same high-quality sample-rate conversion used in *Peak Pro* (although the highest quality setting is capped at '4', as opposed to the '10' of *Pro XT*). According to BIAS, this is still very good, and better even than that offered by *Peak 4*.

The updated *Peak* engine also includes Quicktime Movie/DV-clip synchronisation, plus a new Change Duration DSP process for adjusting the length of audio clips with minimal side-effects and without affecting pitch. The Change Duration feature in *Pro 5* offers an additional tempo-envelope option, and the same technology is used to provide a high-quality Change Pitch DSP function. For easier editing, there's also an Auto-Define Tracks tool that can automatically divide up album-style material into separate tracks by detecting pauses. If all this sounds impressive, check out the full extent of what's been done to *Peak Pro* at http://www.bias-inc.com/products/peakFeatures/. For a limited time, bundles offering free extras in all versions of *Peak* are available. Again, full details are provided on the BIAS web site.

Installation & Copy Protection

The first obvious difference during installation of *Peak Pro 5* is in the copy-protection scheme, which now uses a USB dongle. This is authorised on-line when you register the software and allows the program to be installed on multiple machines, as all you then need to do is plug the key into the machine you wish to use. Clearly, this is good news if you tend to move between a laptop and a studio machine. My dongle is sticking out of a USB hub along with several others and appears to work perfectly happily. Just prior to going to press, BIAS informed me that when the forthcoming *Peak Pro* 5.2 update is released, the USB key will be optional and the program, as well as the additional plug-ins included in the *XT* edition, will be protected by default using a new software authorisation key. This means that a USB key will not be required, which is good news for dongle haters, but it will be available for a nominal fee if you prefer a hardware key.

SOUND ON SOUND

BIAS *Peak Pro 5* **£389**

pros

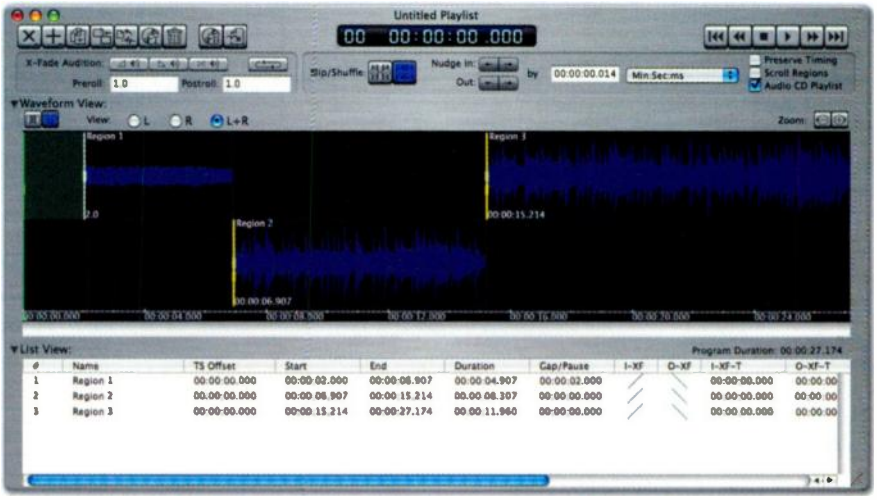
- Extremely flexible editing, mastering and sound-design package.
- Can be configured to work the way you want.
- Versatile plug-in handling, with AU and VST mix-and-match ability.
- Great audio quality.

cons

- The sheer number of available tools and key commands can be hard to remember.
- The Playlist crossfade editor could do with a more direct means of access.

summary

Peak is clearly the most advanced and versatile Mac stereo editor and sound-creation tool currently available. The mastering plug-ins available in the *Pro 5 XT* version are definitely worth the extra expense and could well save you a lot of money on third-party equivalents.



The updated Playlist editor, showing the newly introduced waveform view.

Playlist & CD Production

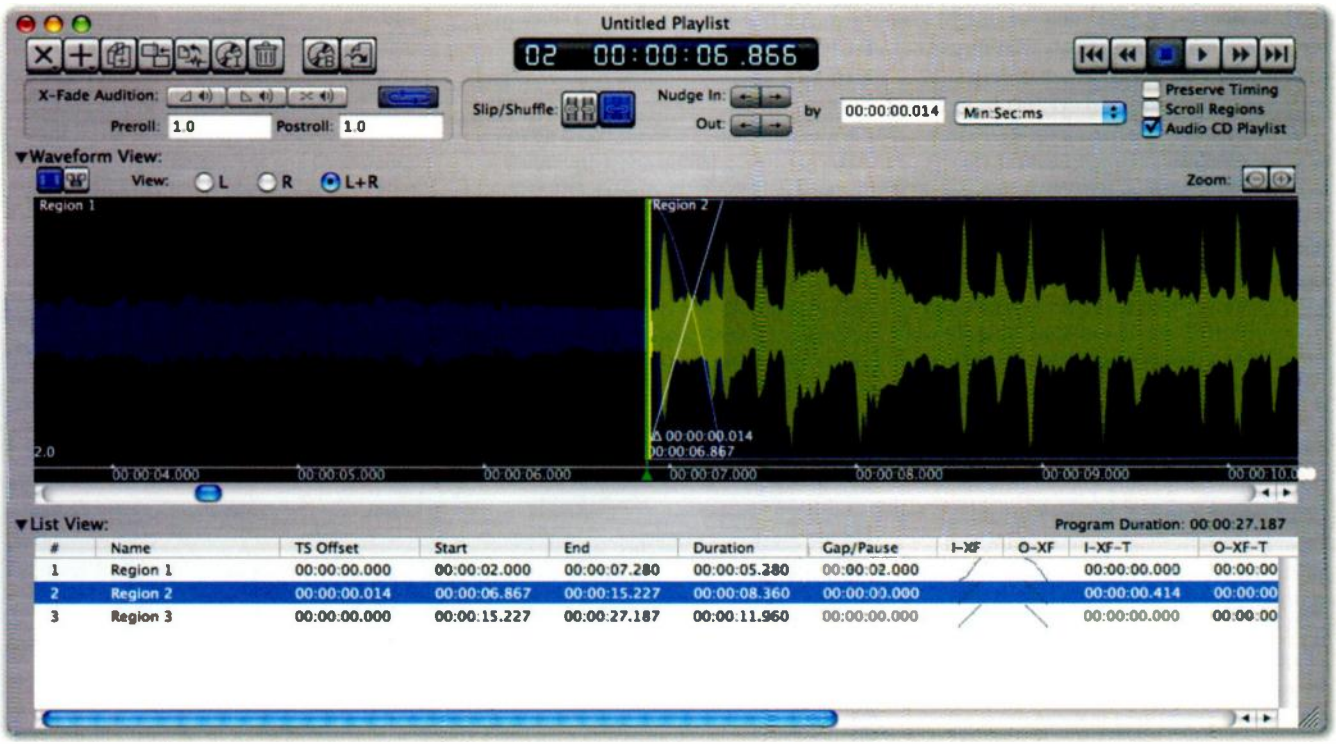
As with most updates, some of *Peak*'s new features and capabilities are added to menus and so may not be immediately obvious, while others take the form of new windows where it is immediately apparent that something has been added or changed. One such area is *Peak*'s Playlist, where the newly designed List view window has been augmented by a graphic waveform view directly above it, showing staggered or linear views with object transparency, allowing crossfades and tracks to be adjusted visually (see screen above). You can zoom in or out much as you can in the main window, and drag region boundaries as required. Several styles of crossfade are available and customised crossfade versions can be saved for future use.

Roxio's *Jam* features a very nice graphical window for adjusting track gaps, fades and start markers, and BIAS have had to add comparable features to *Peak* now that it no longer relies on *Jam* for its playlist and album-editing features. BIAS stopped including *Jam* because both *Peak Pro* and *LE 5* now support full Red Book disc-burning directly. They don't all work in exactly the same way, but the end result is much the same, including the ability to dither while burning. Other important differences are that *Peak* uses the Apple disc-burning engine and *Jam* doesn't offer such comprehensive editing and processing capabilities. Having said that, in order to maintain compatibility *Peak* continues to offer *Jam* Image import and export.

The revised Playlist editor can be used to sequence multiple audio files or regions, with flexible options for crossfading between items. The POW-r dithering algorithm is the same one employed by *Logic* and *Pro Tools* and is well regarded.

Essentially, the designers have tried to ensure that they've at least covered all the options offered by *Jam* and then added some more of their own. The Playlist offers unlimited Undo/Redo, plus keyboard trigger functions for auditioning and nudging audio segments and edits. Because the Playlist burns files that are Red Book compatible, they may be used as the source for professional duplication and they include ISRC entry windows, CD track indexes, PQ subcode editing, CD-Text, and so on. *Peak*'s own *V-Box* plug-ins may also be applied to the files within a Playlist as part of the burning process, which can save a lot of time if you want, for example, to make a CD out of some noisy audio (say, an old recording) and de-noise it as you burn it.

Of course, audio-format CDs are prone to errors and on most systems you have no way of checking these, so the first you know about them is when the pressing plant throws your master back at you and says the block error rate is too high to work from. To get around this problem, *Peak 5* ▶



Within the new Playlist, there's extensive control over crossfading between regions and the placing of track-start markers.

Pro has an extension that supports the standard 'DDP' file-export format. Many pressing plants prefer to receive material in the robust DDP data format, so the availability of this extension is a very welcome addition for anyone preparing pre-masters for duplication. Although it will normally be a paid-for option, I understand that the extension will be offered free to *Peak XT* purchasers for a limited period.

Plug-Ins

The plug-in side of *Peak* follows the same philosophy as it did in earlier versions of the program, using both inserts and a graphical effects (V-Box) routing window, but it now supports Audio Units plug-ins as well as VST throughout. Bouncing and virtual instruments are now also supported in these two formats (instruments are playable either via external MIDI or from an on-screen virtual keyboard), and there's an automatic latency compensation facility for Audio Units and VST effects.

A further benefit of *Peak*'s plug-in management facilities is that both VST and AU plug-ins used in the V-Box matrix (once saved as a user preset) can be opened in other programs, even though the other programs may not support both formats. I'm told that this function is technically not the same thing as a wrapper — it simply supports both AU and VST effects and instruments. In practice, this generally works fine, but I did have 'unexpected quitting' a couple of times when combining plug-ins from different sources and in different formats. This is not really surprising, as not all third-party plug-ins are happy to play nicely together.

BIAS were one of the first software companies to include a convolution reverb process as standard, which is one reason why earlier versions of *Peak* became so popular with film sound designers. Convolution may be used both to create 'sampled' reverbs and to 'convolve' one sound with another, to produce something completely new and abstract. Peak's *Impulseverb* has had a cosmetic update and gain controls have been added for both the impulse response and main audio source. Other than that, it is functionally similar to the previous version.

DSP Processes

Some of the improvements in *Peak Pro 5* are hidden under the hood and are therefore less obvious — for example, the refined time- and pitch-manipulation algorithms that include a new transient mode to help prevent artifacts when percussive sounds are being processed. BIAS have always been very concerned about the quality of sample-rate conversion and their SRC routine now appears to be one of the best available. (After all, there's no point in recording at high sample rates and then using an SRC to provide a 44.1kHz CD version if the SRC is going to negate any benefits that recording at a high sample rate might have had in the first place.) An industry White Paper that I saw at the NAMM show earlier this year compared *Peak*'s SRC to that of 11 other mainstream audio programs and concluded that *Peak*'s was amongst the best out there (the paper is available at www.bias-inc.com/products/peakPro5/resampling, for anyone who is interested in seeing the results in full). The SRC algorithm is also utilised to provide high-resolution tape-style audio-playback scrubbing via Apple's Core Audio. Where quick and dirty sample-rate conversion is required, however, *Peak* can be set to use faster, less sophisticated algorithms.

In addition to upgrading established algorithms, *Peak Pro* includes some completely new DSP processes, such as the aforementioned Auto-Define Regions, a Strip Silence function and the ability to derive an envelope from an audio signal. The already comprehensive Batch File Processor has now been updated to enable the preservation of file resolution and type, while a new Recover Audio File command gives users a fighting chance to recover part or all of damaged audio files.

Bits & Bobs

Some improved features take the form of additional convenience tools, such as the new Region Split command, and Unicode support that allows file names longer than 32 characters to be used. Audio files of up to 10GB can now be worked on without you first having to split them up, while waveform drawing has been updated, allowing more meaningful audio phase information to be shown in the audio

waveform window. SMPTE HD units (relating to frames of High Definition TV picture as opposed to PAL or NTSC) have been added to the time display, but one of the real biggies for those working in multimedia or game development must be the addition of the facility to Snap to CD Frames, PS2 or Xbox Units in the Actions menu.

While *Peak Pro 5* no longer needs *Jam* to take files to the CD pre-mastering stage, the days of included audio-DVD mastering are still some way off, so a special trial edition of Minnetonka Audio's *Discwelder Bronze* DVD-Audio disc-burning software is bundled with *Peak Pro*. The trial version enables users to burn up to five DVD-A discs, after which there's an opportunity to upgrade to the full version at a preferential price. However, I still find it frustrating that although surround sound has been with us for a long time, there's no straightforward integrated editing and Audio DVD-burning application that will let us take a bunch of surround WAV or AIFF files from a DAW and turn them into a playable surround album in one or more of the currently popular surround formats. Maybe if more effort went into this area across the board, surround would catch on more seriously in the project studio marketplace. As it is, it seems to be attracting very little interest.

Alternatives

There are no direct alternatives to *Peak* on the Mac platform, although i3's *DSP Quattro* (last reviewed in *Sound On Sound* April 2003) improves with every incarnation, so that may be an option if you need something less heavy-duty. There's also *Peak LE 5*, as described in the main body of the review, which currently costs even less than *DSP Quattro*. On the PC/Windows platform, Steinberg's *Wavelab* is the obvious 'big-gun' contender.

i3's *DSP Quattro*: worth a look if you don't need *Peak*'s full powers.

Using *Peak Pro 5*

As before, *Peak Pro 5* will only let you edit one stereo audio file at a time — it is not designed to handle multitrack or surround-sound projects — but it still offers vastly more functionality than the waveform editors built into the mainstream DAW packages — including *Deck*, BIAS' own multitrack DAW. While drag-and-drop is supported in the *Peak* Playlist window, you can't drag files from the desktop to the main edit window, or to the contents window, which feels a little odd, as most DAWs routinely allow you to do this. You can drag files to the application icon in the dock and they do open, however.

The batch processor is great for repetitive jobs, such as converting a folder full of audio files to a different format, or applying a sequence of processes to all of them. The RMS Normalisation process, designed to match average dynamic levels, is still a long way from being intelligent enough to optimally match the subjective ▶

levels of different styles of music on the same album (ballads would tend to sound too loud compared with heavier rock songs, for example), but it is infinitely better than using peak normalisation on everything and can be very useful when preparing material for the Internet or for Podcasts. The smoother tape-style scrubbing makes marking up edit points even easier than it was before.

As the manufacturers are very keen to point out, *Peak* is currently the only Mac waveform editor to support both VST and Audio Unit plug-ins within the same environment. Despite the fact that software samplers have virtually decimated the hardware sampler market, *Peak* continues to support the most popular models of hardware sampler, and now that there's also a MIDI input feature and soft-synth support (as I've already mentioned, both VST and AU), sounds can be played, recorded and processed entirely from within *Peak*. However, it might have been nice to include specific support for the more popular soft samplers, such as Emagic's *EXS24*.

As I use *Peak* only for special projects, usually mastering and file editing, I don't feel completely familiar with all of its operation and the toolbar icons still sometimes leave me scratching my head. The 'tool tip' info that comes up when you roll the mouse over an area helps here, but some days I still finding myself longing for the simplicity of the old *Sound Designer II* software, which I'd learned to use very quickly for editing and compiling. In *Sound Designer*, I particularly liked the way you could use a couple of keys to mark the starts and ends of selections on the fly without first having to create markers. However, in *Peak* you can place markers on the fly and then use a keystroke to 'change markers to regions'; although the process is one step longer it's still pretty straightforward.

Perhaps allowing the user to add colour to some of the toolbar icons would be a simple step forward, and adding *Logic*-style screensets might also make sense now that the program has grown a few additional windows. Another feature that might help is to have the currently assigned key command displayed alongside the 'tool tip' information, as you can configure so many things now that it's a job to remember them all. Of course, the beauty of *Peak*'s being so configurable is that if all you need is a *Sound Designer* alternative, you can can set up a limited set of toolbar icons and key commands that feel familiar.

One thing in *Peak* that seems a bit 'clunky' to me is how you have to manually zoom in to the playlist waveform overview to see your crossfades in detail: in *Sound Designer II* or *Jam*, you simply click on the required part of the playlist and the crossfade window opens. As far as I can see, you can also set up only one nudge value in *Peak*, whereas *SD II* allowed coarse and fine values to be set up. Maybe function-key combinations to increase the nudge time by a factor of 10 or 100 would be useful here? A final request, if technically possible, would be the ability to place a track-start marker in the middle of a piece of audio without first having to perform an edit — for example, between two tracks that segue into one another or during an applause section in a live album. To be fair, this is now easier than it was previously, as you can use the new Region Split command, but having it all in the Playlist window would be neater.

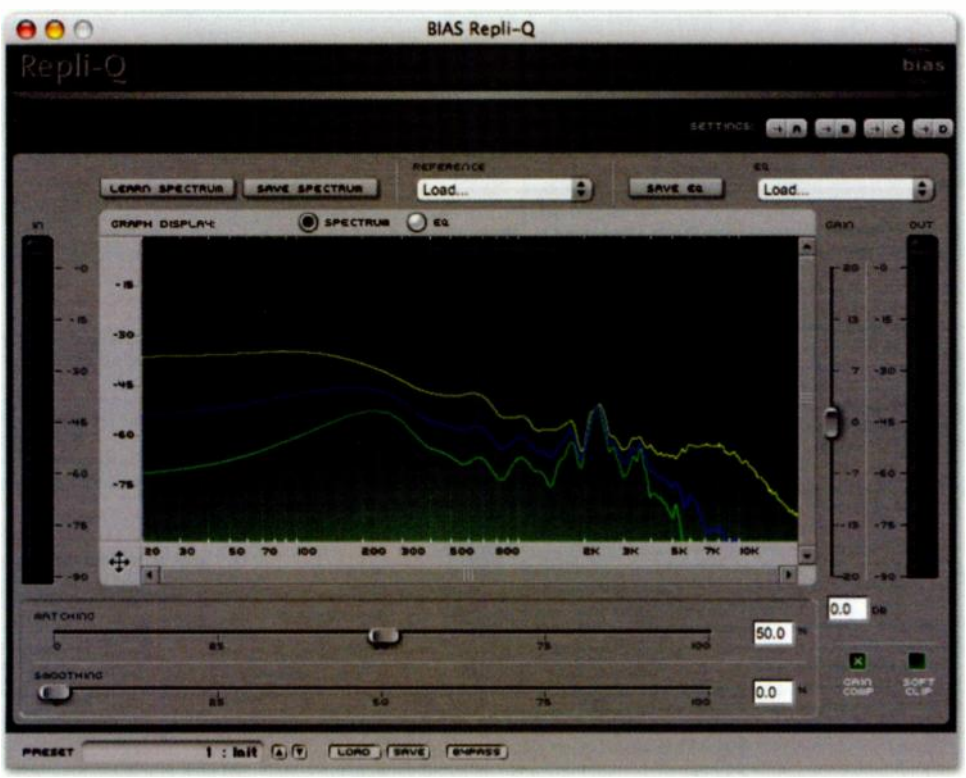
System Requirements

- G3, G4 or G5 desktop Apple Macintosh, iBook or Powerbook (400MHz processor or better recommended).
- Minimum 1024 x 768 screen resolution.
- Mac OS 10.3.9 minimum (CD Text requires 10.4 or greater).
- 256MB RAM (512MB recommended).
- 330MB available disk space.
- Hard drive offering 18ms average seek time or faster.
- QuickTime 6.0 or later.
- *Impulseverb* requires a G4 processor or faster.
- Support for third-party audio hardware may require compatible Core Audio drivers.
- Available USB Port for included USB Key.

The *XT* Option

In the *Peak 5 Pro XT* package you also get both the *Soundsoap 2* and *Soundsoap Pro* noise-reduction and audio-restoration plug-ins, which came out very well in our reviews when we checked out the the stand-alone versions. On top of that, as mentioned earlier, there's the *Master Perfection Suite* of plug-ins, offering mastering-quality multi-band (three and five bands) dynamics, the *Repli-Q* 'fingerprint' equaliser, new analysis and metering tools and a serious parametric equaliser. Unfortunately you can't yet open these plug-ins directly in other DAWs, such as *Logic*, but there are plans for an update that will allow this. BIAS have confirmed that they are going into the beta-testing stage on the *Master Perfection Suite* for AU, RTAS and VST on Mac OS X and Windows XP, with a projected release time of summer this year. It will be a free update for all *XT* users and will also be available as a separate bundled product.

Examining the individual components of the suite in turn, I'll mention *Repli-Q* first. This works in



Essentially a 'fingerprint' equaliser, *Repli-Q*, which is part of the *Peak XT*-only *Master Perfection Suite*, can impose the frequency spectrum of one sound onto another and offers numerous creative and corrective possibilities.

a similar way to other 'fingerprint' EQ products, in that it 'learns' a source frequency response and a target frequency response, then calculates a complex EQ curve to make the source match the target. The degree of detail in the EQ curve can be reduced using the smoothing slider, while the degree to which the source is processed can also be adjusted. Providing you're careful with this tool, you can make mixes of similar styles of music match in sound more closely, and also make DI'd acoustic guitars sound more natural by using a real miked acoustic guitar recording as the reference. You soon learn what this kind of processor can and can't do, but don't be afraid to experiment, as there's lots of potential. One example that came to my mind was creating your own speaker emulator by processing a DI'd guitar sound against a reference from a miked combo.

The *Superfreq* parametric EQ from the *Master Perfection Suite*.

The *Superfreq* parametric EQ is as easy to use as any other parametric, and you can load versions with different numbers of frequency bands, according to your requirements, up to a maximum of 10. In general, when adjusting EQ frequencies using the mouse, I initially found it difficult to arrive at precise settings, until I realised that you can use the mini wheel on the top of the newer Apple mouse to change the values in a far more relaxed way. I also discovered that as you drag the mouse further away from the selected knob the resolution increases, so the designers have clearly thought this through.

The *Sqweez* mastering dynamics plug-in can be thought of as a multi-band dynamic-cut equaliser or as a multi-band compressor, and it has a very informative user interface that displays the dynamically changing gains in the different frequency

bands. Personally, I find three bands enough to manage with for most routine mastering jobs, but it's always good to have the flexibility of more if you need it. I can't foresee ever needing more than the maximum of five bands.

Reveal is the only plug-in of the suite that doesn't actually apply processing, but it is immensely useful, as it offers an oscilloscope display of the audio waveform in stereo, a Peak and RMS power history, a spectrogram, a pan-power graph, real-time spectrum analysis (again, in stereo) and a phase meter. This collection of meters constitutes an extremely useful set of diagnostic and quality-control tools. For example, if one side of your mix is unaccountably dull compared to the other and you haven't noticed it, the spectrum-analyser display will show you right away.

By default, all the displays are visible at once, but there are tabs that allow any single display to fill the entire plug-in window. To the right of the window are high-resolution Peak and RMS level meters whose maximum range can be set to 48dB, 96dB or 144dB, so you can also get a pretty good idea of where your noise floor is during supposed pauses in the material.

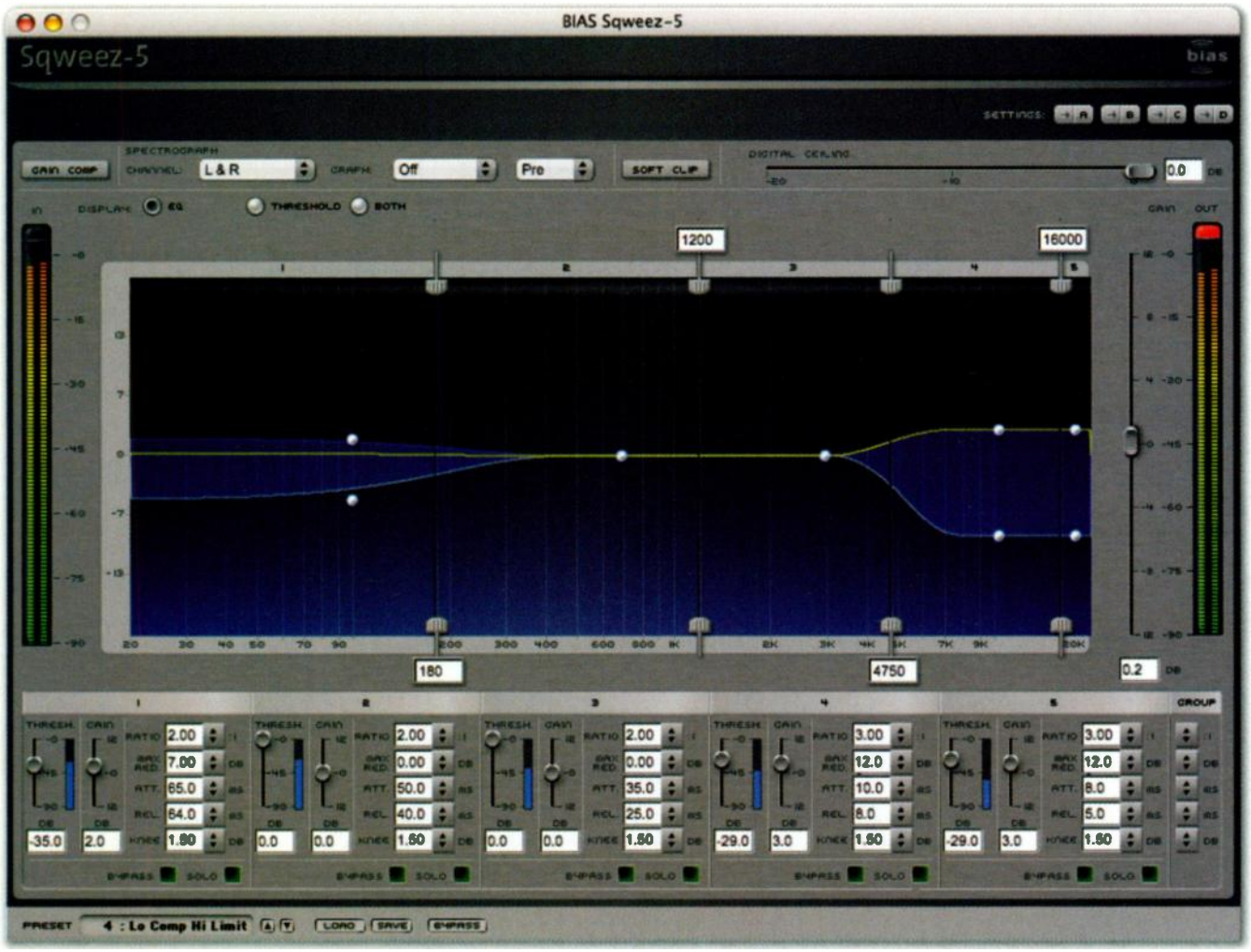
GateEx is a practical combination of gate and downward expander and can be used both as an advanced noise suppressor during pauses and to increase dynamic range. The graphical side of this plug-in has been very well thought out, with a display showing the signal waveform relative to the threshold settings and also a dynamics graph showing expander characteristics.

Pitchcraft covers some of the ground already trodden by programs such as *Auto-Tune*, but it is also capable of pitch transposition with formant correction. A graphic display at the top of the screen shows where and how much pitch correction is taking place, and there's that all-important slider for setting how quickly pitch changes are carried out. Users can set their own scales for correction and the familiar piano-keyboard note display is shown at the bottom of the plug-in window (see screen opposite).

We've covered *Soundsoap* and *Soundsoap Pro* before, but it's worth reiterating that these are very simple to use, yet effective audio clean-up tools for reducing artifacts such as hum and rumble, click and crackle, and broadband hiss. Usefully, there's also a built-in noise gate for more assertive silencing of pauses.

Peak Of Perfection?

Peak has always been an extraordinarily powerful program and this latest incarnation continues the trend. The underlying ethos of the software is very simple to grasp, so the main complication is remembering the keyboard shortcuts for the many functions that are now available. Creating custom toolbar sets is one way to help manage this. The only area of the program that I feel still needs work is the accessing of crossfade regions in the Playlist: I feel that *Jam*'s ability to let you click in the list itself and then open the relevant crossfade editor without any need for manual zooming is just a little more elegant. However, I've no doubt that the Playlist features will evolve in response to feedback from *Peak* users; after all, this is the very first version of the new Playlist window.



Sqweez, the multi-band compressor that may be used for mastering and is also part of the *XT* package.

The *Pitchcraft* pitch-correction plug-in that comes as part of the *Master Perfection Suite* with *Peak Pro 5 XT*.

In the previous version of *Peak*, I really liked the garishly bright *Peak* file icons, as I could always spot these in any Mac window from across the studio. The new icons are much more tasteful and subdued — which makes them harder to spot. A simple thing such as a preferences choice of icon syle would be appreciated by this user.

Being able to freely combine VST and AU plug-ins is certainly a big plus, and I know that a number of people will appreciate the virtual instrument support. *Peak Pro 5 XT* also comes with those serious mastering, audio-restoration and sound-design tools that could save you a fortune in third-party plug-ins. In most instances, these — combined with what *Peak* already provides — will be enough to get you through most mastering or restoration scenarios.

I know from experience that no piece of software will ever satisfy all the needs of all potential users, but the designers of *Peak* have tried very hard to cover as many bases as possible. Inevitably, this has resulted in a program that has much more functionality than most people will ever fully explore, but BIAS have managed it in such a way that the features you don't currently need should never get in the way of the ones you use all the time. *Peak Pro 5* is now a very mature and flexible piece of software, and aside from the minor suggestions I've made that relate to making navigation easier, I have very little to complain about. SOS

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that the A-D conversion is performed with a rock-solid local crystal clock, and the output then transcoded through sample-rate converters to provide the required output sample-rate, externally referenced if required. This approach ensures the highest possible conversion quality with minimal jitter, and the graphs and specs published in the handbook would seem to demonstrate the benefits of this clever technique.

The heart of the ADC1's analogue-to-digital conversion is a 192kHz/24-bit AKM5394 converter chip, which is running at a fixed 221.2kHz sample rate (referenced to a local 28.322MHz crystal). The AKM chip is apparently happy to operate at such an unusually high and non-standard sample rate, and Benchmark claim that this approach has significant benefits in terms of the signal-to-noise ratio and filter response in the subsequent sample-rate conversion.

Separate 22.5792 and 24.576MHz crystals provide clock references for the internal 44.1 and 48kHz output rates (and their multiples), and pass these directly to a Xilinx Spartan FPGA (Field Programmable Gate Array). This device is configured to provide all the operational control logic functions, internal signal routing, output formatting (AES, S/PDIF, ADAT and SMux), word-length reduction and even the metering display ballistics.

External clock inputs are routed to one of two receivers. S/PDIF or AES3-id reference clocks are passed to an AKM4114 chip, while standard word clock or Super Clock references are handled by an RS485 receiver. These particular devices were selected for their ability to recover usable clocks from very low-level and poor-quality signals. The recovered clock input is then passed to a VCXO PLL (a voltage-controlled, crystal oscillator-based, phase-locked loop, using the Analogue Devices ADF4106 and Texas Instruments PLL1706 devices), to regenerate a clean, low-jitter external clock reference, which is passed on to the FPGA.

Depending on the operating mode, the FPGA routes the appropriate internal or external clock signals to either or both of a pair of AD1896 sample-rate converters (the same type used in the DAC1). These accept the 221.2kHz sample rate data from the A-D converter, and calculate the output signals for the required sample rates for the main and Aux outputs. These signals are then formatted as necessary and routed on to the appropriate physical outputs by the FPGA.

In Use

Although the operation of the Mode switch is less than perfectly intuitive, the way it functions becomes apparent after a little fumbling around — and both the handbook and quick start guides explain it all extremely well, if you are the sort of person who reads manuals. The rest of the unit is very clear and simple to set up, and the range of acceptable input levels is unusually wide, making it easy to align the ADC1 with both professional and semi-pro equipment.

I particularly liked the option to switch between calibrated and user-adjustable levels at the flick of a switch, and although I was initially concerned that the detented level controls might make matching levels between channels difficult, this proved not to be the case at all. I was able to match channel gains very precisely at any level setting.

Everything worked exactly as advertised from the point of view of external clocking and output formats, and deliberately trying to upset the ADC1 with low-quality and extremely jittery external clocks had no noticeable effect whatever on the converted audio, which is as it should be. The ADC1 is quite lethargic when syncing to an external clock — it can take anything from about two to five seconds — but it gets there in the end and always reported incoming sample rates accurately during testing. The slow lock-up is an inherent side-effect of the jitter-filtering processes involved.

In terms of absolute sound quality and resolution, the ADC1 impressed me greatly — I evaluated the converter by partnering it with Benchmark's DAC1, and comparing it to my own Apogee PSX100 and the Prism Sound ADA8XR, which was reviewed in *SOS* April 2006 (www.soundonsound.com/sos/apr06/articles/prismada8xr.htm). At elevated sample rates the ADC1 boasts a clean, open sound with extended bandwidth in both directions. It has no discernible character of its own and seems able to capture whatever sonic qualities are present in the source without adding, subtracting or obscuring anything. The bottom end is particularly solid and defined, and I found that stereo images were captured accurately and with a very good sense of depth and spaciousness. My only mild criticism is that there is the faintest hint of congestion at the extreme top end when using standard sample rates, although all the double and quad rates sounded very transparent and neutral, with superb resolution and transient detail. In fairness, I should say that the Prism Sound unit showed a similarly subtle effect at base sample rates, and it was far more obvious on the (now obsolete) Apogee.

Conclusions

The ADC1 is expensive in comparison to typical mid-market converters such as those from RME, Lynx and Apogee (the Apogee Rosetta 200 is similarly priced but includes a stereo D-A stage as well, for example). However, I feel justified in suggesting that the ADC1 performs at a significantly higher level and compares more naturally with serious high-end products from the likes of Lavry, Prism Sound and dCS. In that context, the ADC1 represents substantial value for money, giving only a little away in terms of ultimate resolution.

The Benchmark DAC1 impressed me so much that I felt compelled to own it. I suspect my bank manager will fear the same will apply to the ADC1. The long wait has certainly been worthwhile, as the ADC1 lives up to the excellent reputation established with the DAC1 and the pair form a perfect partnership. The ADC1 is highly recommended for serious applications. SOS

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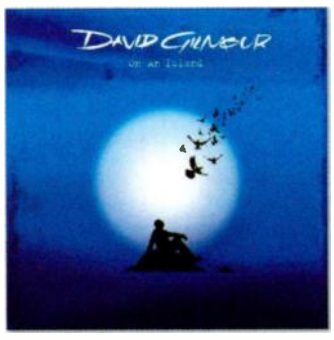
Water Music

Andy Jackson: Recording David Gilmour's *On An Island*

David Gilmour's chart-topping solo album was recorded on his own Astoria houseboat, a floating slice of studio heaven. Engineer Andy Jackson describes the making of the album.

Paul White

On An Island, David Gilmour's third solo album in around 30 years, was one of the most high-profile releases of the year, yet it was recorded and mixed mainly on David's Thames-based houseboat studio, the Astoria, which has little more space than some of the home studios I've visited. Admittedly it is kitted out with Pro Tools and a lovely Neve analogue mixing console, and it has an enviable microphone collection, but a lot of what ended up on the final album started out as David's self-engineered demos. These were made in his home, before subsequently being sifted and reworked with the help of his friend and neighbour Phil Manzanera, who Gilmour has known since before Phil joined Roxy Music. Of course any project of this size needs a safe pair of hands in the engineering seat, and that task fell to Andy Jackson, who's worked with David Gilmour and Pink Floyd for well over two decades. Indeed, this interview had to be postponed for a week or so when Andy was called upon to step in and mix a couple of Gilmour's live tour dates.

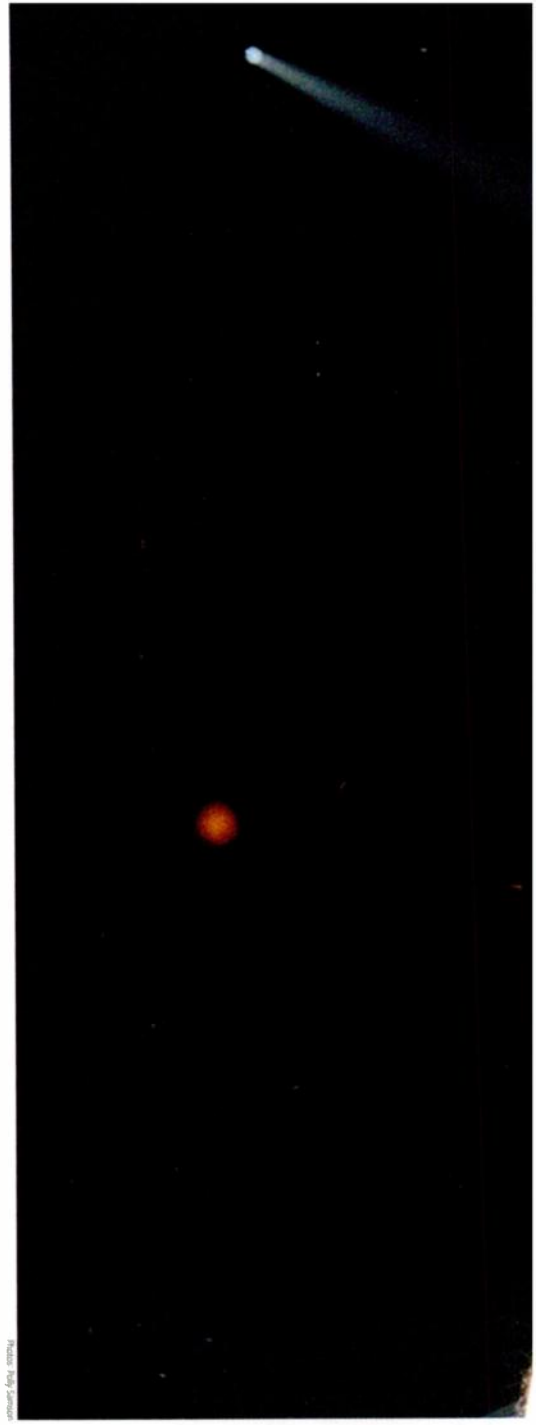


"I started in 1976 when I got a job at Utopia Studios, which was came about after writing lots and lots of letters. One of the engineers there was James Guthrie, and I kind of paired off with him — we tended to work together all the time with me as his second. When he left I moved up, and when the movie for *The Wall* album came along, he needed more bodies and asked if I could help with that, which I did. That film turned into the *Final Cut* album, and after that I never escaped — I did a solo album with Roger, a solo album with David and it just rolled on and on. James moved to the States so I became the engineering branch that did everything here. That was 1981, and here I am 25 years later still doing it!"

Bringing Things Together

A lot of the album's songs started life as bits and pieces David had been working on over the years, and he enlisted the help of Phil Manzanera in a production role to help him go through his ideas and see which ones fitted together. It was at this stage that Andy first heard the ideas. "It was all on hard drives as Pro Tools Sessions, as David has a Pro Tools setup at home, and as a musician/engineer, he's good. He also had a couple of really nice mics, Mastering Lab mic amps and EAR compressors, which we'd marked up with Chinagraph, so what we got in was pretty good. He'd started off with over 100 different ideas, where an idea could be as simple as just a few bars of chords or it could be an almost complete song. This was quite familiar to me as Floyd albums also tended to start life like that. He'd worked with Phil Manzanera to whittle it down and to see which bits might work together, moving it more towards finished songs.

"The indulgent part of the project was that we went into Abbey Road Studio One with a band and screened it off to one third of its size with a curtain. We used a six-piece rhythm section just to knock some of the ideas about really. Then we did another session in Studio Two, which was more of a rock band. The first session was more of a jazz band with Jools Holland on piano, Chris Stainton — Nick France on drums. It also included Bob Close, who had been with Floyd in the old days when they were still the Abdabs or whatever they were. He's a good jazz guitarist so we cut a few tracks like that. This was all recorded simultaneously to Pro Tools and analogue. We all liked the idea of doing everything analogue but I thought that it just wouldn't happen — Pro Tools is like being a kid in a sweet shop and you're just not going to not do that stuff! You move things, copy things, tighten things up and all the other things we can do, and what I didn't want was for us to record on analogue, copy it into Pro Tools for editing, then copy it back to analogue again! So I was fairly insistent that we recorded to both at the same time, which was a bit tricky, but we're fairly body-heavy and we had the people so that's



what we did.

"This was forming the backbone of the album up until the time we broke for the summer holidays and everybody took away a CD of work in progress to listen to. Then David came back and said he didn't like it — he wanted to go back to his demos so we started again. In fact on 'Take A Breath', the drum track that we recorded survived, and we managed to pull some of Jools's playing and some of Chris Stainton playing off those sessions, then lined them up with the demos so we could use it. At that point we were still trying to keep as much as possible on analogue but then David brought in Chris Thomas to oversee the workflow — to wield the cattle prod and get it done.

"One of the things Chris felt is that a lot of the tempos were too slow. It's a fairly laid-back album anyway, completely overtly so, and David's never been apologetic about that. He said 'That's where my head's at, that's what I want to do.' But Chris still felt some of the tracks were too slow so we ended up, after a lot of experimentation, time-compressing some things using Serato *Pitch 'n Time* in Pro Tools, which is generally regarded as being the best-sounding time-manipulation plug-in out there. The time-compression was a very laborious process, but fortunately somebody else did it — listening to each part, deciding which algorithm worked best for which type of material! This was all done on a second Pro Tools rig in another room.

"There were a few things Serato didn't do so well, but we got round it. For example, it really messed up the kick drum — it added some kind of backwards echo — so we just took some of the uncompressed ones and replaced them manually, lining them up in the waveform display. That didn't take as long as you might think — maybe half an hour to an hour per song. It also enabled us to find a good loud one, a good quiet one,

Abbey Road Studio One was used to track live band recordings of some of the songs on *On An Island*, but most of the resulting recordings were scrapped.


Andy Jackson in his mastering suite.

which suited Chris as he likes the kicks to be consistent and well behaved. It worked fine and it didn't end up sounding like a drum kit made of bits.

"It sounds like a lot of cutting and pasting but we did really try to keep some sense of performance. In fact some of the smaller songs, such as 'Walk Yourself Weary', are 90 percent David's demos. In some ways that gives it more of a relaxed feel than if he'd tried to replace some of the parts later.

"'Take A Breath' comprises two pieces that were originally very different — they were even at different tempos and I can't remember now whether we kept the tempo changes or used time-stretching to get them the same. But there's very little of the original that survived, whereas something like 'Smile' is almost entirely David's original demo. That's why I insisted we say the album was recorded by me and him because he did so much of the original recording. If it's good I get the credit and if it's bad he gets the blame!"

An Easy Person To Record

David Gilmour's voice and guitar are both very distinctive, but according to Andy, there's no mystery about how he records them. "Actually, it's very straightforward — voice — nice microphone, nice mic amp, nice compressor. There you go. For reverbs I tend to be old-school and use an EMT plate. I had a couple of plates and a Lexicon Hall — that was our palette. The vocal chain starts with that Sony tube mic with the heatsink on the side, the C800G, and it is the most fantastic microphone. We have a couple of those, one in the studio and one for David to use at home, which is another reason his demos sounded so good. That feeds an old Neumann V72 mic preamp and then EAR EQs and compressors like the ones I use here in my mastering system. I would compress his voice but only fairly gently with a tickle of 2:1, then maybe do that again on the mix. That's with the exception of the rock songs of course, which were completely mashed in a Fairchild! The thing is, David makes my life easy — stick him on the phone and he sounds great! He is not a difficult person to record — great technique and a great voice."

Grounding On Water

Having a recording studio on a boat must throw up a few unique technical issues, especially where grounding is concerned, and I wondered whether the Astoria was equipped with a copper anchor! "I've always thought we should just ground it straight into the river but no, we have gone to great lengths to optimise the electrics there, including fancy cable and the whole works. I do the same here in my mastering studio — it's all esoteric cable, exotic mains leads and a special braided grounding cable. No doubt I'll be shot down in flames about this, but when I changed the regular earth cable for the woven one it sounded better, even though the grounding was electrically fine before."

I was interested to hear this, as we've tried to do listening tests on cables in the past and we sometimes hear quite big differences, but the results are hard or impossible to replicate when you take the cables to a different studio. "Absolutely," agrees Jackson. "I've found exactly this when I go to James Guthrie's place in Northern California, being pretty adamant about how I want to do something because it works at home. Then we find it sounds totally different in his room so we have to come up with a different solution. That's really annoying because the subject of cables is such a minefield anyway! I've got to the point where I just call it voodoo and let somebody else worry about the physics behind it. I set myself up to be shot down in flames and then just ignore the people who are doing the shooting!

"What works works. For example, we have a couple of EQs racked up in the studio that came from a Decca broadcast console, which we call the 'better box', because even with no EQ applied, everything you pass through it sounds better. It may be because it has a couple of big, fat audio transformers in it adding a bit of second-harmonic distortion, and we like to hear second harmonic."

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RECORDING DAVID GILMOUR

During live Pink Floyd concerts, David Gilmour was renowned for using a lot of guitar effects and big amplifiers. I was curious to know what setup he used for recording. "Everyone asks what reverbs David uses, and the answer is none! He uses delays, and it's usually around 700 milliseconds or so. That's on his pedalboard, but apart from that there's probably only one or two different distortion boxes and maybe a compressor. It's not that complex — it's just finely tinkered with and he's got some nice guitars and good fingers. The amp was generally an old Fender Tweed Twin Reverb, with a little bit from his Hi-Watts occasionally. When he's recording at home, he just kind of sticks that Sony mic in a non-specific place in front of the speaker and I tried to replicate that in the studio, but it wasn't really working in our room. Chris wanted to stick an SM57 on it, maybe four inches from the grille cloth and a bit off-axis, but then I put a Coles 4040 ribbon mic next to it, dead in the middle of the cone, and we found that mixing that in behind the 57 really worked. What you hear is mainly the 57, but when you bring up the Coles, the sound just goes 'expensive'. Nearly all the guitar I recorded ended up being done like that — you just have to be really careful about the mic positions and make sure both are exactly the same distance from the speaker.

"Some of the guitars would be from David's home recording as he has a similar amp and effects setup at home. In fact on the guitar solo for 'On An Island' where there are two guitars, the first is a Les Paul and the second one a Strat. He recorded the Les Paul at home using the Sony mic and I recorded the Strat in the studio using the SM57 and the Coles ribbon, so if you want to hear how the two approaches compare, that's a good place to do it.

"With that combination of mics, the guitar had a real bite to it and it's very different to what we have done before. On *Division Bell*, we used a U87 eight inches to a foot away, a bit off-axis, and the floor of the room on Astoria is carpeted. It's actually very small — around 14 feet square — and sounds fairly dead, so when we record vocals, I just make sure the mic isn't right in the middle of the room as that gives a little 'boing'. All the drums were done there apart from a couple of the really live-sounding ones, as were all the *Division Bell* drums, and it's good for that typical Pink Floyd drum sound — even though it is [*Andy*] Newmark on that album, it still sounds like Pink Floyd."

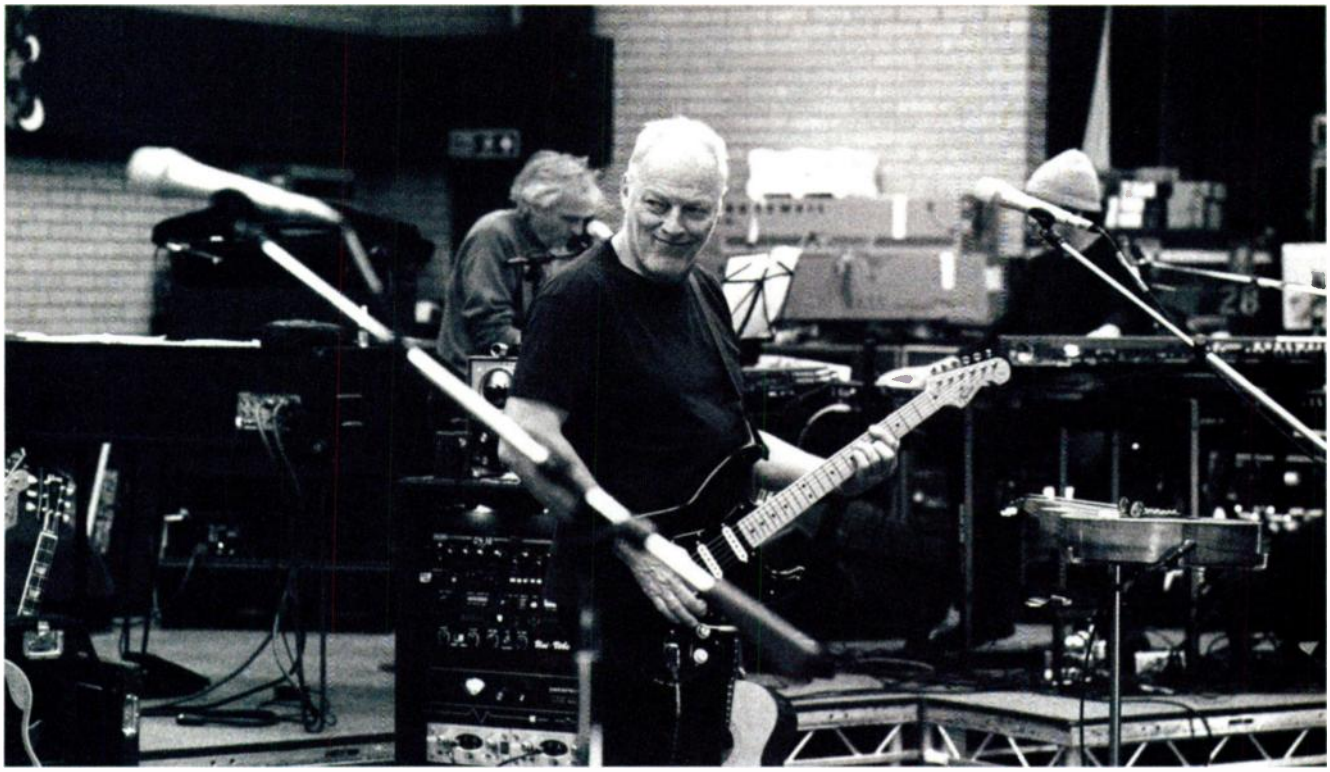
At one time David used a guitar fitted with EMG pickups to play live, but does he record using that too? "He has done, but mostly he's reverted to his old guitar. *Division Bell* was mostly his red Strat with the EMGs, but Phil Taylor, his guitar tech, thrust his old passive one into his hands and said 'Play that!' He plugged it in and it sounded great — it is a good one. I think he bought it new in 1971 or something, but he's swapped pickups and parts on it over the years and it does sound good. Of course, with those passive single-coil pickups, you have to face Mecca when recording to avoid hum problems! Usually we don't need to do any cleaning up afterwards, other than topping and tailing the various sections so they start and finish cleanly."

Some of the many famous guest musicians who contributed to the album: from top, David Crosby (on the deck of the Astoria), Jools Holland and Robert Wyatt.

The album also sees Gilmour playing lap steel, with what sounds like a ton of compression to make it sustain — but isn't. "Actually that guitar just sounds like that. It's a very old Gibson lap steel and it has an unbelievably hot pickup. He just plugs it into his rig and that's what comes out. I might occasionally compress some of the guitars on the album, but it was an 'icing on the cake' type of compression with just a couple of dBs here or there because it helped the focus. We have almost the opposite problem, because David plays so loud that when he stops playing everything squeals! When he's setting up, he turns up the levels so it's just about to go, so that he gets the sustain he needs without having to use lots of distortion. Certainly, on stage his amplifier is frighteningly loud, but it's remarkable — he's still got really good ears."

The Measured Approach

As you might expect, Andy is strongly of the opinion that making the right choices at the recording stage is the key to a successful mix. "It's got the point where I keep everything very simple, but then I suppose I've had 25 years of making the choices that have got me to that point. I know what's going to work in terms of what mics I put on things. I always think that if you simply take a microphone and record the world, it's a rather boxy and low middly place. You spend the rest of your life trying to get rid of that stuff. I tend to pull out a lot of low-mid, and with vocals it might be up to 500Hz. That's just automatic for me, then I'll add a little fairly dust. Ditto drums. The way



David Gilmour rehearses with his touring band, featuring Phil Manzanera and Pink Floyd's Rick Wright, for live dates in support of the album.

I approach overheads is that I take the old-school Glyn Johns approach and use the overheads as kit mics rather than just cymbal mics.

"One of my favourite tools for recording is a tape measure! It sounds really anal but it's worth it. If you get the two drum overheads exactly the same distance from the snare, it sits bang in the middle of them with no phase problems, and in fact on this project, a lot was recorded for surround as well so I used four overheads set up as a square, all equidistant from the snare. The overheads were exactly a metre and a half from the snare, and I used Coles 4038 ribbon mics, which of course have a figure-of-eight pattern so you get a bit of the room in it, but it sounded good. I had to take out some low mid, but then most of the drum mics get the low mid taken out somewhere or another. I try to pay a lot of attention to what goes on at the bottom end really — the relationship between the kick and the bass. I don't know if I can put my finger on any magic technique, it's somehow just a sum of everything — of experience.

"With bass guitars I often find myself pulling out stuff that seems unlikely at around 80 to 100 Hz, which is where all the action is. But you still get the whole sense of weight below it and it just cleans up the whole mix. Same with the kick — sometimes I'll be pulling out stuff in the high bass region and leaving the low stuff in.

"One nice thing about having a pet studio you can go and play in is that myself, and Damon who works there, can try things out without time pressures. There is a studio kit that's left set up there so one day we decided to try out absolutely everything we had to see what sounded the best on kick. I started off with all the things I'd normally choose but ended up liking none of them, which I found really interesting. Finally we tried a Neumann FET47 capacitor mic, and I've stuck with that ever since. It's not the instinctive first choice — most people think of D12s or D112s, but I really liked the result. I believe George Massenburg also likes to use a 47 FET on kicks, and it's fine. Once you stick the pad in it's absolutely happy. I also use a little Beyer M160 ribbon on the snare — there seems to be a phase coherence or something about a ribbon, and it's the same with the overheads. The Coles 4038s are not very bright so it seems odd to choose something that needs a lot of EQ, but once you add some top it sounds great — and somehow better than using a mic that has a naturally extended high end in the first place."

Mix & Mix Again

Despite his emphasis on traditional techniques, Andy is certainly not averse to using new technology where it helps. "Having recall has made such a difference to mixing. Even when we did [*Pink Floyd's*] *Division Bell* we didn't have a recallable mixing board, but there are so many times when you just need to go back and change something by a little bit that it's great to have it now. Of course all the home studio software has it as standard now, but working on an analogue board that doesn't have recall seems preposterous now. I'd be interested to try to do some serious mixing in the box, but I'd need to have first-class interfaces to do that. I'd also need a really ▶

Mastering

As well as manning the controls at the Astoria, Andy Jackson is also building a parallel career as a mastering engineer. "That was a strategic decision I made about four years ago. Aside from the commercial reason that mastering is more buoyant, it's also perhaps a reaction to the Floyd stuff where we'd spend a year on an album — they were enormous projects. With mastering, I really enjoy having a relationship with a record for just five or six hours and my experience in that area has been really positive. It's a time when people have finished the record and are happy!

"I have noticed that some of the stuff coming in now from home studios, where everything has been mixed in a DAW, has a different kind of 'not very well done' feel compared to what I used to get 20 years ago from analogue tape, and in some ways it is harder to deal with. There seems to be a particular kind of digital crunchiness. I don't know what the problem is, but if I play with digital limiters or other processes, they all seem to do it to some extent. I hope I'm not being a Luddite but I still have a degree of suspicion about things digital. It seems that the more you start processing a signal in the digital domain, the more it deteriorates, in a quite different way to analogue."

good control surface.

"David's album was done on Pro Tools, but we used it as a tape machine where everything came out of 48 outputs and got mixed on the Neve analogue board. There was some submixing inside Pro Tools, but then if we'd been working on two analogue 24-tracks locked up, we would have had to bounce some things down to stay within the track count anyway. In that case it means you're doing the summing digitally rather than analogue, which means you can undo the bounce if you have to, which we did on occasions. We had to do a little processing with plug-ins within these submixes, for example, when we recorded the Crosby and Nash vocal parts for 'On An Island'.

"I must admit that it quite surprised me that they recorded their parts one at a time when I'd set up assuming they'd sing together, perhaps with David adding a third part — a kind of Crosby Gils and Nash! That ended up coming up on the board as a stereo pair, but then I had to use a plug-in to EQ Crosby relative to Nash, just to tip him up a little brighter to match Nash. But the bulk of the EQ was done analogue afterwards."

The rest of the album was also recorded and mixed in fairly traditional fashion. "There are odd bits of Kurzweil and maybe the odd bit of synthetic Rhodes, but mainly it was old-school mic recordings plus using the EMT plate and things like that. It was mixed through the Neve analogue board and onto half-inch analogue tape. I think David looks back to some of the things we did in the '80s using AMS reverbs and things like that and says 'What were we thinking?' You can listen to the plate all this time later and it still doesn't sound old-fashioned.

"I did use a Lexicon 480 for the preset hall used on the orchestra and things like that — you'd think that when you record in Abbey Road Studio One the last thing you need is more reverb, but we did! On this album it felt like I needed to have a great deal of control so I could always get back to where we were, and the EMT plates plus the Lexicon were all we needed. But I'm a great fan of things like the TC System 6000 and we did use *Altiverb* on a few daft things, such as to get the effect of the fireworks in the opening section sounding as though they were outside [*using the 'Forest by Lake' impulse response*]. I think they originally came from a stock sound effects library. We had a flirtation with trying to record our own sound effects using Holophonics back in the '80s, but it's really hard to get anything that's any good." SOS

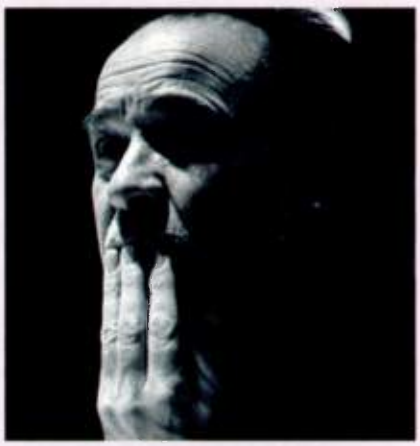
Phil Manzanera: Producing *On An Island*

Phil Manzanera and Dave Gilmour have known each other since the late 1960s, and their friendship progressed naturally into a production partnership. "I live close to David's house in Sussex," says Manzanera, "and I would regularly visit him and ask him whether he'd been recording. It turned out that he'd been doing stuff over a 10-year period, and had so much that he didn't know what to do with it. I suggested that I come round once a week and we'd go through these pieces together. Many were on Minidisc and recorded in his kitchen or living room. Dave began by selecting perhaps 150 pieces, enough for three albums. I picked some of these and took them to my studio. Together with my engineer, Jamie Johnson, I would try and develop these ideas. Sometimes we'd make loops, sometimes I'd take vocal lines from one song and stick them in another, just to create something different and weird. The next week I'd play David what I'd done and he'd give feedback.

"Dave and I spent most our time together at his home studio, where he has Pro Tools, a 12-channel mixer, a fantastic Sony microphone [*the C800G*], and a couple of good limiters, Urei I think. It's all very simple, but of high quality. Dave did all the engineering. It is a nice room with high ceilings, looking out over the countryside, and there are birds twittering around everywhere. The birds you hear all over the album were just picked up on the microphone in his home studio. What he recorded there has a feel that can't be recreated anywhere else. The songs 'Smile' and 'Where We Start' were almost entirely recorded by David at his home studio."

Work began in May 2004 and continued for six months at Gilmour's home studio and Manzanera's Gallery Studio, where he has a Euphonix desk and a Pro Tools setup, but works mainly in *Logic*. Even when sessions moved to the Astoria and Abbey Road, Manzanera still found time at the Gallery to concoct all manner of ambient atmospherics. "The rhythmic thing in the background on 'This Heaven' is a loop based on a sample from Dave's Minidisc," says Manzanera. "It was recorded in the kitchen with children shouting in the background and Polly [*Samson, Gilmour's wife*] talking, and so on. Once she and Dave had decided to call the album *On An Island* it really helped me with my conceptual thinking and with the making of little soundscapes.

"For the opening track, 'Castellorizon' [*the name of the Greek island to which the album title refers*], I took samples from all over the album. I put them all in the same key, and messed around with them. 'Then I Close My Eyes' was originally just a jam of David and [*pedal steel player*] BJ Cole, so I took this guitar riff from the Minidisc and made a backwards loop out of it. I then made an ambient soundscape from the soundtrack of a video shot at Castellorizon, with David in a boat, playing the cümbüs. So there are waves lapping and him playing and a ferry going past, and I slowed down a double bass playing harmonics and created this musical collage that was grafted onto the front of that track.



Phil Manzanera co-produced *On An Island*.

"I also made up an ambient track for 'Red Sky At Night', with children's voices and other sounds, while 'Take A Breath' has underwater sounds and a bell, plus the sound of something being thrown in the water. I may also have some dolphin and whale sounds. I've always enjoyed doing things like that, and with Pro Tools and *Logic* it's become so much easier to do. Basically, the album ended up with sound effects all over the place. We had set up an additional Pro Tools studio in a tea house right by the river, close to the Astoria, where Polly, David, an engineer and myself would go to mess around, experimenting with soundscapes and other things. In the end we had too many of them, so some were taken off during the mix."

Chris Thomas joined to complete the project in the last quarter of 2005. "Chris is a hero of mine," remarks Manzanera, "and we felt that we needed a final push for the last three months. David felt that we needed some fresh energy, and I loved the idea of having Chris around. It did put David in a position where he had to very much fight his corner, with me being a referee, saying 'This is all very well, but it's David's album, it's his choice.'

"Basically a lot of the early stuff was very acoustic; we almost entered into English folk territory. This was very interesting, but I like things a little heavier, and was perhaps trying to push a side of David that he wasn't feeling so much at the time. When Chris said that the tracks were too slow, he articulated something I had been feeling as well. But it needed the two of us to push the speeding up through! We sped up pretty much all the tracks, except for maybe one, by two or three bpm. We're not talking huge amounts here, but there's slow, and there's too slow, and I don't like too slow. We did it in Pro Tools because we already had a lot of stuff recorded that we didn't want to re-record at that stage. Obviously we were very concerned about sound quality and we A/B'ed stuff all the time, but there didn't appear to be any difference at all.

"The other aspects that changed at the last stage was that the album was initially not only very acoustic, but also very orchestral. I didn't think it was a brilliant idea to have so much orchestra, so it was diluted in the end. And finally, David's electric guitar solos were all done in November, which was the last month of recording. He left the electric stuff to the last moment, and I wasn't sure he wasn't going to do it at all. So I was very pleased that we managed to get him back onto the electric guitar, and he put some great things on." *Paul Tingen*

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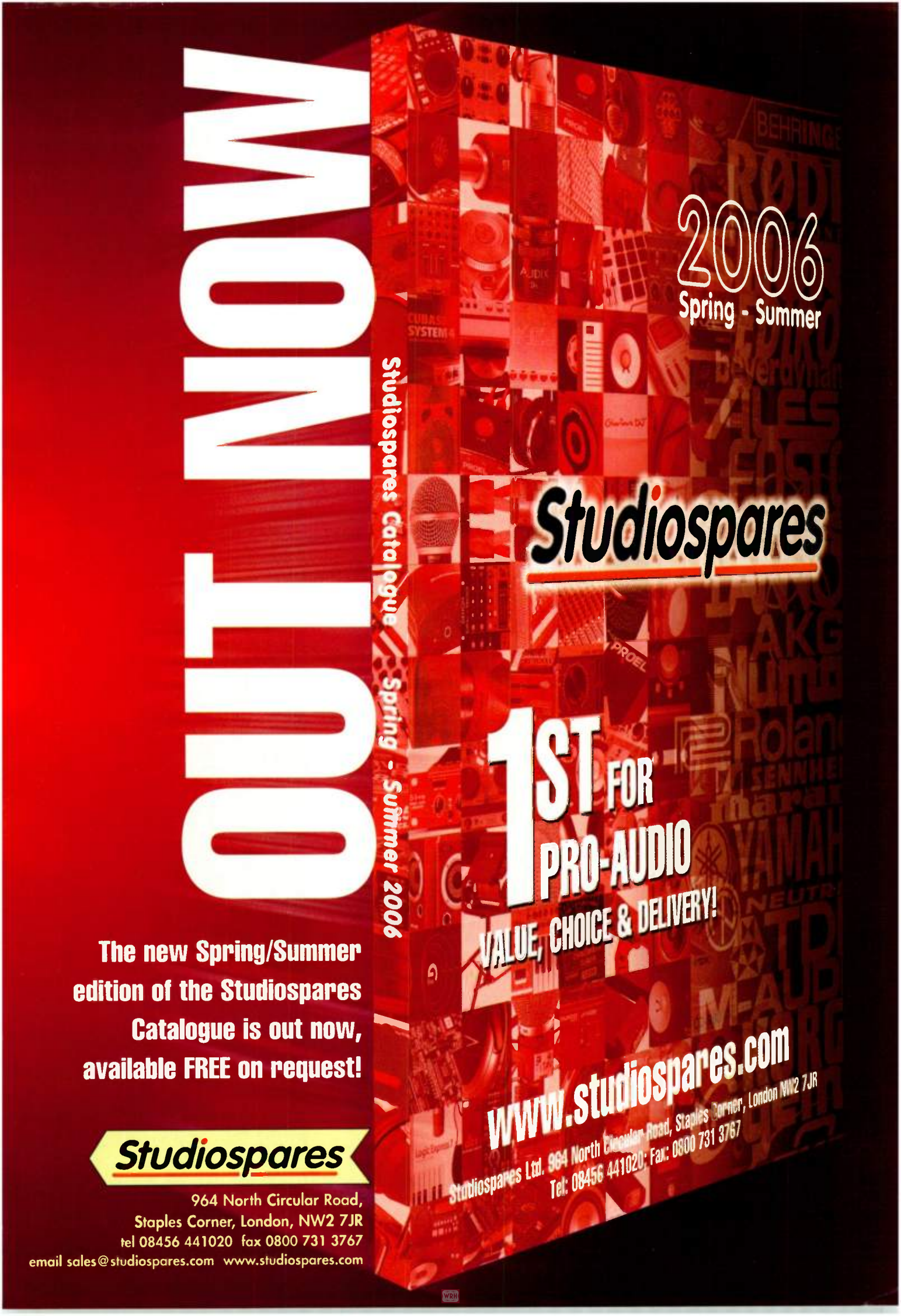
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Focusrite Saffire LE

Firewire Audio Interface For Mac OS X & Windows XP

Focusrite's Saffire already offered an awful lot of Firewire interface for your money, but the new LE version could just be even more of a bargain.

Martin Walker

Focusrite's Saffire audio interface (reviewed by Paul White in *SOS* September 2005: www.soundonsound.com/sos/sep05/articles/saffire.htm) has proved very popular since its release, and it's not hard to see why — as well as audio and MIDI I/O, the Saffire featured Focusrite mic preamps and a selection of DSP effects, which were also included in native versions to run as software plug-ins in your favourite sequencer. The Saffire is still a tempting buy at £349, but Focusrite have obviously spotted a gap slightly lower down in the market, which is where this £239 LE version comes in.

The biggest difference is that the Saffire LE has no onboard DSP to run effects, although you do still get the native plug-in versions. Elsewhere, the number of balanced/unbalanced analogue outputs drops from eight to six, although this still allows you to output in 5.1 surround, and there's only one front-panel headphone output rather than two. The LE only supports sample rates up to 96kHz, but this is hardly a big problem: the original's 192kHz capabilities will be of little practical use in most home studios anyway.

A few metering and monitoring functions are simplified on the LE; inputs one and two now only have a single Overload LED instead of three-LED level meters, while the monitor output loses its Dim and Mute buttons and there's no MIDI Out/Thru button (although all three of these functions are still available in the Control Panel software). The LE software bundle is also different from that of the original Saffire, with *Cubase LE* replaced by a special *Lite* edition of Ableton *Live 5*, compact versions of FXpansion's *BFD* and *Guru*, and 470MB of samples from Loopmasters.

However, it has the same digital audio and MIDI I/O as its bigger brother, and better still, the original two mic/instrument/line inputs featuring those desirable Focusrite mic preamps are now supplemented by two additional balanced/unbalanced line/instrument inputs on the rear panel, each with a dedicated front-panel Overload LED. The Saffire LE can be buss-powered from one of its two Firewire 400 ports, although a 12 Volt DC wall-wart PSU is included for anyone connecting it to a four-pin Firewire port on a PC laptop. All of the analogue I/O bar the XLR mic inputs is on quarter-inch TRS-wired jacks and can be used as balanced or unbalanced.

The front-panel analogue output level

SOUND ON SOUND

Focusrite Saffire LE £239

pros

- Excellent stereo imaging due to low-jitter clock.
- Good-sounding mic preamps.
- Front-panel analogue level control ideal when connecting direct to active speakers.
- More inputs than original Saffire model.

cons

- No GSIF drivers for *Gigastudio* owners.

summary

If you want the hardware DSP effects you'll have to buy the original Saffire, but if you lusted after its cute desktop design, good-sounding mic preamps and excellent audio quality then the Saffire LE might be the perfect small Firewire interface for you, especially as it's £110 cheaper!

control for outputs 1/2 is a welcome feature that I've been asking for from manufacturers for some time, as it greatly helps those who plug their audio interface directly into a pair of active monitors, so they can easily set control-room levels without compromising audio quality by turning down digital faders. Well done to Focusrite for that, although it would be even better if all front-panel rotary controls had calibration marks to enable repeatable settings to be achieved, both for control-room monitoring levels and to help match input gains when recording in stereo.

Installation

Drivers for the Saffire LE are provided for Windows XP and Mac OS X, and I had no problems installing them on my PC. However, one thing I didn't pick up from Paul White's original Saffire review is that the hardware incorporates a Pace Interlok dongle to protect the native plug-ins. In order to use them you need to send a generated challenge file to Focusrite's web site, along with the Saffire serial number and plug-in registration code, to receive a response file. You can do this up to four times, so you can use the Saffire LE and plug-ins on different computers.

The Saffire Control LE panel is rather different from that of the original Saffire, largely because there are no longer any DSP effects to be adjusted. Like all such utilities, it took a few minutes to get my head round it at first, but I found it flexible and easy to use after that. Controls for the four analogue and two digital inputs are at the top left, with individual level meters. 'Hi Gain' switches are available for line inputs 3/4 to make them more suitable for instrument duties, and there are six monitor faders and pan controls with stereo link buttons controlling the input signal mix to analogue outputs 1/2. Below this section are six more almost identical sets of monitor faders, pan and stereo link controls; these set up the mix of input signals that gets routed to the headphone output and analogue outputs 3/4. On the right-hand side are four stereo level meters displaying software playback channel levels, plus two rows of four stereo faders controlling the amount of each of these signals that gets routed to the main and headphone outputs.

Test Spec

- Focusrite Saffire LE driver version 1.0 build 5801.
- Intel Pentium 4C 2.8GHz processor with Hyperthreading, Asus P4P800 Deluxe motherboard with Intel 865PE chip set and 800MHz front side buss, with 1GB DDR400 RAM, running Windows XP with Service Pack 2.
- Tested with Steinberg *Cubase SX* 3.1 and *Wavelab* 6.0, Cakewalk *Sonar* 5.2, Tascam *Gigastudio Orchestra* 3.1, NI *Pro 53* 3.0, Rightmark *Audio Analyser* 5.5.

Brief Technical Specs

- **Sample rates: 44.1, 48, 88.2 and 96 kHz from internal clock.**
- **Mic/guitar/line inputs: two, balanced XLR with switchable global +48 Volt phantom power and +13 to +60 dB gain range, balanced/unbalanced TRS quarter-inch jack line instrument with +13 to +60 dB gain and 1MΩ impedance, or line with -10 to +36 dB gain.**
- **Line inputs: two, balanced/unbalanced line-level TRS jack at -10dBV or +16dBu sensitivity.**
- **Analogue outputs: six balanced/unbalanced TRS quarter-inch jack at +16dBu level, analogue level control for outputs 1/2, plus headphone output with analogue level control.**
- **Digital I/O: S/PDIF in and out up to 24-bit/96kHz on phono coaxial, AC3 and DTS compatibility, MIDI in and out, two Firewire ports.**
- **Dynamic range: 105dBA (analogue inputs and outputs).**
- **RMS jitter: <250 picoseconds.**
- **Frequency response: 20Hz to 20kHz ±0.1dB.**
- **THD + Noise: 0.001 percent measured at 1kHz.**

A central vertical strip of controls offers master level controls for the three analogue output pairs, which — usefully — can be linked for surround work, plus associated Mute buttons. There are pre-fader meters for the main and headphone outputs along with sliders that balance input against playback levels for main and headphone outputs, plus a handy Dim button to drop main output levels by 18dB (not 12dB as stated in the manual), and a further button that optionally sends the main output signals to the S/PDIF outputs.

The Saffire Control LE utility can also operate in two preset modes. The default Track mode is ideal during recording, setting up both monitor and headphone mixes from all the inputs as well as playback channels, while S/Card mode mutes all input signals from the monitor and headphone mixes, and sets playback controls so that only Playback 1/2 emerges from the main out and Playback 3/4 from outputs 3/4 and the headphones.

Sound Quality

It's good to see Focusrite bucking the trend by quoting the Saffire LE's internal clock jitter level, and at less than 250 picoseconds, their figure is considerably better than that published for any other interface below £500 that I've reviewed to date. A low-jitter clock is vital for excellent stereo imaging, so I was very keen to see whether this figure translated into an obviously audible improvement.

After matching the output levels of my own Echo Mia and Emu 1820M, and the review Saffire LE, to within 0.1dB I started to

Although the Saffire LE loses a pair of analogue outputs compared with its big brother, it actually offers an extra pair of line inputs.

▶ listen to them in turn, but it only took one audio snippet to dispense with the Echo Mia — it was so easily spotted as being slightly harsh and distinctly less focused. Switching to the Saffire LE was like lifting a veil from the sound, letting me hear the instruments beneath. It took me slightly longer to make up my mind between the Emu and Focusrite interfaces as the differences were smaller, but while the Emu 1820M, as always, turned in a refined performance, the Saffire LE took the lead with even tighter and more focused imaging, making the Emu sound a little 'woolly' by comparison. Emu's range has done incredibly well to retain my personal vote for 'best audio quality for audio interfaces under £500' for the last two years, but I think it's finally met its match in the Focusrite Saffire series: the Saffire LE provides truly excellent subjective audio performance for a £239 interface. I was also impressed with its mic preamps, while the 1MΩ impedance of the instrument inputs was high enough to avoid any premature roll-off of high frequencies.

To double-check what my ears were telling me I ran my usual raft of Rightmark *Audio Analyser* tests. Frequency response was -0.3dB down at 18Hz and 20.5kHz with a 44.1kHz sample rate, and at a much wider 20Hz to 42kHz at 96kHz sample rate, with a tiny 0.3dB peak at about 32kHz. Measured dynamic range confirmed the manufacturer's figure of 105dBA, and although you can buy 'quieter' interfaces, I personally feel that the Saffire LE's lower-jitter clock is more important to overall audio quality.

Driver performance was also good, with *Cubase SX* managing the lowest 4ms latency at 44.1kHz on my PC without any glitching, as it did in *Sonar* 5.2 *Producer Edition* (with the WDM/KS drivers it managed the lowest 10ms Effective latency with ease). NI's *Pro 53* managed a slightly better than average 30ms Play Ahead setting with the Direct Sound drivers, and 45ms with the MME ones. My only disappointment was that the Saffire LE doesn't have GSIF drivers for *Gigastudio*.

Paul White covered the Saffire plug-in effect suite (reverb, compressor, EQ and amp simulator) in great detail in his original review, and since the LE versions are identical I won't elaborate here, except to say that I found that the three rather basic *Saffire Reverb* controls produced a surprisingly versatile set of sounds after a little experimentation, and the Template/Advanced modes of the *Saffire Comp* and *Saffire EQ* are an inspired way of providing you with initial settings to suit a particular application, while still letting you get at the individual controls for further tweaking.

The Jewel In The Crown?

It's always a gamble introducing a new cheaper model to an existing range, since the manufacturer risks losing sales of the more expensive version. So have Focusrite thrown out the baby with the bathwater by dispensing with the hardware DSP effects of the original Saffire, or have they got their sums right? I think they have judged the Saffire LE very well indeed — built-in DSP effects can be handy, but many musicians will be perfectly happy to save money and rely on plug-in or outboard effects.

The Saffire LE provides two excellent mic preamps, instrument inputs that work well for DI'ing guitars, two extra line inputs compared with the more expensive Saffire, enough outputs to suit stereo or surround work, and excellent audio quality throughout. The Firewire interface market has never been so crowded, but with the Saffire LE Focusrite have carved themselves yet another niche — at £239 this is an absolute bargain! SOS

With comprehensive monitor mixing and metering of input and output levels, the Saffire LE's new Control utility is effective across a variety of recording and playback scenarios.

Alternatives

If you want a compact Firewire-based interface between £200 and £300 with a versatile selection of input options including mic and guitar, enough outputs to cope with surround, and with good audio quality, there are quite a few on offer. The Presonus Firepod has a very similar spec to that of the Saffire LE, though its headphone output benefits from being a separate feed and not tied to the main 1/2 output, and it can now be bought for about £250, making it the Saffire LE's closest competitor. Edirol's FA66 has a similar input complement, but only four analogue outs, and has optical S/PDIF instead of coaxial (which may be more compatible with your other digital gear). However, it does have an attractive analogue limiter to keep recording levels in check, and is only £200. If you want more analogue I/O and don't mind losing the limiter, Edirol's FA101 provides a total of eight analogue inputs and outputs, plus optical S/PDIF and MIDI I/O for around £290. M-Audio's Firewire 410 only has two versatile inputs, but eight analogue outputs plus two headphone outputs with individual level controls, the extra versatility of both optical and coaxial S/PDIF, plus MIDI, and can now be found on the street for £240 — the same price as the Saffire LE. However, unlike the others mentioned here, its analogue I/O is all unbalanced.

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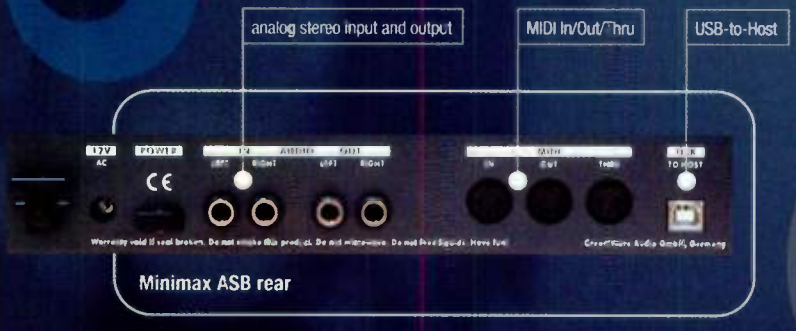

Minimax ASB rear

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pros

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- Narrow vertical directivity helps to reject unwanted sounds.
- Broad uncoloured horizontal directivity similar to a small-diaphragm mic.
- LED to signify when powered.

cons

- Prone to popping.

summary

An unusual cardioid microphone that has significantly different polar responses in the vertical and horizontal planes thanks to its unusual rectangular diaphragm. The sound quality is also more reminiscent of a ribbon than of a capacitor mic.

Hugh Robjohns

Pearl Mikrofonlaboratorium started in 1941, originally producing crystal and dynamic microphones. However, the Swedish company's intriguing rectangular condenser capsule design was introduced at the end of the 1950s and has enjoyed continuous development ever since.

The Pros & Cons Of A Rectangular Capsule

The traditional circular capsule membrane is easy to produce consistently, but its dimensional symmetry means that there is a single, very prominent resonance which has to be tamed using various engineering techniques. This often impacts on the frequency response and sound quality — particularly in the time-domain response. Pearl's rectangular capsule has different dimensions in each axis, which results in multiple resonances of reduced strength, which are therefore easier to control. Pearl claim that this produces a capsule with a more accurate and uniform frequency response, and that the improved time-domain response makes the mic sound more natural. The company's new ELM-C mic, under review here, retains the rectangular capsule concept, but this is an all-new cardioid capsule design with an enormous 7:1 length-to-width ratio.

The rectangular diaphragm also manages to combine the best aspects of both large- and small-diaphragm round capsules at the same time. For a start, the diaphragm's surface area equates to that of a circular diaphragm of about 30mm — roughly the same as in a typical large-diaphragm condenser mic. This large surface area translates into high sensitivity and an improved signal-to-noise ratio. The ELM-C produces an output of 22mV/Pa with a self-noise figure of 9dBA. To put that in context, the Neumann TLM103 and Microtech Gefell M930 share the top of the league tables with self-noise figures of 7dBA.

However, because the rectangular diaphragm is relatively narrow, the accuracy of the horizontal polar response is significantly more consistent with frequency than that of a large circular diaphragm — just like a small-diaphragm round capsule. Of course, the very long aspect of the rectangular capsule means that the vertical directivity is inherently pretty narrow at high frequencies, but that can be used to advantage when positioning the mic to minimise the pickup of floor and ceiling reflections, or when avoiding spill from unwanted sources. Of course, you also have to take a more thoughtful approach to placing the mic, because small changes of vertical angle can have dramatic effects on the captured sound.

Hardware Overview

The Pearl ELM-C is an unusual looking mic. It's body is rather long at 192mm, with a base section of 28mm diameter, increasing to 32mm for the grille. It weighs a modest 305g and is only available with a matt 'black chromium' finish. The grille is sufficiently transparent to reveal the spectacular side-address capsule in all its glory — to the uninitiated it may appear to be a ribbon mic rather than a condenser mic! One rather handy feature is that a small red LED is tucked at the base of the rearward side of the capsule, confirming at a glance that the mic is receiving phantom power. The mic's phantom power current demands are moderate at 2.7mA, and the specifications suggest the maximum SPL figure is 126dB for 0.5 percent distortion. The mic has no built-in high-pass filter options or pads, but doesn't seem to need them either.

The mic is shipped in a tough, foam-lined aluminium case,

Alternatives

The ELM-C is an expensive mic in anyone's terms, and there is some extremely strong competition at this price range. However, none of these mics have the unusual but potentially very useful quality of different polar patterns in the vertical and horizontal planes. Broadly comparable (on price) microphones from my own collection would be my favourite Sennheiser MKH40 small-diaphragm cardioid microphone, or (pushing the budget slightly) the Neumann TLM127 large-diaphragm mic. Alternatively, the AKG C414B XLS is a highly regarded and competent mic, and with the advantage of five switchable polar patterns.

▶ with an individual serial-numbered quality-control chart. Unfortunately, the plot spans a rather unhelpful 100dB dynamic range, but the trace still suggests the frequency response to be flat within about 2dB from 40Hz to around 12kHz, with a smooth and gentle roll-off above that, reaching roughly -5dB at 20kHz.

The review model was supplied with two optional shockmounts. The Type 1927 looks like a child's toy tractor tyre trapped in a ring with a standmount adaptor on it! The alternative Type 1928 is a variation on the more traditional 'cat's cradle' idea, but with typically Swedish styling, using a square central frame instead of a round one. The stand adaptor is mounted in one corner, and upper and lower rubber loops stretch from each corner to support an inner tube structure that holds the mic. The upper collar of the tube has a thumb screw to hold the mic securely in place, and the whole thing works well.

Studio Tests

The first thing that strikes you when positioning the ELM-C is just how big the capsule is, and that automatically forces you to think carefully about how to orientate the mic and where to point it. Since the vertical polar pattern is so narrow at the high end, small changes of angle can alter the sound quality and level of off-axis sounds enormously — the vertical pickup angle is remarkably narrow and well defined. However, rotations have comparably little effect, because the horizontal polar pattern is very consistent, with a pretty broad frontal pickup area and a healthy lack of off-axis coloration.

As an example, if you place the mic vertically in front of a vocalist, tipping the mic up or down slightly really changes the character of the voice, as the chest, head, lips, and nose come more on or off axis. A large-diaphragm condenser mic behaves in a similar way, of course, as you tilt it back or forward, but the effect is extremely mild compared to the reaction you get with this Pearl mic. Mounting the mic horizontally instead produces a far more uniform or consistent vocal sound, but high-frequency reflections from side walls, or spill from other instruments in the studio are usefully attenuated. The extremely open single-layer wire-mesh grille means that the ELM-C is rather prone to popping, so a decent pop-shield is absolutely essential with this mic.

The marked difference in vertical and horizontal polar responses caught me out on a couple of occasions. The classic example was in miking a clarinet (although I'd expect the same problem with a sax too) — the sound was just badly balanced and a poor representation of the fullness of the actual instrument. What I had done, without thinking, was to place the mic vertically in front of the clarinet where I would normally position a mic to 'see' the whole instrument. Of course, the vertical polar pattern is so narrow that it was actually only seeing the mid-portion of the neck — hence the naff sound! Re-mounting the mic sideways (which looked rather odd and unnatural somehow) produced the much more balanced sound that I had been expecting in the first place.

"This mic's unique directional characteristics allow a far greater level of control and selectivity than almost anything else — approaching the source discrimination of a figure-of-eight mic in the vertical plane, while retaining the broad smooth pickup of a cardioid horizontally."

Verdict

Once you have mastered positioning this mic, its sound quality is impressive. There is something of a ribbon mic's smoothness and naturalness about the sound — it doesn't have the slightly dull top or emphasised mid-range that is characteristic of many large-diaphragm mics. In fact the ELM-C seems slightly brighter than a typical large-diaphragm mic, even though the diaphragm surface area is roughly the same, and the top end seems a little more detailed, yet the sound is also smoother and less 'hyped' or artificial. In many ways it sits tonally in the space between large- and small-diaphragm condenser mics. Not quite as bright, crisp, and clinical as a typical small diaphragm, yet brighter and smoother than a typical large diaphragm while retaining much of the same sense of scale.

After using the mic for a few days I realised I was thinking about the mic as a kind of high-output ribbon mic with a cardioid pattern. The ELM-C seemed to excel with the same kinds of sources that ribbons always shine on — percussion, brass, voices, and especially pianos all sounded superb. I didn't get a chance to try a string section during the review period, but I would expect the mic to perform extremely well if presented with this challenge.

The ELM-C is not the easiest mic to position, or the most forgiving if placed poorly. Its tech specs, while respectable, aren't the most spectacular either, yet I would gladly add a couple of ELM-Cs to my mic box given the opportunity and budget. The mic's unique directional characteristics allow a far greater level of control and selectivity than almost anything else — approaching the source discrimination of a figure-of-eight mic in the vertical plane, while retaining the broad smooth pickup of a cardioid horizontally. The only other mic I have used with a similar disparity in directivity between the two dimensions was the huge multi-capsule Microtech Gefell M970, and that proved very useful in a wide range of applications from miking choirs and orchestras to covering speeches from lecterns. The only thing that counted against the M970 was its enormous size, and the Pearl ELM-C is much more manageable in that regard. The ability to reject unwanted sources or acoustic anomalies in one plane, while capturing wanted sounds in the other is remarkably useful and even addictive!

This mic is relatively expensive in the UK, and that puts it in direct competition with some far more familiar and respected high-end large-diaphragm condenser mics. However, the ELM-C offers some intriguing and surprisingly useful features that are unmatched by its more traditional rivals, while matching the sonic quality expected at this price level, and that's what makes it important to audition this mic. Impressive. SOS

information

£ ELM-C, £953.48; Type 1927 shockmount, £43.63; Type 1928 shockmount, £179.90. Prices include VAT.
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Photos: Mark Ewing

Korg Pad Kontrol

USB MIDI Drum Pad Controller

Pad controllers are aimed at releasing the frustrated drummer in all of us, allowing desktop composers access to rhythm sounds without having to use sticks — and Korg's Pad Kontrol adds a couple of nice twists to the concept...

Nicholas Rowland

While the growth of computer-based music-making has seen an almost bewildering proliferation of keyboard-based MIDI control devices, there are still relatively few aimed at the specialist task of triggering drum sounds and programming rhythm tracks. In fact, even with the arrival of Korg's Pad Kontrol you don't actually require all the fingers of one hand to count them! Like the other main contenders in this area, namely M-Audio's Trigger Finger and Akai's MPD16, the Pad Kontrol is designed to be enable you to trigger sounds using your fingertips (as opposed to controllers like Roland's SPD20, which are intended to be played with sticks).

Physical Matters

A quick shufti reveals the Pad Kontrol to be equipped with 16 trigger pads, two assignable knobs and buttons, and an X-Y controller, plus enough intelligence to store up to 16 sets of user-programmable MIDI assignments or Scenes (man) as they are known in lingua korga. While it is primarily designed to interface with computer-borne software instruments through its USB connection, the Pad Kontrol will also communicate directly via MIDI with any other hardware devices. And while its main role is supposed to be for real-time programming of drum sounds, its designers have given it many features that mean it can be employed for much more than simple one-shot sample-triggering duties. For example, it can be used to control the transport functions of a sequencer, or as a way of remotely switching between different programs and patches on another device.

Korg have consistently produced some of the best-looking hi-tech instruments around, and the Pad Kontrol is no exception. With its smart white and silver colour scheme — surely the colours that will be associated with the early years of the 21st century, thanks to the all-conquering iPod — this is an

appealing and very sleek-looking package. It easily passed my wife's 'Cor! What's that you've got now?' test (to which all review items in our house are subjected as soon as they are liberated from their packaging). On closer inspection, I have to say that the unit does feel a little bit plasticky, especially the rather lightweight data knobs, though there's no reason to assume it won't bear up under normal use. In that respect, normal use is obviously intended to be on a tabletop or other flat surface, as there's no way to attach a clamp for standmounting.

The first job is to use the supplied cable to attach the Pad Kontrol to an available USB port on your computer. For Windows XP users this procedure also requires the installation of the supplied driver from the CD-ROM. While the Pad Kontrol would normally be powered over its USB connection, there's also an input for an optional 9VDC adaptor if this proves a problem in any way — for example, if there are too many devices on the buss already. Naturally, you'll also need the adaptor if you are using the Pad Kontrol as a stand-alone controller to trigger other MIDI devices, or if you are bypassing the USB connection and attaching the Pad Kontrol to a computer using its five-pin MIDI In and/or MIDI Out sockets. Incidentally, the presence of these sockets means the Pad Kontrol can function as a USB MIDI interface — a useful extension to its role if, like me, you never seem to have enough free MIDI ports.

Ergonomically, the Pad Kontrol follows what seems to have become the established convention for this type of controller with its 16 pads arranged in a four-by-four matrix. The pads are velocity sensitive and offer a range of eight different dynamic curves to suit different 'finger-drumming' styles. For more predictable results, each pad can also be set to send specific fixed velocities. As well as being used to trigger sounds, the pads multi-task as program and parameter selection buttons when you're programming the Pad Kontrol itself. Oh, and because they are backlit with red LEDs they can also provide part of the entertainment too. I'm not just talking about the way they light up every time you hit them; it's the fact that they can be set to blink in various different patterns when a pad is struck. You can also set them up so they will start to blink randomly whenever you leave the unit idle for a minute or two. It's rather cool in a kind of '1970s sci-fi movie computer set' kind of way, and it certainly impressed visitors to my studio. But the pads don't just look good; they also have a very firm comfortable feel, and a great response which is even over their entire surface, including the very edges. They're definitely a cut above the usual squidginess you get with a lot of pads on drum machines.

Along with its trigger pads, the Pad Kontrol also offers a couple of programmable buttons and a brace of controller knobs — all very useful for controlling parameters on the target device. There's also an input for a footpedal which can be used either for control and switching duties, or as an extra trigger, in a rough approximation to a bass-drum pedal. It's worth emphasising that the programmability of the pads and other controllers goes far beyond what you'd expect for a triggering device of this type. Along with being able to assign individual note numbers, velocity values and MIDI channels to each pad, you can also assign control change numbers, change the switch type between momentary and toggle, and set release values. Put this together and it means you can use the device to control operations on software synths and computer sequencers.

The other main difference between the Pad Kontrol and its rivals is the presence of that X-Y controller pad (see overleaf). This type of controller is one of Korg's particular specialities — we've seen it incorporated to particular good effect on their Kaoss range of effects units. Resembling a control surface version of a joystick, the pad gives you finger-tip control of any two continuous MIDI controllers in real time, one assigned along the horizontal axis and one along the vertical — so as you move your fingertip around the pad, you get a proportional mix of two continuous controller values. A typical use with a synth patch might be to control filter cutoff and resonance, but by programming ▶

SOUND ON SOUND

Korg Pad Kontrol £139

pros

- Looks great, feels great.
- Features the X-Y controller from the Kaoss Pad.
- Excellent flam and roll functions.
- Programming abilities go beyond simple triggering.

cons

- Nothing significant!

summary

Neat, creative, versatile and all-round kool, the Pad Kontrol raises the game for the fingertip drummer.

Round the back, MIDI can be transmitted via a pair of traditional five-pin sockets (Out and In) or via USB. The pedal jack allows the connection of a 'kick-drum pedal' (ie. a MIDI footswitch which can be *used* as a kick-drum trigger), and the switch allows you to select between DC power and buss power from the USB connection. Note again that you have to pay for the optional DC power supply; it's not supplied as standard.

Brief Specification

- User controls: 16 velocity-sensitive, backlit trigger pads, one X-Y pad, two assignable knobs, two switches.
- User memory: 16 user scene memories.
- Dimensions: 55 x 314 x 234mm (HWD).
- Weight: 0.96kg.
- System requirements: Windows XP, Mac OS 10.2 or later.

two continuous controllers that aren't normally related — like reverb depth and pitch bend — you can achieve some more interesting effects.

In their considerable wisdom, Korg have given the X-Y pad a couple of dedicated 'drummy' features in that you can use it to control the timings of flams and rolls. Depending on how you set it up, moving your finger from left to right increases the speed of a drum roll, or the distance between 'flammed' notes, while moving it up and down changes either the volume of the roll, or the second note in the flam. It doesn't seem such a big deal when you try and describe it in words, but when you start to play around with the effect, you realise just how brilliant it can be. For example, this makes it easy to create realistic-sounding drum and percussion rolls with very precise control of swells in volume and the 'microtiming' of the roll. Basically, if you want to recreate those dramatic drum rolls that you always get as the hero is led to the gallows, then the Pad Kontrol provides you with the perfect execution (ouch!). This feature of the Pad Kontrol also enables you to easily add those little grace notes that real drummers put in as they play — thereby making it a doddle to create very human-sounding performances.

In Use

Editing the Pad Kontrol is pretty simple and straightforward thanks mainly to the generous helping of function buttons, and despite the limitations of the three-digit LED display which does its best to inform you what's going on. This shows what message or value is being sent whenever you strike a pad or move a controller. As I said earlier, you can store up to 16 User Scenes in the Pad Kontrol's internal memory. In addition, the CD-ROM that accompanies the Pad Kontrol comes with 30 preset Scenes designed for use with popular programs. Among this list you'll find setups for various GM kits, drums from Korg's own OASYS and Triton synths, and kits from software such as Native Instruments' *Battery*, Ableton's *Live*, Apple's *Garage Band*, Propellerheads' *Reason*, Steinberg's *Groove Agent* and FXpansion's *BFD*. One of the Scenes is also designed to work out of the box with the accompanying special version of Toontrack's *DFH* (for more on this, see the box above).

However, while you can do all the editing via the buttons on the front panel, this seems a bit of a long-winded way to go about things given that the Pad Kontrol's accompanying software includes a rather excellent — and very free of charge — editor and librarian program. This provides you with a neat visual representation of the Pad Kontrol's surface, allowing you to click on the various pads and assigning values through the corresponding pop-up windows. Once a Scene is assembled, another click of the mouse squirts it down the USB connection into the Pad Kontrol's memory. Or conversely, if you've done your programming directly via the Pad Kontrol, a click of a button sends the information back up the wire to be stored in the computer.

Verdict

With the Pad Kontrol, Korg have done some out of the beatbox thinking and come up with a device with lots of creative potential. Naturally, your excitement about this product (or lack of it) will depend on how important real-time drum programming is to your musical *modus operandi*. If that's your bag, then I can state uncategorically that the Pad Kontrol is the best of the bunch — although admittedly, it's not a very big bunch! At the same time, though, Korg have given the Pad Kontrol a range of features which means it is going to be useful for general control of other MIDI devices too.

I do have one moan, and it's the same as I had about M-Audio's Trigger Finger — you can't program more than one MIDI note per pad, thereby allowing you to layer and crossfade sounds. But the flam and roll functions help to make up for this omission. And on a visual note, the Pad Kontrol looks pretty cool too, especially with all those pretty lights. I wouldn't quite go as far as Korg's web site, which claims that the 'great feel of the trigger pads and the Illumination mode work together synergistically to catapult your performance to a higher intensity level than ever before', but if this sentence is the result of some copywriter's enthusiasm for the product, then that enthusiasm, at least, is entirely justified! SOS

DFH — Drums Fine & Handy?

Although it has no sounds built in, the Pad Kontrol doesn't come to the party empty-handed. Along with the editor/librarian program on the accompanying CD-ROM, you'll find a special 'Korgified' version of Toontracks' well-respected *DFH* virtual drums software. It was my first encounter with this particular package, and I was mightily impressed. Toontracks' intent is apparently to give you the sound of drums as they are really played, and they seem to have achieved this, offering all the nuances and countless variations of tone that you get even from a single drum. You can also play around with the virtual miking of your kit, even determining the extent to which the sound of a drum will bleed into the mics on the rest of the kit!

While the version of the program available with the Pad Kontrol gives you only a fraction of what you get with the full version, it still adds up to a generous 260MB of drum multisamples, which in turn translates into a complete kit of bass, snare, toms and cymbals. Not only is this enough to give you a sense of just how good the program is, it also spectacularly showcases the Pad Kontrol's flam and roll functions. And of course, if you like what you hear, an included coupon gives you a discount off the full version.

Haven't we seen this somewhere before...? The excellent X-Y pad from the Kaoss Pad may seem an odd addition to the Pad Kontrol, but it can be put to good use for controlling Pad Kontrol flams and rolls, as well as any other MIDI controller pairs you wish to assign.

information

£ Pad Control £139; PSU £12.50. All prices include VAT.
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WRH

Toontrack EZ Drummer

Virtual Drum Instrument For Mac & PC

Toontrack's simplified version of their flagship *DFH Superior* offers fewer kits, but with the same sound quality and in a more user-friendly interface.

Paul White

Toontrack were amongst the first companies to provide really good multisampled acoustic drum-kit sample libraries (specifically *Drum Kit From Hell*), with both dry and ambient sounds that could be mixed to give the required degree of liveness.

Test Spec

- Dual 2.7GHz Apple Mac G5 with 4GB RAM running Mac OS 10.4.6
- Toontrack *EZ Drummer* v1.01

Since then they came up with *DFH Superior*, an excellent-sounding drum-sample-based virtual instrument that was probably less successful than it deserved to be, because its user interface was perceived as being rather complicated. In the light of competition from the likes of FXpansion's *BFD* and the rather more abstract *Stylus RMX* from Spectrasonics, Toontrack set about developing a simpler, easier drum kit virtual instrument, and the result of their efforts is *EZ Drummer*. *EZ Drummer* requires a minimum spec of an Intel Pentium III or AMD Athlon 1.8GHz PC with 512MB RAM running Windows XP, or an Apple Mac 1GHz G4 running Mac OS 10.2.8 or higher with 512MB RAM. AU and VST plug-in formats are supported, and authorisation is via the familiar challenge-and-response system.

Streamlined User Interface

While the user interface is extremely simple, the samples are of the same quality and range as those in *DFH Superior*, but with a narrower

SOUND ON SOUND

Toontrack *EZ Drummer* £99

pros

- Cost effective.
- Excellent drum sounds.
- Very user friendly.
- Expandable.

cons

- Comes with a relatively limited range of drum sounds.

summary

This simple and effective plug-in instrument provides a small but well-targeted range of studio-quality drum sounds along with a very usable library of MIDI drum rhythms and fills. It is easy to use, inexpensive, and can be expanded.

Included Drums

KICK DRUMS:
- 14 x 22-inch and 18 x 22-inch GMS with felt-and-plastic beater
- 14 x 22-inch GMS double head

SNARE DRUMS:
- Rogers Wood 4.5 x 14-inch
- Slingerland '70s 6.5 x 14-inch
- GMS Picollo 13-inch

TOMS:
- 12-inch, 14-inch, 16-inch, and 18-inch GMS

HI-HATS:
- 14-inch Zildjian HHX Manhattan
- 16-inch Zildjian Crash Hats

CRASH CYMBALS:
- 19-inch Sabian AA Medium Thin
- 16-inch, 17-inch, and 18-inch Sabian HHX Evolution
- 16-inch Sabian HHX Evolution Ozone
- 13-inch and 18-inch Sabian Jack DeJonette Encore

RIDE CYMBALS:
- 21-inch Sabian Handhammered Vintage
- 22-inch Sabian Handhammered Raw Dry

YAMAHA CLUB JORDAN COCKTAIL DRUM KIT:
- kick drum
- snare drum
- 5 x 8-inch and 5 x 10-inch toms
- 14-inch Zildjian hi-hats
- Mikaelsson Custom ride cymbal

choice of drum sounds. All the drums for this instrument were specially recorded at Avatar Studios in New York. Toontrack's own TPC II data-compression system is used to fit over 7000 sound files at 16-bit, 44.1kHz (equivalent to 5GB of uncompressed WAV files) into only 1.5GB of hard drive space, and they've also included a huge library (more than 8000 files) of very usable MIDI drum grooves and fills that can be dragged and dropped directly into a sequencer's arrange page in much the same way as is possible with Spectrasonics' *Stylus RMX*.

A Humanize button adds slight variations to give the patterns more of a 'played' feel, and the velocity response of the whole kit can also be controlled by a single knob to adjust the playing feel. Grooves and fills can be auditioned direct from the MIDI-file browser, and the arrangement is very logical, making it simple to pick out variations on the type of rhythm you want, as well as suitably styled fills. Another nice touch is that you can store your own rhythm MIDI files in the library, so if you've bought some of the excellent Twiddly Bits MIDI rhythms (or something similar), you can now access and audition them much more easily.

The drum kit mapping is preset to the GM standard, with some duplication on otherwise unused keys to help with two-handed playing, and there's a choice of using the plug-in with a stereo output or up to seven separate stereo outputs. Separate and main stereo outputs can also be used together, so you could, for example, keep most of the kit as a stereo mix, but isolate only the kick and snare for further processing. As with *BFD*, there's now a much more visual mixer that allows for the level adjustment and panning of the close mics on the drums, plus separate control over the overhead mic level and the stereo room-mic level. Mixer presets are available for fast changes in kit sound, and bleed between mics can also be switched on or off for the snare and overhead channels. However the ability to tune or damp the drums has been removed for the sake of simplicity.

The rather busy interface of *DFH Superior* has been replaced by a visual representation of a three-tom, three-cymbal drum kit viewed from the player's perspective. Clicking on the heads auditions the sounds, while a drop-down menu activated at the lower edge of the drum or cymbal rim shows the alternative sounds that can be loaded if you don't want to use the default kit. An info bar shows how big the samples are, so you always know how much RAM you've used. (See the 'Included Drums' box for a list of all the various kit options.)

The architecture of *EZ Drummer* allows for expansion kits to be added at a later date, and the Cocktail Kit that came with *DFH Superior* is included as standard. This has a wonderful light jazzy feel that contrasts nicely with the main pop and rock kits. When an expander is loaded, the drum kit in the display changes to reflect the contents of the expander. Further expanders are planned for around September, and the original *DFH* will also be turned into an expander. According to Toontrack, if there seems to be a demand, specialist percussion expanders may be added later.

The new visual mixer window makes it simple to balance the kit sound.

Is It A Hit?

Although the range of drum sounds is a little limited, what you get comes pretty close to the kind of drum sound I tend to strive for anyway, and by switching the snare drum and/or the kick drum and adjusting the amount of room and overhead mics in the mix, you can get anything from a general-purpose miked acoustic kit suitable for most pop styles to a fairly brash rock kit. The quality of the samples is excellent, with no obvious truncation and plenty of velocity levels, and this extends to the Cocktail Kit, which should fit in well with any contemporary jazz or Latin-influenced pop compositions. The sounds also work well when triggered from an external electronic drum interface such as Roland's V-Drums.

As to the user interface, that is now as simple and intuitive as you could hope for, and the only thing I think might usefully have been added is the ability to add swing to the rhythm patterns before dragging them into your host sequencer. Having said that, most sequencers will allow you to add swing to your MIDI parts retrospectively, so it's not an insurmountable issue. I also think it is worthy of special note that the vast majority of the MIDI drum rhythms are sensible variations on the 'bread and butter' drum patterns that most of us use most of the time, so you shouldn't have trouble getting them to fit your songs unless you're into writing in weird time signatures or using odd rhythms. And if you are, then there are always third-party MIDI drum loops that you can use with EZ Drummer. Given its low cost, great sounds, and ease of use, I can't see how *EZ Drummer* can fail to do well. SOS

Alternatives

There are no obvious direct alternatives for *EZ Drummer*, as most other drum virtual instruments are either philosophically different or more complex. FXpansion *BFD* probably comes the closest in terms of approach, but also check out Glaresoft *iDrum*, Submersible Music *Drumcore*, and Steinberg *Groove Agent 2* if you like the idea of a drum instrument that also includes rhythms.

information

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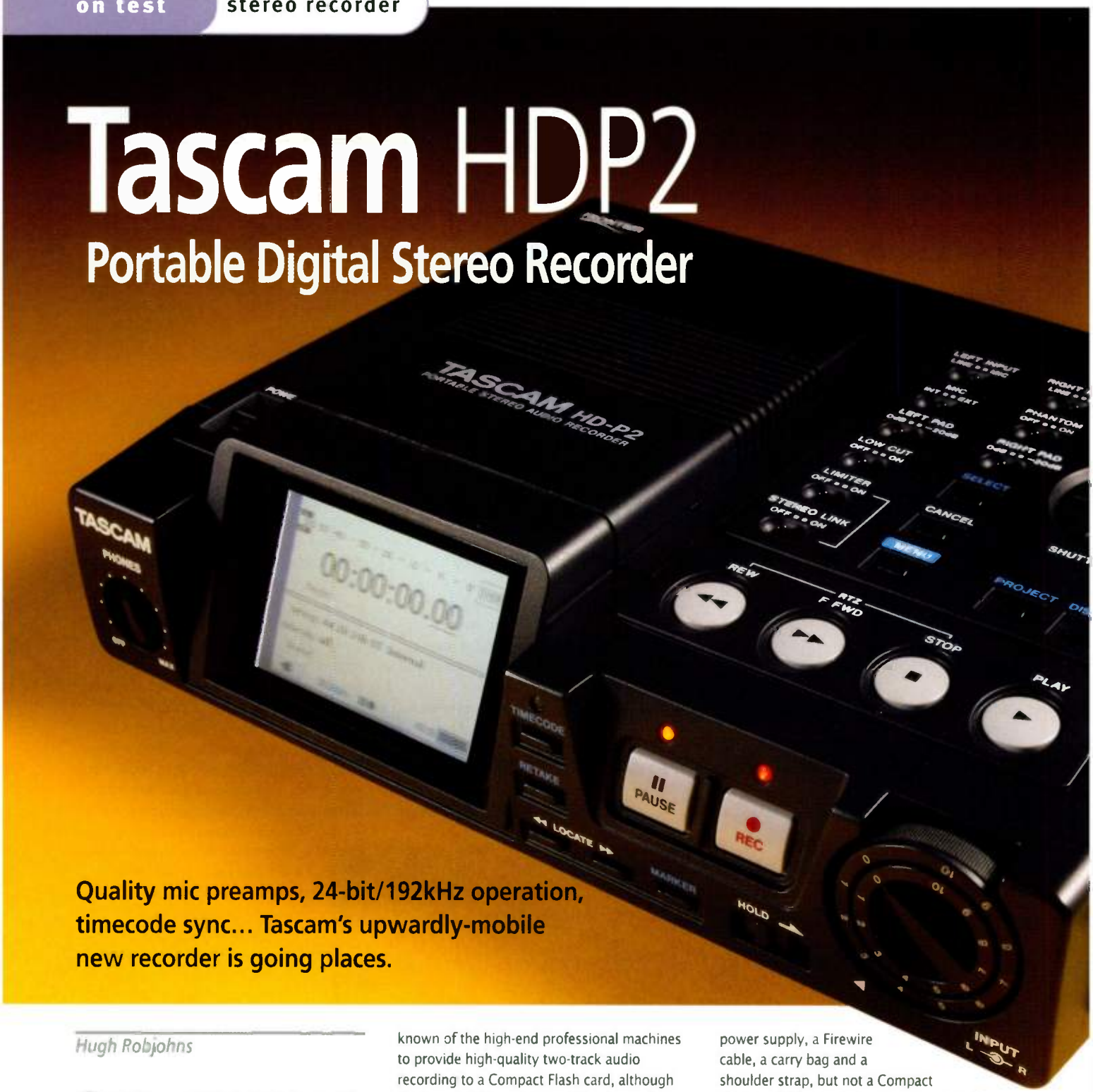


Quality mic preamps, 24-bit/192kHz operation, timecode sync... Tascam's upwardly-mobile new recorder is going places.

Hugh Robjohns

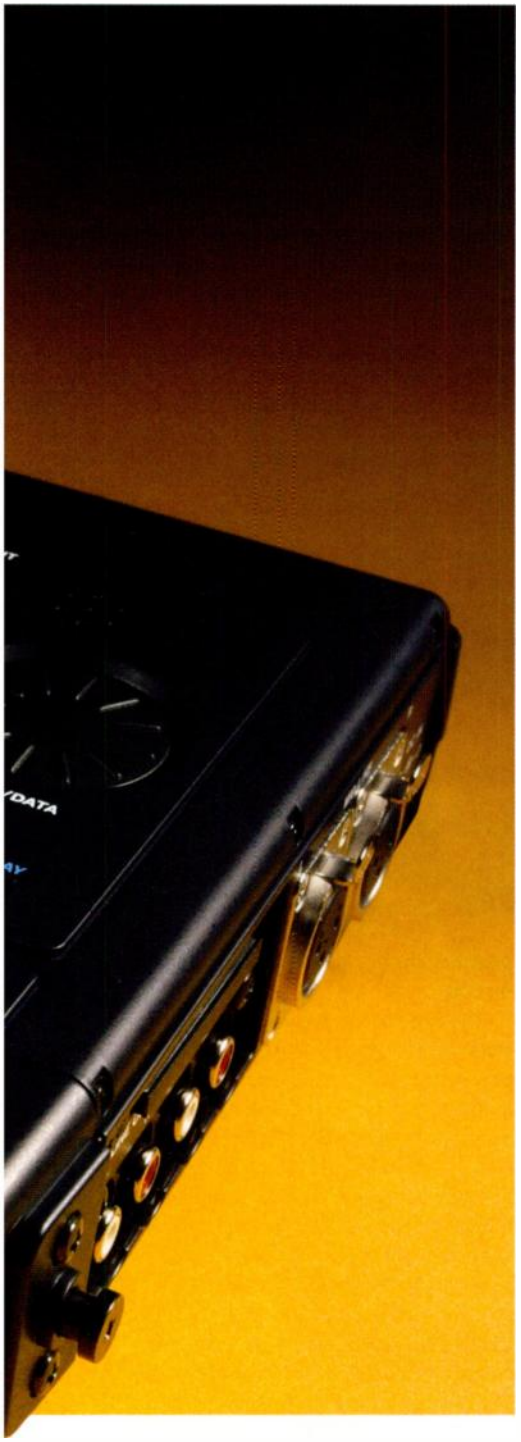
Over the years, manufacturers have used a variety of different media to make portable recording possible, from magnetic tape reels and cassettes to DAT tapes and Minidiscs. These formats are all but obsolete these days, and hard disk drives and solid-state memory cards are quickly becoming the 'standard' formats for all manner of data acquisition, whether in digital cameras, PDAs or specialist audio recorders. Compact Flash media now seems to be the most widely adopted format, as it offers very workable capacities at reasonable prices, together with ease of use and exceptional ruggedness and reliability.

The Fostex FR2 is amongst the better known of the high-end professional machines to provide high-quality two-track audio recording to a Compact Flash card, although there were a few more specialised recorders aimed at the broadcast journalist market before it, and several more cost-effective models have emerged since. The subject of this review is a new offering from Tascam (in conjunction with Frontier Design) which is designed to compete directly with the Fostex machine and to meet the needs of much the same market — the high-end enthusiast and the budget-conscious professional.

Overview

The HDP2 is a neat, compact machine, weighing under 2kg including batteries (eight 'AA' cells), and measuring 245 x 188 x 60 mm (WHD). The unit comes with an in-line mains power supply, a Firewire cable, a carry bag and a shoulder strap, but not a Compact Flash card. This omission is understandable, but could be frustrating if you get your shiny new machine home only to find you can't use it because you don't have any fast Compact Flash cards. I suggest you barter with your dealer to include at least one suitable card with your purchase. You'll want to make sure the HDP2's carry bag is included too (pictured on page 188), since this will help protect the machine if you plan to use it outside (and why else would you buy a portable machine?).

Battery life is estimated at up to five hours with Alkaline 'AA' batteries, and about three hours with rechargeable NiCad or NiMH batteries, but the actual time depends on how often the LCD backlight is used, how loud the



headphones are, and whether phantom power is on, amongst other factors. I achieved well over four hours of running time with Duracell Ultra Power batteries when using phantom-powered mics, so the manufacturer's estimated times appear reasonable.

The machine is laid out much like most other similarly-sized portable recorders. The top panel contains various input and menu configuration buttons and controls, while the front carries a large backlit LCD display, the main transport controls and the record level knob. The left-hand side contains various ancillary connectors and the right-hand side has the analogue audio inputs and outputs, plus the Compact Flash card slot. Batteries are loaded in a tray underneath the machine. Everything seems familiar and fairly intuitive on first sight, and most people will be able to get up and running at a basic level without needing to refer to the well-written handbook, which is relatively modest in size at only 26 pages.

The HDP2 records linear PCM audio files using the Broadcast Wave (BWF) file format. No data-reduced formats (MP3, for example) are supported at all. Audio can be recorded in mono or stereo, the word length can be switched between 16 and 24 bits, and all the standard sample rates between 44.1 and 192 kHz are supported. The HDP2 also has a pre-record buffer, with up to 10 seconds of memory at base sample rates. External clocking options include the S/PDIF input, video syncs (NTSC, PAL and HD tri-level syncs), word clock and Longitudinal Timecode (LTC) — the latter accepting 23.976, 24, 25, 29.97 and 30 frames per second, the last two rates in both drop- and non-drop frame versions.

The HDP2's audio specifications are all quite good, although not exceptional — total harmonic distortion is below 0.01 percent at 1kHz, for example. The converter delay between analogue input and output is a fairly typical 1.5ms at 44.1kHz sample rates (lower for higher sample rates), and crosstalk from mic input to line output is below -80dB.

Inputs & Outputs

The two microphone inputs are on electronically balanced XLRs, with globally switchable 48V phantom power. Each channel has an independent -20dB pad switch to accommodate high-level signals, and there is about 46dB of gain available from the dual-concentric level control. In practice, signal levels ranging between about -52 and +14dBu can be accommodated, which should satisfy most demands. A switchable peak limiter is also available. A second switch allows it to operate on each channel independently or in a stereo-linked mode, and there is also a global high-pass filter (18dB-per-octave below 100Hz).

In addition to the microphone input XLRs, the right-hand side panel also carries two pairs of phono connectors for unbalanced line-level analogue inputs and outputs. The inputs accept signals between -46 and 0dBu. The pad switches cannot be used to enable higher signal levels here as they only affect the balanced mic inputs, but normal line-level signals — from a CD player or hi-fi, for example — are not a problem.

The external power supply also plugs into the right-hand panel using a non-latching coaxial connector. Any 12V DC supply could be used, including an external battery pack, providing it is capable of supplying 6W of power with a 0.6A maximum current limit. The final facility on this side panel is the Compact Flash card slot and its integral eject button. There is no slot cover to prevent dirt or moisture getting in, but perhaps Tascam assume users will always keep a card in the slot.

The left-hand panel carries a quarter-inch TRS headphone socket (a recessed volume control is on the front panel), another pair of phono connectors for S/PDIF digital audio in and out, a BNC socket for word clock and video sync signals, another XLR for a timecode input, a six-pin 400Mbps Firewire socket, and a PS2 keyboard socket. The latter allows a standard keyboard to be attached for easier file naming as well as surprisingly comprehensive remote control functions — not just of the transport commands, but also to access and configure the menus and other machine attributes. The headphone output provides up to 55mW per channel into 32Ω headphones, which is sufficient for most applications, though it may not be quite enough when working in very noisy locations. If headphones are not plugged in, audio can be auditioned via an internal speaker mounted on the top panel behind the LCD screen. A single internal electret microphone is also built into the top panel for convenient low-quality mono recording purposes.

Controls

The top panel is neatly laid out with clear white and blue legending. Nine recessed toggle switches provide separate input source selection for the right and left channels (balanced mic or unbalanced line) and individual pad on/off switching, with global switches for phantom power, internal or external mic selection, low cut filter, limiter and limiter stereo linking. Five recessed push buttons on the top panel are used to navigate and configure the machine's menu structure, ▶

SOUND ON SOUND

Tascam HDP2 £899

pros

- Decent mic preamps and high-quality converters.
- Timecode record/chase included as standard.
- Compact and lightweight.
- Ergonomic and intuitive controls.

cons

- Not supplied with a Compact Flash card.
- Very limited internal editing facilities.

summary

A well-thought-out, high-quality portable recorder that is extremely capable and competent. Decent mic preamps coupled to 24-bit/192kHz converters and a range of sync options ensure high-quality recordings. Operation is largely intuitive, and it is hard to find significant fault with this machine.

labelled Select, Cancel, Menu, Project and Display. The Display button provides direct access to a menu page for adjusting the contrast and brightness of the LCD screen, along with the backlight duration, and the Project button leads to the Current Project menu. Two-button shortcuts provide access to some of the other menus too (Menu-Project for the New Project window, Menu-Display for the System menu, Menu-Timecode for the Timecode settings menu, and so on). A data wheel is also provided to scroll around the menus, adjust values and move through the audio timeline when selecting edit points.

Along the front edge of the top panel are four transport buttons — Rewind, Forward, Stop and Play. Pressing Rewind and Stop together forces the cue point to the very beginning of the file. Over on the left-hand side is a non-latching power slide switch to turn the unit on or off.

The front panel is laid out in an equally simple manner. To the left is the recessed headphone/speaker volume control, with the large LCD screen dominating the left-hand side of the panel. This display is conveniently angled back slightly to make it easy to read both when the machine is placed on a table and when slung over the shoulder.

To the right of the display are six more recessed buttons. Timecode switches the timecode chase mode on or off, Retake deletes the last recording made, the backwards and forwards Locate buttons jump to the previous or next marker point, and Marker inserts a marker in the file. Finally, the Hold button is a sturdy slide switch that locks out all the other keys to prevent accidental operation. Sensibly, pressing the Retake button brings up a dialogue box asking if you are sure you want to delete the last take. This gets very tedious if you need to do a lot of retakes, but the dialogue box can be circumvented by pressing the Stop button at the same time as the Retake button, which is a thoughtful feature. Set back into a separate recessed section are the transport Pause and Record buttons, each with bright LEDs to confirm their current status, and, last but not least, a dual-concentric input level control.

Menus

In normal operation, the LCD screen carries all the essential information required when recording or playing back a file. There is a horizontal stereo recording level meter, a timeline display, the current project and file names and last marker number, the capacity of the pre-record buffer (if active), the recording settings (sample rate, wordlength, mono/stereo and sync reference), locator time value, and various icons to indicate the transport and connection status — external or internal power, Hold lock on/off, transport mode, Firewire connection, and remaining Compact Flash card capacity. This last item is one of the more critical bits of information on display, so it seems odd that it should be consigned to a small graphic in the bottom right-hand corner. However, in practice, I found it easy enough to read.

By pressing the Menu key on the top panel, the display changes to show the main menu structure. Submenus are selected using the data wheel and Select button. As with most other aspects of this machine, the menu structure is intuitive and largely self-explanatory, and the handbook quickly clarifies the few slightly less obvious aspects. For example, one slightly unusual facility tucked away in the Project Settings menu is that the headphone output can be set to provide stereo, summed mono, mono from left or mono from right signals or to 'Follow Record Mode'. The last option is probably the most sensible, otherwise there is the possibility of ending up monitoring something other than what is being recorded without realising it!

A more useful option in this submenu is that markers can be inserted automatically whenever the input is overloaded and/or if the timecode input disappears. The meter decay rate, peak hold time and clip indicator duration can also be set here, as can the default file name, the timecode and chase settings, input (analogue or digital) and clock sources, sample rate and word length. Another useful facility is that project templates can be set up, allowing the user's preferred settings for different applications to be established and recalled quickly when preparing the machine for a new recording session.

The Systems menu allows the real-time clock, date and time stamp to be set and provides the option of audible beep alerts for low battery and memory capacity, media management tools (Get Info, Clean Up, Scan, Erase and Format), and the means to install firmware updates. A very handy facility here is the Media Speed Check function, which writes and then reads some test files to assess the speed of the Compact Flash card. It then displays a table showing the expected status for the full range of sample rates and file formats (mono/stereo and 16/24 bits). Using 1GB 80x Lexar cards the test claimed that all modes were acceptable, but with slower cards it is likely that the high sample rate modes would be unreliable or even unworkable. Firmware updates are performed simply by copying the appropriate upgrade file to a blank Compact Flash card, and then booting the machine with the card installed.

The timecode facilities are more than adequate for most applications, with a comprehensive selection of frame rates and pull-up/down modes. The internal sample rate can be synchronised to an external timecode feed if required, and when chasing timecode for playback, an offset value can be entered or a specific timecode frame can be set to trigger playback. The freewheel period can also be specified to determine how long the machine will continue in play or record once timecode disappears. The HDP2 will automatically stop if the timecode value exceeds a 24-hour time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period after the default start time of 01:00:00:00. So, for example, if you were recording after midnight, the machine would stop automatically at 00:59:59:29! To overcome this limitation, the Timecode Origin time can be set to something more appropriate to ensure the actual recording time falls within the 24-hour window allowed by the machine.

Transferring Files

The HDP2 is equipped with a Firewire interface and is recognised by Windows XP and Mac OS X v10.3 as an external Firewire drive. Unlike USB 2 Plug and Play interfaces, the Firewire connection has to be 'undocked' through the computer's software before unplugging. Once undocked, the HDP2 reboots automatically and then functions normally. If the host Firewire port provides power, then the HDP2 will select this in preference to internal batteries or even its own power supply, depending on the voltage.

An alternative to using Firewire is simply to remove the Compact Flash card and insert it in a standard card reader. This was the way I made most transfers, simply because that is the way I generally work when accessing digital camera pictures. Using a USB 2 card reader the download time is minimal, though obviously dependent on size and number of files.

Whether accessed via Firewire or directly from the Compact Flash card, the file structure is clear, with individual mono or stereo audio files appearing within separate Project folders, all with the appropriate default or user-defined names. Maximum file sizes are determined by the card's formatting — the FAT16 or FAT32 file systems can be used — but if a recording exceeds the file size limit, the first file is closed and a new one started seamlessly. When imported into a DAW, the files are placed automatically according to their time stamp and join together with sample accuracy. If your DAW software can't handle this arrangement, Tascam also supply a stand-alone software application for Mac and PC that will automatically conform a number of files contained within a project into a single contiguous file. Various reformatting options are also provided such as changing the word length and channel format.

A Delight To Use

I found the HDP2 a delight to use, and the more I used it the more I liked it. There are a few little operational traps for the unwary, but overall it is a well-designed, straightforward tool that does exactly what it is supposed to.

Synchronising the unit to external references and recording from digital sources is very simple and I couldn't fault the performance or quality at all. When recording analogue sources, the line-level inputs performed very well, indicating the presence of good-quality converters and electronics. However, the real test is what happens when decent mics are plugged in and I'm pleased to report that the quality of the HDP2's mic preamps is surprisingly good. I was able to make very creditable location recordings using an M&S pair of Sennheiser MKH mics, and recorded a couple of indoor interviews using cheaper electret mics — both relying on the machine's own phantom power supply.

I had no problems setting levels to optimise noise and headroom, and the preamps are of a similar standard to those of a decent compact mixer, which is pretty good for a device at this price. Of course, with 24-bit converters, you can afford to leave a significant amount of headroom and still acquire very usable material, so the limited gain range of the preamp isn't a problem in practice.

Although there is no M&S decoder facility in the input or monitoring chains, you can at least select to monitor only the left input on both ears, which makes working with an M&S mic array much easier than it otherwise would have been. I didn't use the limiter in earnest since I prefer to allow sensible headroom margins, but on testing it seemed effective and reasonably benign in action. The high-pass filter is perhaps a little heavy-handed, but very effective in reducing rumbles and wind noise.

I was reassured by the large and obvious warning displays when the batteries are running low or the Compact Flash card is nearly full — it's embarrassing to say the least if the machine stops halfway through an unrepeatable take! Similarly, another important feature to note is that the HDP2 continuously rewrites the file header as a recording is being made. This means that should there be a break in the power, the entire file isn't lost — only the last tiny portion captured after the last file header update will be lost. Having worked with a lot of Minidisc and first generation card recorders in the past, I can vouch for just what a life saver this can be!

The only real disappointment is that the internal editing functions are too basic to be a lot of use. Files can only have their start or end trimmed. It is not possible to delete portions within a file, or to split a file into two to trim the ends and then rejoin. Of course, most people would prefer to load the files on to a computer and use the far more effective editing facilities of their DAW software, but sometimes you have to fix the recording while out on the road, and the options are severely limited with the HDP2 as it stands.

Summing Up

All in all, this is a well-thought-out recorder that provides very serious competition not only for the Fostex FR2, but also some of the more upmarket professional offerings from the likes of Sound Devices and Nagra. Although a little more expensive than some broadly similar units from Marantz and others, the margin is not that great and many potential purchasers may feel the HDP2's more comprehensive feature set and better analogue electronics more than justify the difference too.

The HDP2 is easy to use, well specified, and flexible to configure. The inclusion of the timecode record and chase facilities expands its potential uses and market considerably, and the adoption of the Broadcast Wave file format makes its data files universally accessible. For the time being, this neat Tascam machine sets the benchmark in terms of features at this price, and will sit at the top of my wish list! SOS

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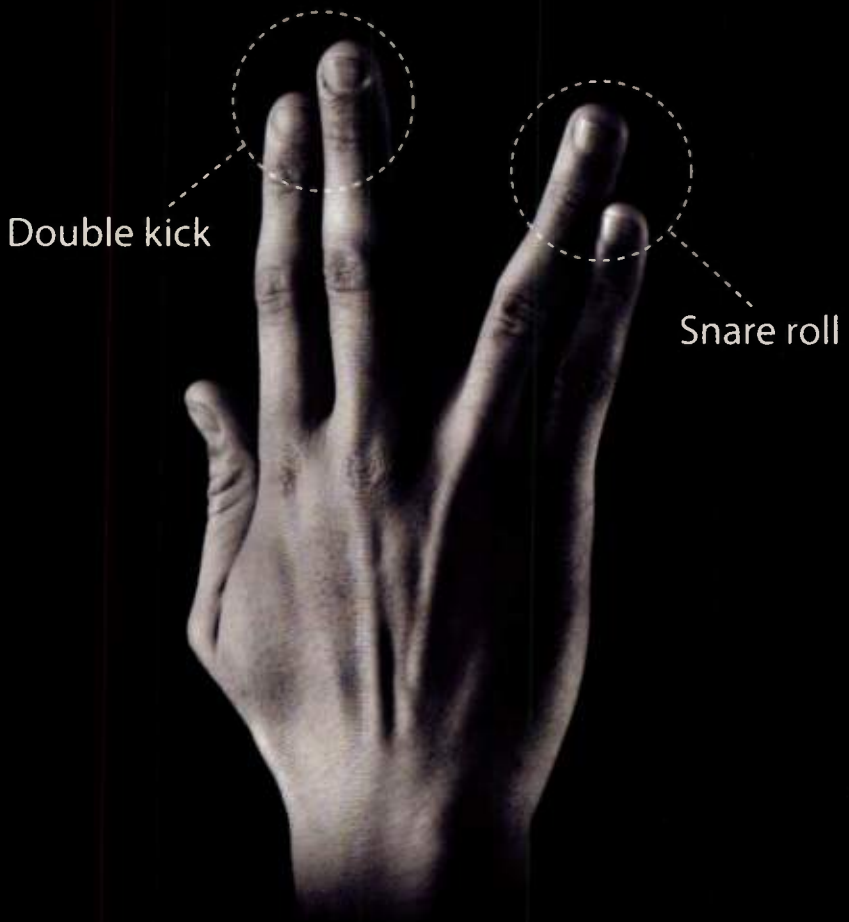

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Eventide Anthology II

Plug-in Bundle For Pro Tools TDM Systems

Mike Thornton

Eventide's range of high-quality Pro Tools plug-ins continues to grow, and the new *Anthology II* bundle collects together no fewer than 15 separate effects and processors.

Since they launched the *Clockworks Legacy* bundle three years ago, Eventide have been adding to their range of TDM plug-ins, and the new *Anthology II* package combines that original suite of effects with the more recent *Anthology* and several even newer products. The result is a suite of processors and effects that brings together 'retro' and contemporary sounds not only within one bundle, but also in some cases within individual plug-ins.

Clockworks Legacy Bundle

This bundle consists of precise software models of the Eventide hardware favourites from the 1970s — the Instant Phaser, Instant Flanger, Omnipressor, H910 and H949 Harmonizers. Eventide have resisted the temptation to 'update' them, other than to add some MIDI control and automation of course, preferring to reproduce them warts and all as software plug-ins. *SOS* reviewed this bundle back in September 2003 (www.soundonsound.com/sos/sep03/articles/clockworks.htm) so I am going to skip past these processors other than to reaffirm that *Instant Flanger* is still one of the best flanging effects around. I have always felt that most flanging processors don't quite match the original tape-machine-created effect, but *Instant Flanger* does it for me.

The only other things I want to mention are to bring up to date some of the findings of the 2003 review. On the reviewer's Mix system, the *Harmonizer* plug-ins took a whole chip for one mono instance, but I can report that on my HD2 Accel system, *H910* took 25 percent of one of my HD Accel chips and *H949* took 50 percent of a chip. I too agree that for sci-fi effects these plug-ins are the business, and I wouldn't be at all surprised if Paul McFadden, who is part of the audio post-production team for the current *Doctor Who* series, has them in his 'toolbox'.

Finally, I can't leave this section without commenting on the Omnipressor emulation. I too took a little time to get to grips with the Attenuation and Gain Limit controls and what the meter in Gain mode was telling me, but having seen the logic behind it, I am very taken with being able to limit the amount of gain or attenuation a compressor puts into a signal path; this makes it possible to have a quite high ratio setting whilst making sure the compressor can only apply so much gain or attenuation. I would value a make-up gain control on the *Omnipressor* plug-in, but I appreciate that would step away from the faithful model of the original hardware unit.

Anthology Bundle

Eventide later added four more plug-ins to the *Clockworks Legacy* bundle to make the *Anthology* bundle. These, again, originated in Eventide's hardware processors, but they are much more contemporary in origin. Two of the new plug-ins, *Band Delays* and *Factory*, were derived from the H3000-series Harmonizers, and the other two, *Reverb* and *Octavox*, are lifted from their flagship Orville hardware unit.

Band Delays (right) and *Factory* share a similar graphical user interface, which is divided into three sections. At the top there is a virtual H3000 front panel, while the middle is taken up by a Preset Parameter section offering controls such as wet/dry mix, input and output levels, filter settings and tempo-related controls. The bottom section can be set to display in three different modes: Program is the simplest and the most graphical, Expert lists all the settings in a table format, and Function gives you access to things like soft-key settings, and a Function Generator for controlling sweep effects and so on.

Band Delays comprises eight tempo-based multitap delays, each with a programmable filter, feeding back into a stereo mixer. The filters can be set to low-pass, band-pass or high-pass, and can be 'played' using MIDI. The Program section of this plug-in really uses the concept of a graphical interface to the utmost. The left-hand side shows the eight delays, each colour-coded with an 'X'. The right-hand side shows what is happening in a 3D chart, with the X axis showing frequency, the Y axis showing filter gain and the Z axis showing delay length. These track with any modulation programmed in. Try running through some of the presets and watch the graphs bounce around. It is definitely a case of a picture painting a thousand words!

SOUND ON SOUND

Eventide *Anthology II* £840

pros

- There are so many different types of plug-ins in the one package, from EQs and compressors to delays and Harmonizers.
- Combines 'retro' with contemporary plug-ins in the one bundle.
- It is excellent value for money at under £60 per plug-in.
- *Ultra Channel*, *Factory*, *Reverb* and *EQ65* are all stand-out plug-ins.
- There is an excellent range of upgrade paths.

cons

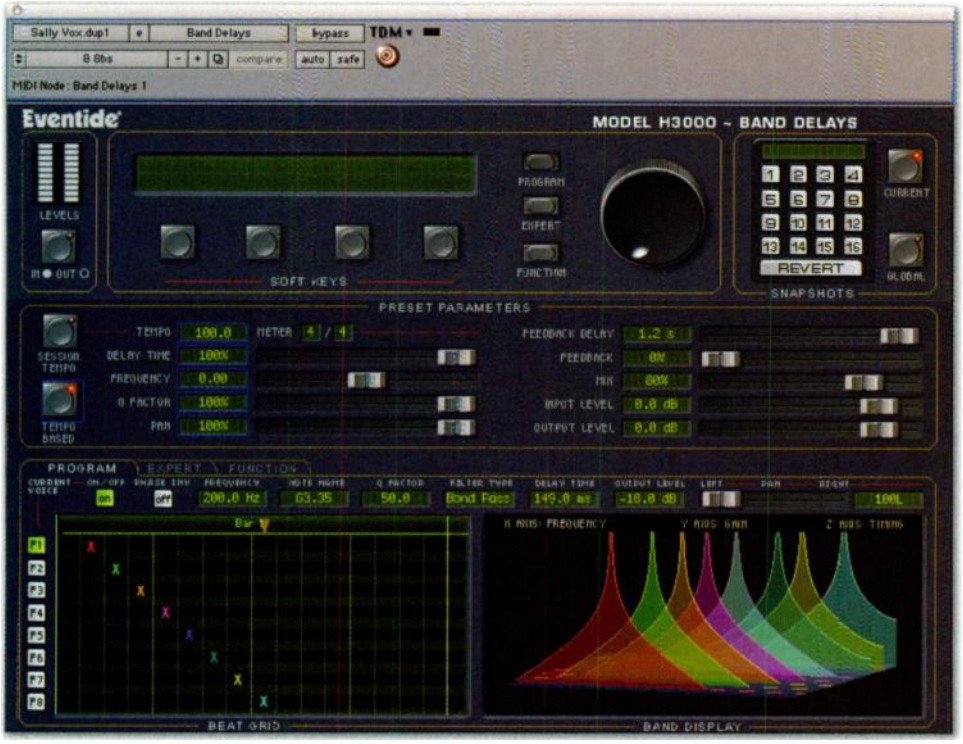
- No surround versions.
- De-esser needs improving.
- *Quadravox* doesn't seem to be less DSP-intensive than *Octavox*.

summary

Eventide's *Anthology II* is a comprehensive bundle of good-sounding plug-ins, combining modern and retro effects and processors.

Alternatives

Waves' *Gold* bundle includes direct alternatives to almost all of the processors and effects in the *Anthology II* bundle, plus a number of other plug-ins, but the TDM version is rather more expensive than *Anthology II*. However, there's no obvious equivalent to the *Factory* plug-in in the Waves range, and perhaps the only real alternative is DUY's rather more complex modular effects plug-in, *DSpider*.



What is it like to use? Well, the enhancements Eventide have added to the plug-in version, together with the ability to sync it up to the song tempo, mean that if you want complex multiple delays then *Band Delays* can deliver with the correct programming. It takes a bit of getting used to, but I found a trip through the presets gave me a very useful tutorial on what is achievable and then you are really only limited by your imagination and time in achieving some really wild delay-related effects. For example, *Band Delays* makes it very easy to set a stereo slap echo: try using four delays alternatively panned left and right, spaced a quaver apart and setting low-pass filters to roll off the high frequencies with each successive repeat. I was impressed, although I actually found it easier to set this same effect up on the Waves *Stereo Supertap* plug-in; while it doesn't have the dancing filter effects, the display is much clearer, and it gives you access to all the controls, rather than having to select the delay and then adjust a common set of controls.

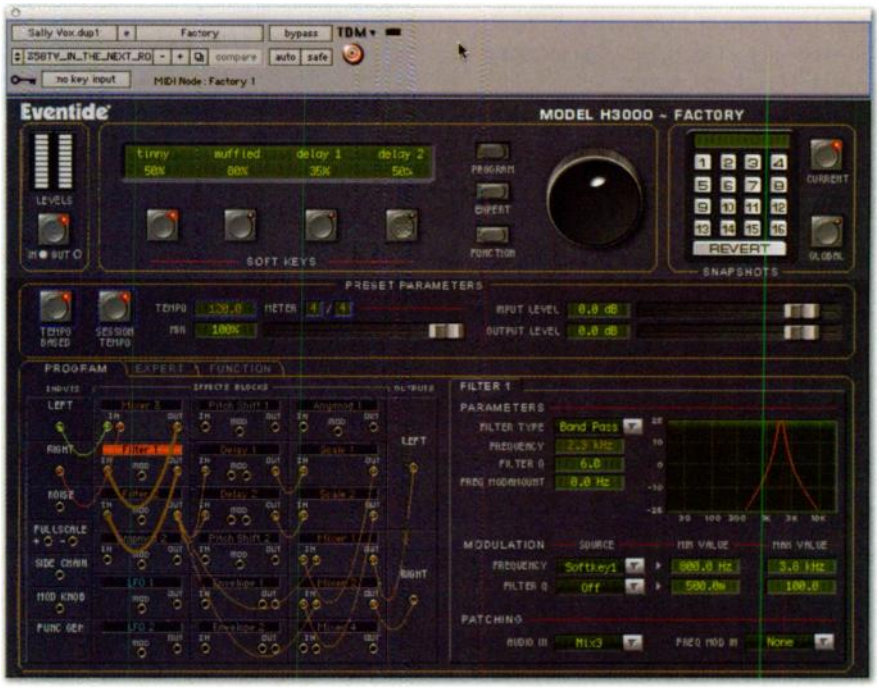
Factory (right) is a modular processor offering two of each of the following modules: delays, filters, virtual VCAs called Ampmods, Scale modules that can be used for either audio or control signals, LFO modules and envelope generators, plus four two-input/one-output mixer modules. In Program mode, the bottom section sports a patch panel just like those early synths, so you can configure the modules in the desired order. However, the quickest way to understand how these can be used is to run through the presets and see how the effects have been created. I was particularly impressed with the section of 'post' friendly presets in the list.

It is difficult to compare *Factory* with any other plug-in from any other manufacturer, as to me it seems pretty well unique. The possibilities are endless, so I'll just offer one example of the sort of thing that *Factory* can do: to remove the snare drum spill from the overhead mics on a drum kit rig. You know the scenario — you spend ages getting the snare sound just right, maybe with a gated reverb sound, and then you open up the overhead mics and it all changes. So what if you could duck out the snare from the overhead channels? Well, with *Factory* you can. Route the overhead mics through the two Ampmod modules and feed the key input from the snare mic through a band-pass filter set to only pass the snare sound and reject the toms, cymbals and so on. Route the output of the band-pass filter to an envelope generator and take the 'ducker' output into the control input of the Ampmods, and by adjusting the envelope generator's attack and release times, the snare sound is ducked out of the overhead channels.

I found that the control surface implementation was very impressive and soon was adjusting paramenters direct from my Command 8.

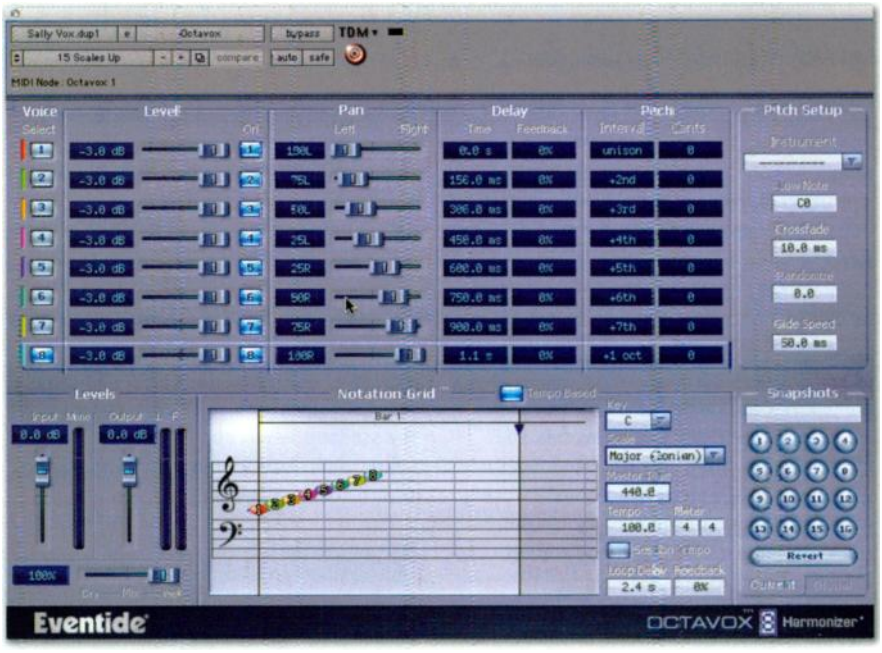
Reverberation

Reverb is not a particularly imaginative name for a reverb plug-in, especially when you realise that it is a whole load more than just a reverb unit. It includes no fewer than four EQ modules (pre EQ, reverb EQ, delay EQ and post EQ), plus a compressor that can be positioned either before or after the reverb. The reverb section includes the usual controls such as decay, pre-delay, size and so on, but there is also a Lo-fi control to enable you to wreck your lovely clean sound, decreasing the bit depth as you increase the percentage. To cap it off there are two delay lines, one in each output, with up to one second of delay. All of this makes for a very powerful reverb. ▶



I had heard that the basic reverb algorithms weren't anything special, but for me they compared remarkably well with Waves *Renaissance* and Sony reverbs, although it would be fair to say that if I wanted a 'quality reverb' alone I wouldn't go straight for Eventide's *Reverb*. Having said that, I tried replacing a Waves *Renaissance Reverb* on a strings subgroup with the 'Strings Chamber' preset on the Eventide plug-in, and very quickly I had a much lusher sound, so there is definitely some mileage to be had in this reverb plug-in. Add the other features like the EQ options, compressor and delays, all in one plug-in, and *Reverb* is a very useful tool that stands out from other reverb options.

Octavox is an eight-voice harmoniser with a mono input and mono or stereo output. Each voice has up to 2.4 seconds of delay, pan, volume, feedback and two octaves of pitch-shifting. You can use a combination of preset intervals and pitch cents to produce the desired amount of pitch-shifting. The graphical display on this plug-in is similar to the H3000 emuations but is presented in a more musical way, with a stave which displays both the pitch intervals and the delay loop points. Again, a good range of factory presets give you a good starting point to get the creative juices going. I tried using this on a track where I had originally used Waves *Ultrapitch* to thicken out a low-level vocal loop, which I had actually created from the original guide vocal and liked it so much it stayed in the final mix. I would have to say that *Ultrapitch* and *Octavox* are very similar, but if pushed to point out a difference I would say that *Octavox* produces a smoother sound and *Ultrapitch* a richer sound. If you don't have *Ultrapitch* then you will find this an excellent tool for thickening out vocals, especially backing vocals.



Anthology II

So to the new plug-ins that Eventide have added to the *Anthology* bundle to turn it into the *Anthology II* bundle. The *EQ45* parametric equaliser is a recreation of their vintage analogue EQ unit, including 12dB-per-octave high and low cut filters as well as four bands of fully parametric EQ. I have got so used to graphical representations of filter curves and the like, especially on EQ plug-ins, that I felt a little blind presented with just a set of knobs. It is interesting how soon we adjust to new user interfaces! On a control surface or an analogue desk, just having knobs isn't a problem, but when working in Pro Tools I have got so used to having a graphical display to see what the EQ is doing that not having it would be, for me, a good enough reason to go and select a different EQ that did. Perhaps Eventide could consider adding a graphical section to the user interface whilst retaining the model of the original analogue EQ?

The *EQ65* filter set is a recreation of an analogue filter set, with 18dB-per-octave high and low cut filters and two tunable notch filters with a depth of up to 150dB! There is also a very high Q setting, so it

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▶ really enables you to filter out any troublesome tone-type problems with minimal impact on the wanted audio. To put this to the test I set up a *Signal Generator* to output a 1kHz sine wave tone, and then inserted an *EQ65* after it. At first I thought there was a problem, because the tone disappeared completely, but as soon as I adjusted the filter frequency, back came the tone. So I went back to the signal generator and I adjusted the frequency of the tone to see when it would reappear and the 3dB points were at 1035 and 965 Hz — that's a very narrow notch! Again, there isn't really anything to compare this with, but the deep notches make it an excellent tool for solving problems (or should that be 'opportunities'?).

You don't normally associate channel strips with Eventide, but when you look at the range of plug-ins they now offer, it is a sensible and logical move to combine them in this way. *Anthology II* includes two, *Ultra-Channel* and *E-Channel*. The former includes a gate, de-esser, compressor/limiter with side-chain, a separate Omnipressor, a five-band parametric EQ section, stereo delays, and a Harmonizer micro-pitch-shifter for thickening. All the modules except for the last two can be reordered by dragging and dropping to your preferred sequence. However, *Ultra-Channel* is a chip-hungry plug-in — one instance takes a complete Accel chip — and the point of *E-Channel* is to enable you to have multiple instances without eating up too many chips. This channel strip still includes a gate, compressor limiter, and a five-band parametric EQ section. One instance takes 17 percent of an Accel chip, but I actually managed to get six instances onto one chip.

I tried both channel strips on a number of Sessions and found them both very useful. I especially liked *Ultra-Channel* on solo vocals, particularly with a dash of Harmonizer thickening. I was also able to quickly get a very nice bass guitar sound just using *E-Channel*. Unlike the version from the *Clockworks Legacy* bundle, the Omnipressor on *Ultra-Channel* has a make-up gain control as well. However, I have several other observations on the channel-strip plug-ins. Firstly, the de-esser on *Ultra-Channel* didn't do it for me: I couldn't get the vocal free of sibilance without messing up the sound, so I found myself reaching for my preferred de-esser again. Second, although *Ultra-Channel* is available as a stereo plug-in, *E-Channel* doesn't come up as stereo plug-in on a stereo track — you have to use it in multi-mono mode. Finally, I found the limits on the range of the EQ sections a pain; for example, the HF shelving only comes down to 5kHz.

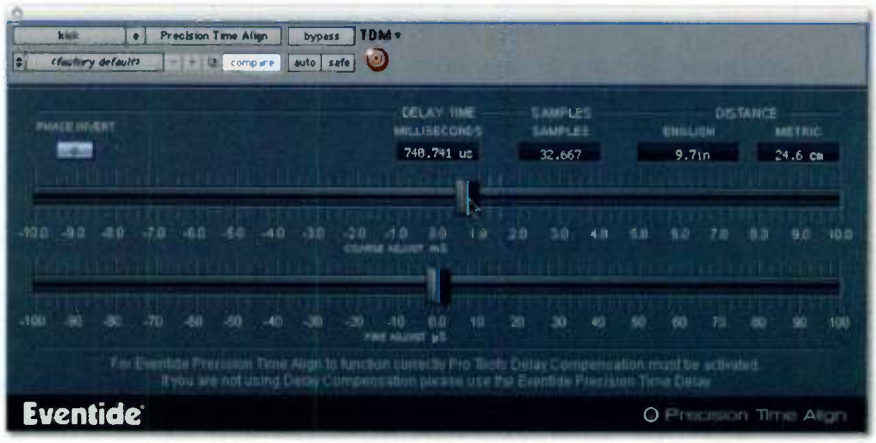
Eventide have included the *Quadravox* plug-in as a cut-down version of *Octavox* from the original *Anthology* bundle; like *E-Channel*, it's designed to enable you to use multiple instances without eating up too many chips. This is a fine idea in principle, as most of the time a four-voice harmoniser is all that is required, and there's no point in wasting DSP resources on unused features. In practice, however, both the *Octavox* and *Quadravox* plug-ins took up a complete chip per instance on my Accel

system, so there was no gain. As to the sound of this, it is no different to *Octavox*, so all my comments about *Octavox* apply to *Quadravox*.

The final plug-in in the *Anthology II* bundle is called *Precision Time Align*, and as the name suggests, it's intended to offer sample-accurate positive and negative time alignment of individual tracks. Actually there are two versions of this plug-in: *Precision Time Align*, which will only fully function if you have enabled delay compensation in *Pro Tools* (if you want to use negative time delay, delay compensation must be enabled), plus a second version called *Precision Time Delay* for those who don't use delay compensation. What I like about both these plug-ins is that they display the time adjustment not only in time and samples, but also in distance (Imperial and metric!), so if you know that two mics were nine inches apart, you can adjust the plug-in until the distance display reads nine inches.

Conclusions

Eventide have put together a very interesting bundle in *Anthology II*. For me, it certainly contains some surprises, including the channel strips, *Factory*, *Reverb* and *EQ65*. It is also excellent value for money: 15 plug-ins for £840 means each plug-in comes in at £56, and there is an excellent range of upgrade options for those who already own some Eventide plug-ins. Is it worth it? I would say that if you don't own one of the larger Waves bundles, *Anthology II* is a compelling alternative, and even if you do have a good number of Waves plug-ins, there are enough different plug-ins in this bundle to make it worth considering. At under £60 per plug-in it is hard to say no! SOS



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Refining Rhythm In Reason

Last month we took a look at some basic applications of the *Redrum* module in *Reason* drum programming. Now it's time to move on to more sophisticated techniques for your rhythm parts.

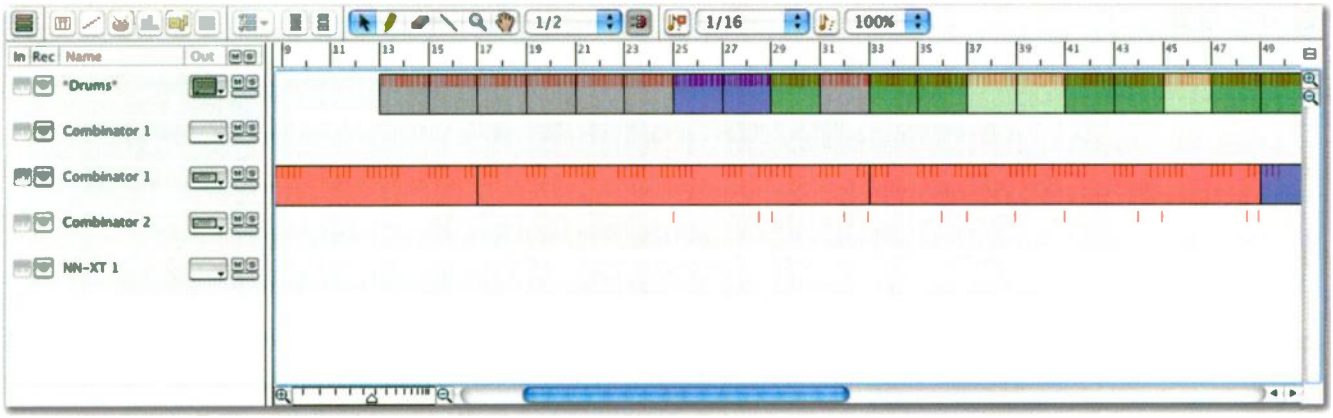
Simon Price

In last month's *Reason* workshop feature, we looked at the *Redrum* drum machine, its built-in step sequencer, and how to build drum arrangements by chaining patterns together. At the end of the process, we converted the whole arrangement into individual note events in the sequencer, using the Convert Pattern Track to Notes command. This time, we'll look at some of the ways in which this data can be manipulated. The same techniques also work with drum tracks created using the *NNXT* sampler.

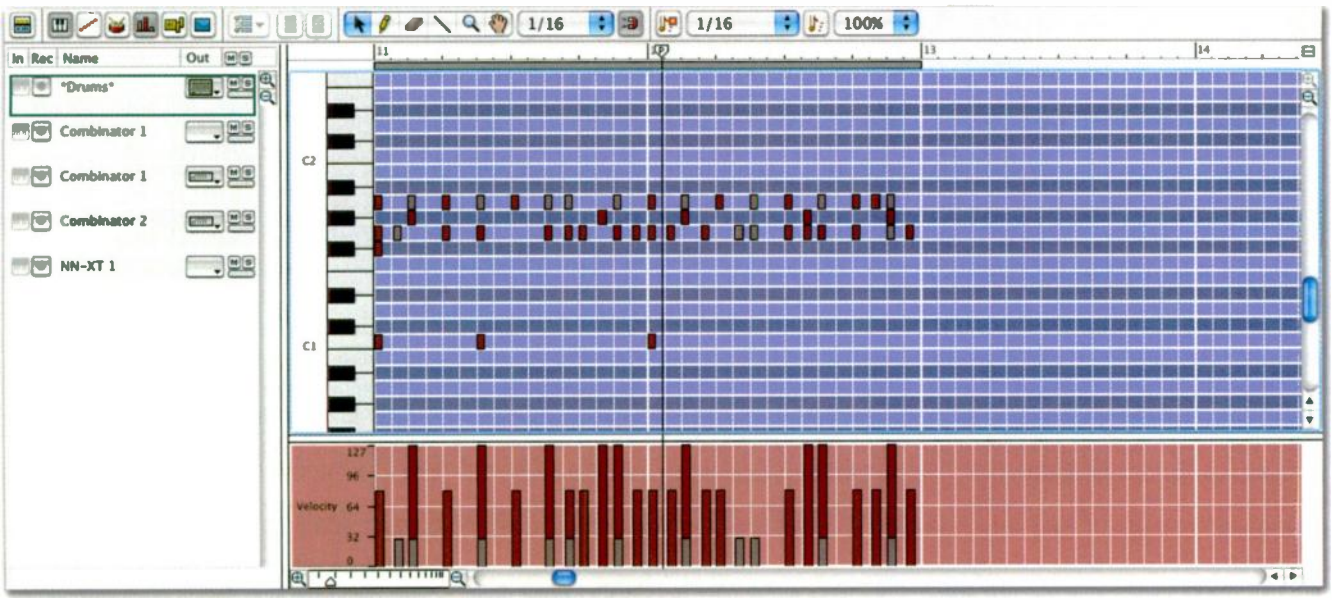
Firstly, why bother getting into all this when you've got the perfectly adequate pattern sequencer inside *Redrum*? There are several answers to that question. Firstly, if you want to make small alterations to data in some places and not others, it may be more practical to do it with individual notes than to create a new pattern and slot it into the pattern track. Also, when you use *Reason*'s traditional 'linear' sequencer in its Arrange View (all tracks showing), it's much easier to see what's going on in drum tracks if they contain notes rather than patterns (although you can have both). A more musical consideration is that *Redrum* patterns only allow three velocity levels per sample, so if you need anything subtler or more expressive than this, you need to use notes. Also, while the pattern sequencer can have its timing shuffled, if you want to groove-quantise the drums, or do anything that's not strictly grid-based, patterns don't cut it. Finally, there are other instruments in *Reason* that can be used for drums and percussion, in particular the *NNXT* sampler, and using the linear sequencer will allow you proper access to these.

Arrangement Groups

Before getting down to the nitty-gritty, let's have a look at how arranging drum tracks as notes differs from using patterns. The first screen opposite shows an arrangement in the sequencer Arrange View. The top track is the drum part I discussed last month, which we converted to notes using the Convert Pattern Track to Notes command (under the sequencer's Edit menu). The track is divided into coloured blocks, called Groups. Groups act as single objects, so are easier to deal with than big clusters of individual notes. (They're useful in many situations besides drum programming: see the 'Get Into Groups' box for more details.) If you record drums in as notes using the keyboard, there will be no automatic Grouping on the track, and you'll just see a series of red lines, as in the '*Combinator* 2' track in the screenshot. The reason why our drum track is nicely grouped is that when you use the Convert Pattern Track to Notes command, *Reason* cleverly places Groups where each pattern was in the pattern track. What's more, each instance of the same pattern will become a group with the same colour. In the regions where we used an empty pattern to drop the drums out



The notes created by the 'Convert Pattern Track To Notes' function are arranged into colour-coded edit groups.



Double-clicking a group in the Arrange View switches you to Edit View.

entirely, no Group is created. For large-scale arrangement changes, you can pick up and move, or copy and paste Groups.

The second screen (above) shows what you see after double-clicking the first Group in the track, to open the Edit View. A useful feature of *Reason*'s sequencer is that it remembers the display settings you've set up in each of the two views (Arrange and Edit). These settings include zoom factors and which of the data 'lanes' are displayed. In the screenshot, the piano-roll (notes) view and the velocity view are showing.

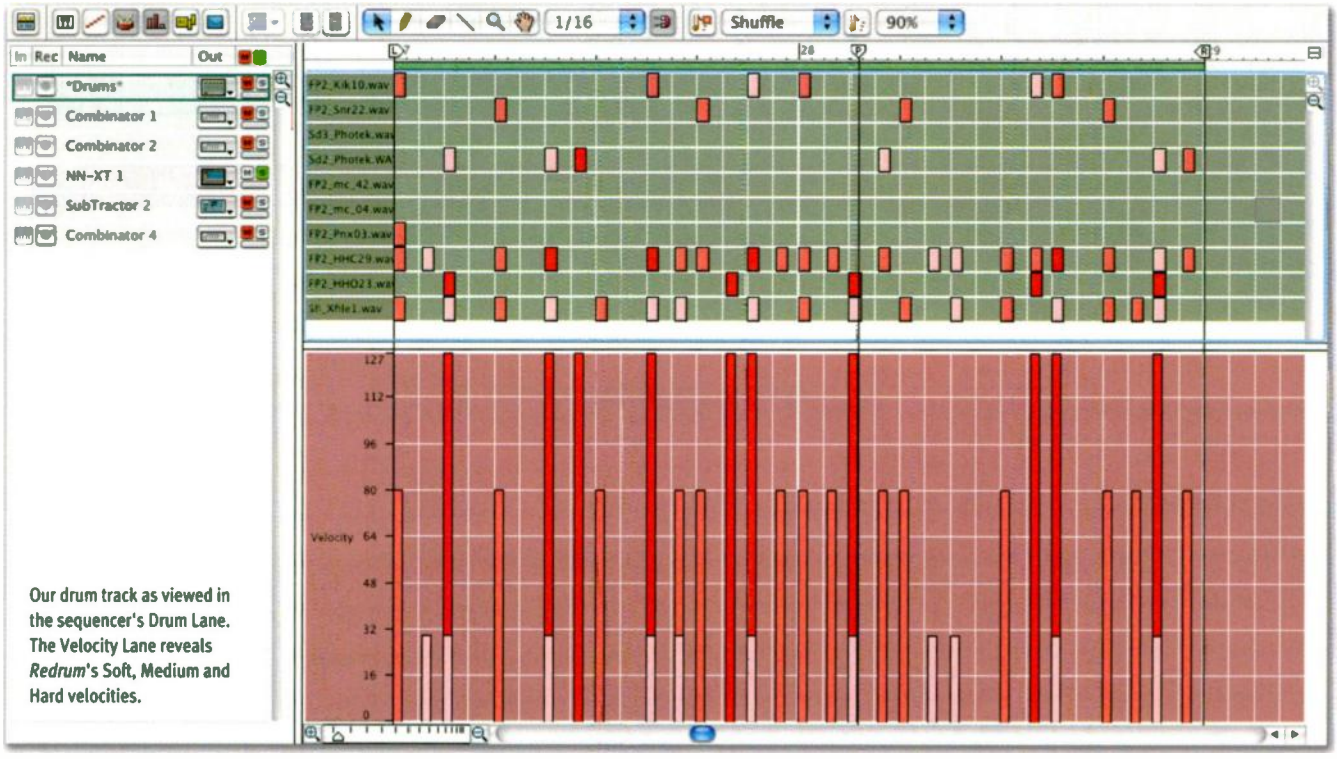
Check Out This Groove

With all the notes selected, this is a good time to try some groove quantising. I like to do this in the piano-roll view, as it's then easy to see the results of quantising. Choose a groove from the pop-up menu in the sequencer's tool bar (see the bottom screenshot overleaf). There are three preset Grooves, plus Shuffle and the User groove. (Note that if you had Shuffle enabled on *Redrum* when you converted the pattern to notes, the shuffle will already have been written into the sequence.)

The User groove is a clipboard that temporarily stores a groove template that has been created elsewhere in the song. This has been discussed before in these pages, but is worth a recap, as it is one of the most powerful ways to make different tracks in a song sit well together. You first select a sequence of notes from any instrument, then choose Get User Groove, from the Edit menu. *Reason* will analyse the selection and create a groove template based on how the selected sequence deviates from the grid. The selection must therefore have some rhythmic feel to it, or your User Groove will be no different to a regular grid quantise. Normally, you would select one or two bars from the performance you want to 'sample'. A common trick we've ▶

Get Into Groups

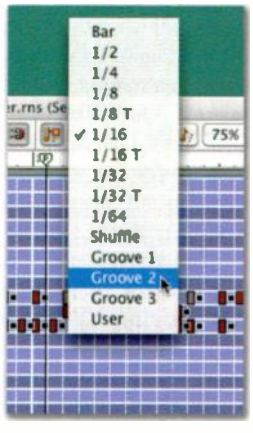
Groups are commonly seen in the *Reason* sequencer as a result of using the 'To Track' button in the *Dr Rex* loop player, but they can be used freely in any track and make arranging in *Reason* much easier. To create a Group, select a range of notes in any sequencer track and press Command-G (Mac) or Ctrl-G (Windows), or choose Group from the Edit menu. The selected area will become a coloured 'brick' that can be moved and edited as a single object. Different Groups are coloured differently, but Groups with identical contents are automatically assigned the same colour. Double-clicking a Group opens the Edit View, with the Group's contents selected and placed at the start of the viewable area. If you trim a Group by clicking at its edge and dragging, the bounds of the Group will be extended. This does not affect the contents, it merely extends the Group to encompass more adjacent notes. If there is another Group adjacent to the first, this will automatically be shrunk, as groups cannot overlap. Multiple Groups can be combined by selecting them and choosing Edit / Group again, or you can extend the first group until it encompasses all the others. You can also edit the boundaries of a Group with the Pencil tool.



Our drum track as viewed in the sequencer's Drum Lane. The Velocity Lane reveals *Redrum*'s Soft, Medium and Hard velocities.

mentioned before is to find a REX loop that has a groove you like and use the Copy to Track command to create notes in the sequencer from it. You can then use 'Get User Groove' on these notes and apply it to your other tracks, instantly forcing the components of your song to gel together rhythmically.

Getting back to our selection of notes, now that you've chosen your quantise type, choose a percentage from the pop-up to the right. If your drum sequence came from a *Redrum* pattern, this value determines how far the hits will be moved away from the grid. If you recorded the drums directly into the sequencer, the quantise percentage controls how far notes are moved from where they were recorded towards the groove template. Most grooves tend not to affect notes that fall on the beat, so applying groove quantise to a MIDI recording will tend to tighten up the timing, as well as introducing the groove. However, if your timing was quite sloppy, you can first apply a grid quantise (typically using the 1/16 setting), and then apply the groove quantise afterwards. Once you're happy with the results of your quantising, you can switch back to the Arrange View (click the first button in the sequencer's toolbar) and apply it to the rest of the track.



Drum tracks created by step sequencing can be 'humanised' with a little groove quantising.

Editing Notes

There's much more you can do to your drum tracks in the sequencer — in fact, your preference may be to start work here, drawing in notes on the grid. Notes can be added, moved, deleted and extended, and may have their velocity altered. In the previous example, we worked in the piano-roll, or 'Key Lane' view. There's also a Drum Lane, as shown in the screen above. The data displayed here is exactly the same as in the piano-roll, except that only the 10 *Redrum* notes are shown, and the note order is inverted — i.e the channels run top to bottom. If you're working with drums in an *NNXT* sampler, the Drum Lane shows all 128 notes, simply labelled Note #1, Note #2, and so on.

To add a note on the Drum Lane or Key Lane, use the Pencil tool. The best method is to stay in Selection tool mode and hold down the Command key (Mac) or Alt key (Windows) to temporarily access the Pencil. This is because you can't delete notes with the Pencil tool, and if you're experimenting with a pattern you'll want to add and remove notes on the grid quickly. You should have Snap mode active, and mostly likely set to restrict your drawing to 1/16ths. If you click in the grid, a note 1/16th long will then be created; however, if you need to create longer notes you can click and drag. To

The 'Converted Notes' Sustain Problem

Reason is very clever in the way it creates note sequences from pattern sequences and groups the patterns, but there is one problem that can occur when you do this. The screenshots in this article reveal that the notes created by converted patterns are very short. This is fine, except for notes sourced from any channels in *Redrum* that have their Gate/Decay switch set to Gate (see picture). When played from the sequencer, channels in Gate mode only sound for as long as the note triggering them is held. This means that they will be short and clipped off after you convert patterns to notes. In some cases, you may just want to switch the channels to Decay mode and set an appropriate Length. If that doesn't sound right, the solution is to select all the notes in the sequencer that trigger the problematic channel and extend them.

The two-position switch (centre right of the channel) sets Gate (top) or Decay mode for the sound going through that channel. The choice of position may be significant when a pattern is converted to notes.

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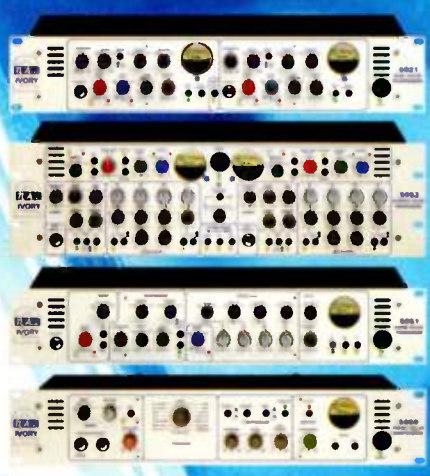

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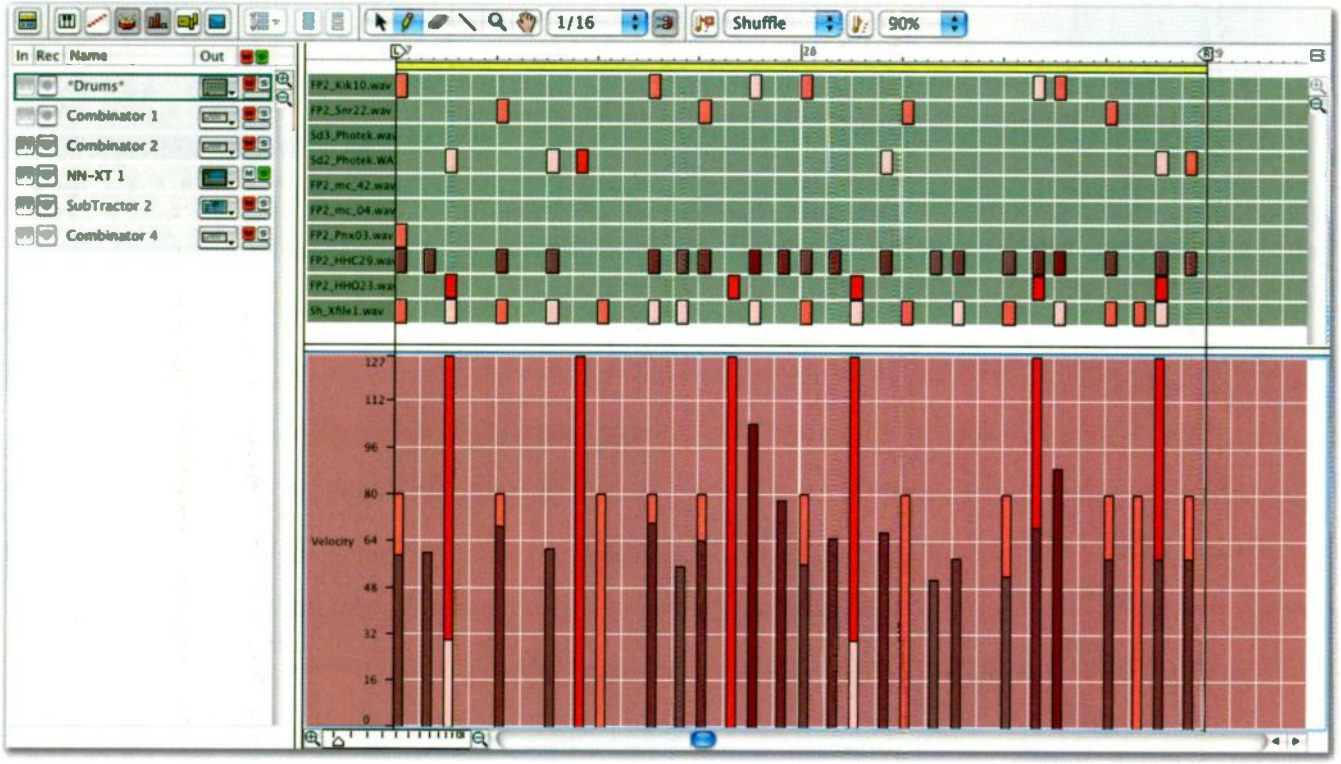

Here, the Pencil tool has been used to add some velocity variation to the hi-hat part.

create notes whose length is not restricted to multiples of the grid, hold down the Shift key. The initial placement of the note will still snap to the grid, but you can drag it to any length. If you need to remind yourself which note plays which sample, click the name of the note if you're in the Drum Lane, or click the keyboard graphic if you're in the Key Lane.

Velocity Editing

If your drum track started life as a *Redrum* pattern, the Velocity Lane will look something like the one in the top screen on the previous page, with most drums' velocities hard-quantised to just three values. Some variation in velocity will greatly enhance the feel of your drum programming. Velocity has an even greater effect on the *NNXT*-based acoustic drums, which respond differently to different velocities. Velocities are edited using the Pencil tool, or you can use the Line Drawing tool to set several notes to the same velocity or create crescendo or diminuendo effects.

Looking again at the top screen on the previous page, you may spot a problem: how can you change the velocity of the first kick drum when there are three other hits at that point, all with their velocity bars on top of one another? If you draw a velocity value at this point with the Pencil tool, all four hits will be affected. What you must do is click the note you want in the drum lane, which brings its velocity bar to the front, then hold down Shift while drawing in the value. This works across a range of notes too. For example, you might wish to introduce some velocity variation across all the hi-hat hits. To do this, first select all the hi-hats in the Drum Lane (or Key Lane). Now, while holding Shift, draw across the Velocity Lane with the Pencil tool, and each hi-hat that you pass over will be set to the Pencil's position (see screen above). As long as you keep using the Shift key, you will be able to make further changes to any of the hi-hats until you have the sound you want.

Final Flourishes

Once you've reached the point where your drum track has been fine-tuned and arranged, you may want to add fills, cymbal crashes and other points of interest throughout your song. It's much easier to do that at this stage than trying to account for it all from the beginning, when you don't know exactly how the arrangement for your song is going to turn out. The easiest method is to play through the song a few times and play along with your *Redrum* or *NNXT*, controlled from your keyboard or pads. When you know what you want, you can drop in and record. There are two options here.

First, you could record on to the existing drum track. If you do this, make sure you have the record mode set to Overdub (not Replace) in the transport bar. The other method is to use a separate sequencer track, which is safer and will make it easier to locate and edit your new additions. Choose Create / Sequencer Track, and name the new track 'Fills' or 'Extras', or whatever. Then, in the 'Out' column in the sequencer, choose the *Redrum* or *NNXT* and you have an extra track controlling the same device as your main drum sequence. SOS

Reason News

- **Free Refills: Line 6 are offering free 'Refills of the Month' at www.line6.com. A new download will be added each month for the rest of the year. The Refills are 'lite' versions and compilations of commercial Refills, and are each about 60-70Mb in size.**

- **Props On Tour: Propellerheads are taking *Reason* on tour around Britain and Ireland. The nine-date tour will stop at Sound Control stores in England and Scotland and at the London Calling exhibition and conference at Earls Court, before crossing the Irish Sea to Dublin. There are sure to be great give-aways at each event! Dates are as follows: 28th June, Sound Control, Bristol, 12-8pm (+44 (0)117 934 9955); 29th-30th June, London Calling at Earls Court, London; 1st July, Sound Control, London, 10-4pm (+44 (0)207 631 4200); 3rd July, Sound Control, Leeds, 12-8pm (+44 (0)113 242 6601); 4th July, Sound Control, Newcastle, 12-8pm (+44 (0)191 232 4175); 5th July, Sound Control, Edinburgh, 12-8pm (+44 (0)131 229 8211); 6th-7th July Dublin, Ireland (visit the www.futuresounds.ie web site for full details).**

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There are some pretty high-profile new features in *DP5*, including six rather good bundled instruments and a fast and flexible Meter Bridge. However, it's some of the less flashy and more unexpected developments that are receiving a lot of attention.

What's New

In *Digital Performer 5*

After being extensively trailed at this year's NAMM show and at Sounds Expo in London, *DP5* is finally here. But besides its shiny new additions, such as the six bundled MAS instruments, there are some unexpected and intriguing new features that might just prove to be more useful to some...

Robin Bigwood

When MOTU publicised details of *DP5* earlier this year, there was some grumbling. Various contributors to online discussion forums and email lists made the point that they'd be willing to forgo the bundled synths and Track Folders if only MOTU could make *DP* more efficient and banish the processor-spiking issue that had affected some users under *DP* 4.5 and 4.6. So it's encouraging to see that in *DP5* we've been given the synths *and* some intriguing tweaks to the MOTU Audio System. The latter seem to have hit the nail on the head when it comes to improving user experience on a range of Macs.

First of all, *DP5* just seems to run better. On my dual 2.0GHz G5 I instantly noticed a lower processor hit when *DP* was hosting various virtual instruments. And on my 1GHz G4 Powerbook the spectre of the spiking CPU meter seems to have all but vanished. It now seems to be possible to push the processor much harder without running into spikes, with *DP*'s graphic interface remaining comparatively smooth and responsive.

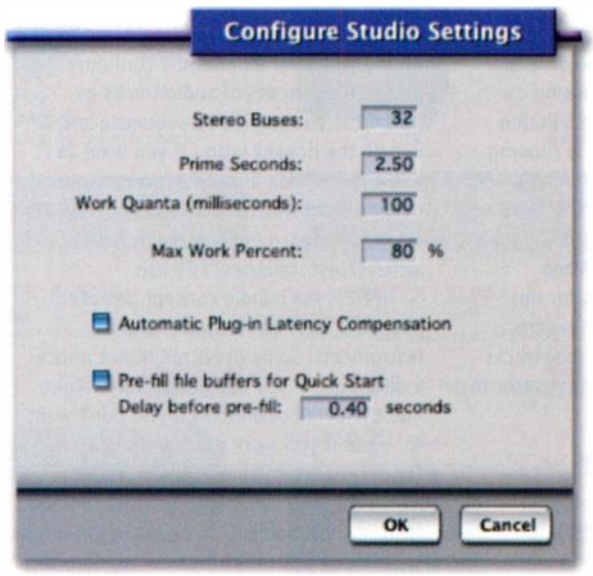
This in itself is a great development, but there's more, especially for those who like to get 'under the hood'. Go Studio menu / Configure Audio System / Configure Studio Settings and you might get a shock. You can no longer set a Studio size, control how many audio voices *DP* has available, or set the rather arcane 'disk read/write size' or 'buffer size per voice'. Instead, *DP* itself takes care of all the audio-voice allocation stuff and the user-configurable aspects of MAS performance are now represented by two rather bizarrely named settings: 'Prime Seconds' and 'Work Quanta'.

Work Quanta is apparently the interval (in milliseconds) at which the MAS engine does its calculations. Work Quanta is not Buffer Size — that's still there in the Configure Hardware Driver dialogue box —

and MOTU suggest it should normally be left at its default value of 100, but they also say that increasing it favours audio performance at the expense of graphical interface smoothness. At its maximum value of 500, I fancied there was a slight reduction in processor hit for an instrument-heavy sequence I was working on, so it may be worth experimenting with on your own Mac.

Prime Seconds is more mysterious. This setting seems to control how far ahead *DP* looks to 'pre-cue' the audio in your tracks when you locate the playback wiper in a sequence. Higher values potentially help with processor spikes that occur shortly after playback begins. But Prime Seconds also relates to something else new, and more puzzling still: pre-rendering.

Specific details on this new feature are tantalisingly scant, but it seems to work like this: when you place a plug-in on an audio track, dial in some settings and then close the plug-in's window, *DP5* 'pre-renders' audio in the track just ahead of the playback wiper, to sound exactly as though it was being treated in real time by the plug-in. The amount that is pre-rendered would seem to be equal to the



Perhaps pointing to some reworking of *DP*'s audio system, MAS, the Configure Studio Settings dialogue box in *DP5* sports some interesting new settings.

Prime Seconds value, and is heard as soon as you start sequence playback.

What is absolutely not happening is any sort of automated Freeze of entire tracks — pre-rendering only works on a temporary basis and even then only for a couple of seconds worth of audio. But presumably, by whenever possible cueing up this short amount of audio, complete with plug-in effects, *DP* is able to reduce the processor usage associated with playback (and particularly with starting playback).

Pre-rendering is permanently 'on' in *DP5*, but the only time you might be made aware of its presence is if you hear a burst of dry audio as you open and close plug-in windows — apparently, an unavoidable side-effect. It only works on audio tracks, not Aux tracks or Instrument tracks, and even then not for mono-to-stereo, mono-to-surround or stereo-to-surround effects. If, for any reason, you don't want a plug-in to use it, you can check a new 'Always Run in Real

▶ outputs. Take, for example, MOTU's *Model 12* drum module that ships with *DP5*. By default, all its drum voices are routed to its Main output — the output pair that is set for the Instrument track. But individual drum voices in *Model 12* can be routed to one of six additional Aux outputs instead of the Main output, via their audio-output menus. Voices routed like this are removed from the Main mix and can be dealt with separately, allowing you to use entirely different plug-in effects on them, for example. In *DP5* this has become laughably easy to set up, because as soon as you instantiate *Model 12* its Aux outputs (and its two Sends) instantly 'publish' themselves to the mixing environment. They become available as inputs in audio tracks, via the 'New Stereo [or Mono] Bundle' option in input pop-up menus.

To see how it works in practice, try the following. You're about to use *DP5*'s new capabilities to treat a *Model 12* snare sound with a reverb, while keeping the rest of the kit completely dry.

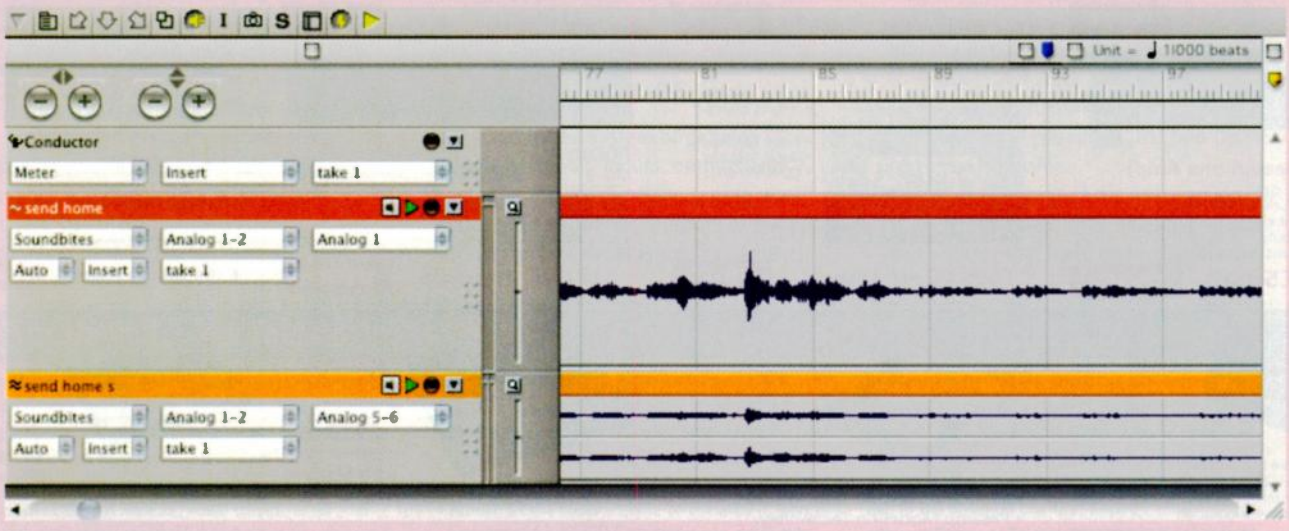
1. Instantiate *Model 12* on an Instrument track and make sure you give the track a valid output pair.
2. Load up a standard kit, such as 'Acoustic Kit 1', from the mini-menu in the title-bar of *Model 12*'s window. Then, for the third drum voice from the left (titled 'Snare Cent...'), click where its information pane says 'MAIN' and choose 'AUX1' instead.
3. In the Project menu, choose Aux Track from the Add Track sub-menu. Switch to the Mixing Board (Shift-M) or Tracks window (Shift-T) and locate this new Aux Track you've just created. Configure it with a valid audio output and then for its input choose New Stereo Bundle and scroll down the (potentially long) list of inputs until you find '*Model 12*-1 7-8'. This should be amongst a group of *Model 12* inputs running from '*Model 12*-1 3-4' to '*Model 12*-1 17-18'.
4. On your new Aux Track, instantiate your reverb plug-in of choice and set it up as you wish.
5. Play *Model 12* and you should find that all the drums except your snare remain dry, reaching *DP*'s mixing environment via *Model 12*'s main outs. But your snare is handled by the Aux Track, having been routed using an Instrument Bundle.

In the example above, in step three you chose '*Model 12*-1 7-8'? But why? What do the numbers mean? First of all, '*Model 12*' is obviously the name of the virtual instrument you're setting up. The '-1' bit indicates that you're working with the first *Model 12* instantiated in your sequence. Instantiate another one, on another Instrument track, and you'd be able to choose '*Model 12*-2' if you wanted to. And the '7-8'? To understand this, you need to think about how many outputs a *Model 12* has in total. There's a main stereo out (two channels), six stereo aux outs (12 channels) and two stereo sends (four channels). That's 18 outputs in total. The numbering goes like this:

Model 12 Output	Numbering In Input Pop-ups
Main Out	Doesn't appear as an input!
Send 1	3-4
Send 2	5-6
Aux 1	7-8
Aux 2	9-10
Aux 3	11-12
Aux 4	13-14
Aux 5	15-16
Aux 6	17-18

It's a shame that users have to deal with a numbering scheme like this, rather than choosing from a more descriptive list of names, but it's not that hard to get to grips with. Furthermore, it provides a flexible scheme for dealing with any virtual instrument — in MAS or Audio Unit format — that has multiple outputs. I've tested all the ones I own and it works perfectly. SOS

Digital Performer 5 Tips



Some graphical elements in *DP5*, such as pop-up menus, have been tidied up to give a rather smart new appearance. In the Sequence Editor, the information panes at the left of each track lane are now resizeable, so you can choose how many pop-up menus are displayed — particularly useful when a track's vertical size is very small.

If you've just upgraded to *DP5*, don't forget a great new feature that can easily be overlooked. The Meter Bridge, displayed by clicking its button in the central pane of the Consolidated Window, or by hitting Shift-Z, is almost laughably easy to use. There are buttons for displaying meters for hardware Inputs and Outputs, *DP*'s Busses, pre-configured audio Bundles, the outputs of multi-output virtual Instruments and Rewire channels in use and, last but not least, audio tracks themselves. The Meter Bridge resizes to fit as best it can into its window, and buttons at the top left allow you to choose a linear left-to-right or vertically stacked layout. The width of meters can be adjusted by clicking and dragging on the magnifying glass button, and their range adjusted by dragging either end of the vertical 'Scale' bar.

Also check out the new information-pane resize handle in the Sequence Editor. Just to the right of a track's settings pop-up menus is a little group of six dots. Click and drag here to resize the information pane and the pop-up menus rearrange themselves to fit. With this enhancement, you can have full access to all of a track's settings in the Sequence Editor, via dedicated pop-up menus, even when the track has been vertically resized to be quite small.

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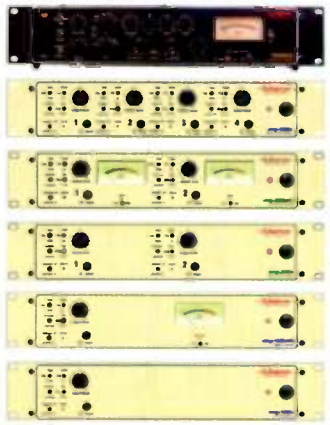

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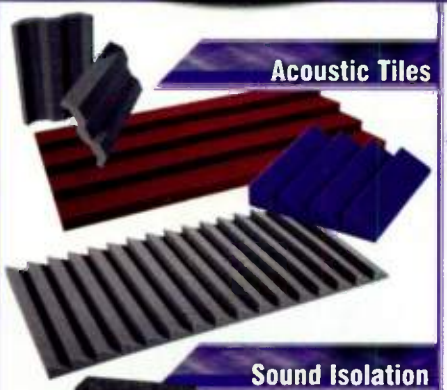

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Editing Chords & Signatures In Logic With Global Tracks

The Global Tracks can not only display chords, key/time signatures, and transpose values — they also let you edit them easily.

Stephen Bennett

In keeping with *Logic*'s other Global Tracks, the ones dedicated to time signatures and pitch manipulation generally clarify existing *Logic* functions and make them available in an way that makes them easier to use than they were in earlier versions. As is usual in *Logic*, the Global Tracks affect either the display of, or the actual data in, other areas of *Logic*. The Global Tracks covered in this month's *Logic* article deal with the display and manipulation of transposition (for you budding rock-ballad composers!), chords, time signatures, and key signatures.

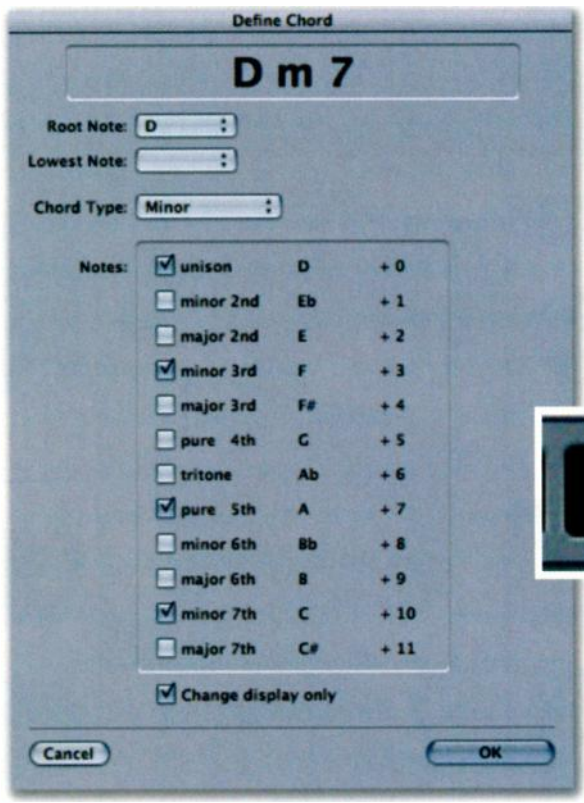
The Chord Global Track

You may have noticed that when you play a chord on a MIDI keyboard, its name is displayed in the MIDI input/output section of the Transport window — really useful if you want to pass on chordal information to other musicians. If you've recorded a whole sequence of chords, *Logic* can analyse and display chordal information in the Chord Global Track. To see this in action, play in some chords, make sure the sequence is selected, and then open the Chord Global Track from the View menu. If you then click on Analyse, *Logic* will display the detected chords on the Global Track. The Change Display Only tickbox will be ticked automatically to prevent your sequences from accidentally being transposed.

You can edit the chord display by double-clicking on its name (if the analysis algorithm has got it wrong) or add chords using the Pencil tool (if it has missed any out). You can also drag and delete chords from the Global Track display using the mouse, Eraser tool, and delete key. But this feature isn't just there to help you pass on your parts to other musicians. The Chord Global Track is closely linked to the Transposition Global Track, and any changes made here can have an affect on the overall pitch of MIDI sequences and Apple Loops.

The Transposition Global Track

If you open up the Transposition Global Track alongside the Chord Global Track, you'll see the effect the different Chords have on the 'root' key of the recording. Just playing back MIDI parts with Change Display ticked will not affect their pitch. However, if you untick it MIDI parts will be transposed accordingly — you can see the amount of pitch change displayed on the Transpose Global Track. Audio-based Apple Loops are always affected by the chord's affect on transposition, even if you have the Change Display Only parameter ticked, so it can cause a bit of confusion when combining the two. Using the Chord Global Track, you can thus easily insert key



Incoming MIDI data is analysed and its chord type automatically displayed in the Transport window. The Chord Global Track can extract chord information from any selected MIDI sequence in much the same way. If either analysis proves faulty, you can adjust how the detection algorithm responds using the Define Chord window.

changes into your Song using basic musical rules, if you like to work this way.

Audio files recorded or imported into *Logic* aren't affected by these transpositions, so if you want to keep everything transposing correctly, you'll need to convert your recording into an Apple Loop first. First select the sequence containing the recorded audio and choose Open in the Apple Loops Utility submenu of the Arrange window's Audio menu — the *Apple Loops Utility* will start to load and you'll be asked to enter the number of bars you want the Apple Loop to be. It's always best to choose an integer number of bars when you're creating an Apple Loop. In this case, the guitar part is eight bars long. The utility will open and you can choose the key of the recording and some search tags if you want to make the loop easily searchable from the Apple Loop browser. You may also want to add some transient information too if it's a rhythmic recording.

When you're done, click on Save and exit the *Apple Loops Utility*. Now the guitar recording will follow chord and transposition changes just like any Apple Loop. Note that only Apple Loops which have a Key Definition will be transposed — some, such as the Apple-supplied drum loops, will not. You can use this feature to make sure only the Apple Loops you actually want to change key will be transposed. If you like, removing the Key Definition of any Apple Loop will make your drums to play two octaves lower, trip-hop style.

Of course, you can also edit key changes directly on the Transposition Global Track with the mouse. You can add transposition nodes using the Pencil tool, delete them using the Eraser Tool, and drag them around with the mouse and Arrow tool. If you click on a transposition node while holding down the Control, Alt, and Apple keys, a small text box will open where you can enter the transposition value directly for accurate Whitney Houston impersonations. Setting the transposition value to zero resets the playback to the original pitch. You may notice that changing the value of a node on the Transposition Global Track produces a corresponding change on the Chord Global Track, and vice versa. The 'root' key or zero value of the Transposition Global Track is defined in the Signature Global Track, and the transposition values show the difference between the first key signature value displayed there and the root note of the chord. So if the Key signature is set to 'C', an 'E' chord will produce a transpose value of +4 or -8 (depending if it's transposed up or down), a 'G' will display +7 or -5, and so on.

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

A Shift Too Far

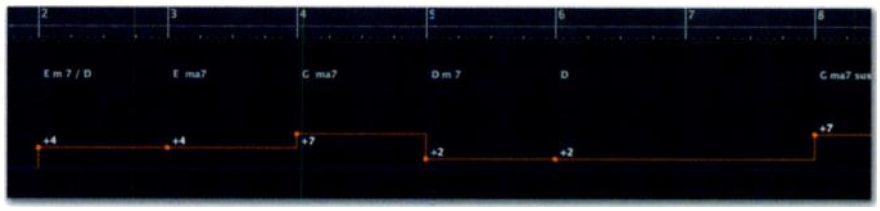
When using Global Track components that deal with tempo or pitch changes of audio, the key to getting usable results is to keep it simple, because *Logic*'s pitch- and tempo-matching facilities are not that brilliant (at least up to version 7.2), and you'll only get decent results if you go for minimal changes. Try pitch-changes of a few semitones only, and choose the most suitable algorithm from the Time Machine submenu in the Arrange window's Audio menu — you'll have to use your ears to determine which produces the best results. If you want to go for more exotic pitch changing, you may be better off looking at using some third-party software alongside *Logic*, such as Celemony *Melodyne* (which integrates with *Logic* using Rewire) or Serato's high-quality *Pitch 'n Time* plug-in. Although the latter is an AU plug-in, it appears in *Logic* v7.2 or greater as an extra choice in the Time Machine submenu menu. Don't overlook Apple's Tiger-based *AUPitch* Audio Unit either, as it's capable of some pretty high-quality transpositions. It's exactly the same when dealing with tempo changes too. Try not to apply overly radical tempo changes to audio files, and keep sections small and fixed to the nearest whole beat. *Logic* can't perform miracles, so sometimes the only solution is to play it again, man...

The Signature Global Track

We've seen that the Signature Global Track is where key signatures can be displayed, added, and edited, but it's also the home of time signatures too. A lot of music these days is resolutely in 4/4 or one of its siblings — albeit with the occasional excursion into triplet territory. But if you're more musically ambitious, or into progressive rock or jazz, you may want to experiment with some more esoteric time signatures. If you've been listening to early Dave Stewart recordings, you may even want to try using multiple combinations of exotic time signatures in the first four bars of a song!

If you want to record alongside a click in 7/8 or 11/8, you'll need to set these values at the correct Song Position Line (SPL) position. You can do this by typing time signatures directly into the Transport bar, which will make sure that you'll get a click that will be in the correct time signature for you to play along with. To do this, just move the SPL to the required bar, double-click on the time-signature field in the Transport window and enter the required values. If you do this, you'll see that these values are also displayed in the Signature Global Track. However, if you want to add overdubs to previously recorded audio, or set up a recording session with multiple time-signature changes for you to play along with, you may want to add these directly into their required timeline positions on the Global Track itself using the Pencil tool. Clicking on the Global Track opens a box where you can enter the required values.

It's important to understand that time-signature changes made on the Signature Global Track do not directly affect the playback of any MIDI or audio recordings or Apple Loops — only the click-track playback will be affected. However if you open the Score window you'll see the time signature displayed there too, and you can also add and edit time signatures directly here alongside any key signatures from the Signature Global Track. The Signature Global Track is intimately linked with the Score window, and both will therefore find most use by those who like to edit and print out their dots. SOS



When the Chord and Transposition Global Tracks are linked, Apple Loops and MIDI sequences can be made to follow changes in the chord progression.

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.

Virtual Instruments

In Ableton *Live*

When Ableton added MIDI support to *Live*, they also added virtual instrument support — this month we look at how to take full advantage of software synths within *Live*.

The MIDI track contains a *Scale* MIDI effects plug-in, an instance of Cakewalk's *Rapture*, and *Live*'s *Saturator* audio plug-in at the output. The MIDI track's Monitor is set to Auto, so the track needs to be record-enabled in order to pass the incoming MIDI data through to the instrument.

Craig Anderton

One of *Live*'s more appealing aspects is how it has managed to add increased functionality while retaining the simple, yet clever, operational paradigm that was its initial source of appeal. It's understandable that when *Live 4* added MIDI, the '*Live* faithful' were concerned the program would veer precariously in the direction of bloat. Instead, Ableton crafted a seamless MIDI implementation that added another dimension to the program, including virtual instruments.

However, *Live's* virtual instrument implementation requires an understanding of how signals are routed within Live, as well as the differences (and similarities) between MIDI, audio, and instrument tracks. We'll cover these topics, and more, in this issue's column.

Let's start by covering the simplest way to use a virtual instrument, then build on that foundation.

Simply Soft Synths

A virtual instrument requires a MIDI track, but once inserted, the track becomes a hybrid MIDI/audio track. Here's how the process works.

Begin by dragging an instrument from the *Live* Browser into the Mixer Drop Area. This automatically creates a MIDI track, inserts the instrument into the MIDI Track View, and opens up the instrument GUI for editing. An alternate method is to create a MIDI track by going Insert > Insert MIDI Track, then dragging a VST instrument from the browser into the track's MIDI Track View.

In the track's upper MIDI From field, select your MIDI controller (which should be enabled in the Preferences menu) and select the controller channel you want to monitor in the lower MIDI From field. Typically this will be All Channels, unless you need the instrument to respond to a specific channel; in this case, choose a specific channel.

Set Monitor to Auto if the track is record-enabled. If it isn't, set monitor to 'In'. 'In' will also work in Auto mode, but if set to 'In', remember to return the Monitor function to Auto when you want to play back the track.

With an instrument inserted, the MIDI track accepts MIDI at its input, but outputs audio from the soft synth's audio output. As a result, you'll need to specify Audio To; usually this will be the Master Out. However, as we're now dealing with an audio track, the Master Out could also feed the Sends Only, or the input of another audio track.

Finally, play a few notes on your controller, and you should hear the soft synth play. If not, check the MIDI setup and the track input and output assignments.

More To The Track Than Meets The Ear

You're not limited to dragging an instrument into a MIDI instrument track. You can drag MIDI effects into the MIDI Track View prior to the instrument (to the instrument's left), or audio effects after (to the right of) the instrument.

Also note that the configuration of the track itself changes automatically, depending on whether you're recording MIDI data or playing back the instrument sound. When the track is record-enabled, the meter shows incoming MIDI velocity. When it's set for playback, the meter changes into an audio output meter.

Simultaneous Audio Recording

It's possible to simultaneously record the instrument output into an audio track as you record the MIDI data. Sound useless? Well, remember that *Live* is about being able to make decisions and changes on the fly, so consider this: You're using *Live* in a live performance, and building up loops over which you can play. You're playing a soft synth via MIDI, and then plan to switch over to another sound and play that. Of course, you can just record the sound of the soft synth into an audio track, and be done with it. But if you record into MIDI and audio simultaneously, you have the option of

The MIDI track has a USB keyboard chosen as its input, and is sending the output to audio track one. The audio track is receiving its input from MIDI track two, and sending its audio to the Master out. As both tracks are recording simultaneously (Ctrl-click to enable multiple record buttons), both are record-enabled and Monitor is set to Auto.

going with the audio track 'as is', or tweaking the MIDI track if there are problems, then recording this data through the instrument, which converts it into audio for the audio track. Once recorded as audio, you can move on to recording the next MIDI part.

Doing this takes advantage of *Live*'s flexible routing, which you can consider as a sort of 'matrix modulation without the matrix' as you can choose an input from any of several sources, and send an output to any of several destinations.

It's easy to set up *Live* for this scenario: first set the input of the MIDI instrument track to receive input from your MIDI source, record-enable the MIDI track, and set monitor to Auto. Send the MIDI track output to an audio track and set the audio track's input to monitor the MIDI instrument track. Record-enable the audio track by control-clicking on its Arm Session recording button, and set monitor to Auto. Now when you record in *Live*, you'll be recording MIDI data into the MIDI track and audio data into the audio track.

There are a few fine points. Because the audio track is monitoring the MIDI track output, it doesn't really matter where you send the MIDI track's out — it could go to the sends, or even an external out, and you'll still hear the MIDI instrument through the audio track. Similarly, if you're sending the MIDI instrument track out to the audio track input, then the audio track input can be set to any input because it's always going to receive input from the MIDI instrument track.

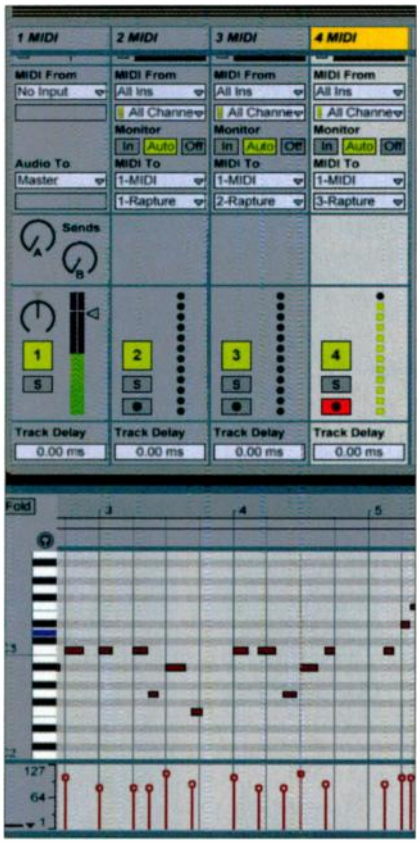
Let's Get Multitimbral

In case you're wondering why I used Cakewalk's *Rapture* in the first example, it's because it has the '*Live* mentality'. *Rapture* has six different 'elements' (basically, voices) that offer lots of looping and tempo-sync options, and can be set up for multitimbral operation so that each element receives data over a separate MIDI channel. When loaded into *Live*, this allows a variety of MIDI-based looping options that complement the way *Live* treats digital audio.

Anyway, taking advantage of this requires setting up *Live* for a multitimbral synth. In this instance, the 'MIDI + instrument' track becomes a kind of container for the soft synth, and is fed by multiple MIDI tracks (one for each channel you want to address). As an example, we'll set up *Live* to play back three MIDI tracks into *Rapture* elements one, two, and three, which are set to those respective channels.

Create the MIDI instrument track, which we'll call track one. I generally set this to No Input to avoid confusion, as I want to add inputs only from other MIDI tracks and create as many MIDI tracks as there are multitimbral channels to be driven. In this case, there are three. Set the input for each of these MIDI tracks as desired.

In each MIDI track's upper MIDI To field, select the track



Data is being recorded via a MIDI track (four), and sent to channel three of the MIDI track (one) containing the instrument.

▶ containing the instrument (in this case, MIDI track one).

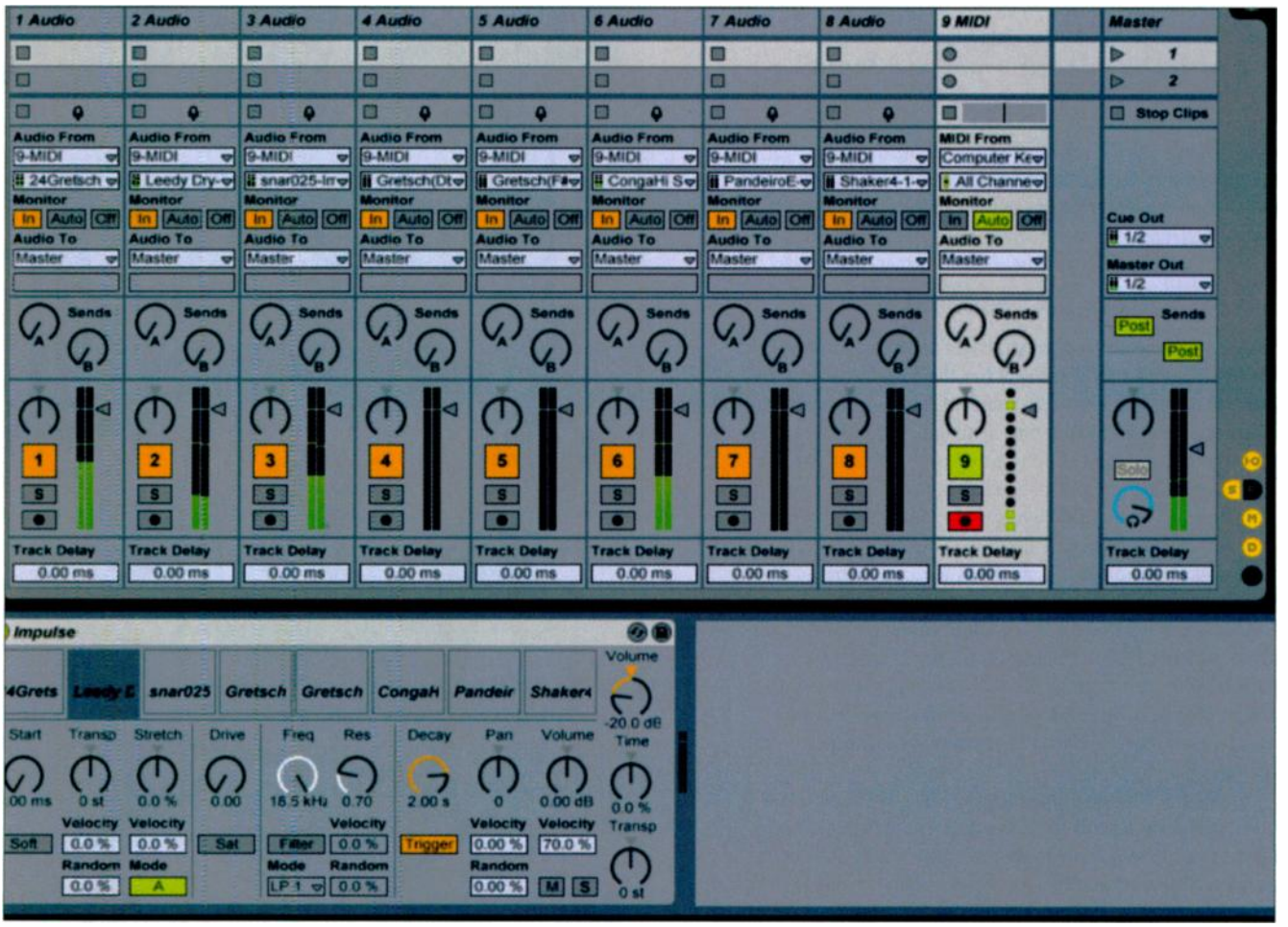
In the lower MIDI To field, assign the MIDI output to the desired instrument channel (in this case channels one, two, and three for the three element channels). Now set the Monitor switches for the three MIDI tracks to Auto. If you record-enable a track, it will send the incoming MIDI data to the channel that you have just selected, and of course, you'll also be able to record this into the sequencer.

Now you can record as desired into the various tracks, and play them all back when completed into the MIDI Instrument track.

What About Multiple Outputs?

Live can also accommodate instruments with multiple outputs. We'll use the *Impulse* drum machine as an example, as it comes with *Live* and is a cool little instrument anyway. In this scenario, we'll play the drum sounds from a MIDI controller, and send each *Impulse* sound to its own output for individual processing. Here are the steps required to accomplish this.

- Create a MIDI track.
- Set the MIDI From field to whatever will receive your controller's MIDI output (such as All Ins).
- Decide how many individual outputs you want to use, and create an equivalent number of audio tracks. For example, if you want to assign each of the eight *Impulse* sounds to an individual output (and you can indeed do this), you would create eight audio tracks dedicated to *Impulse*.
- Set each audio track's upper Audio From field to the MIDI track.
- Each audio track's lower Audio From field provides a pop-up menu showing *Impulse*'s individual outputs. Choose the desired output for each track.
- In terms of monitoring, you have the usual two options: Record-enable a track and set Monitor to Auto, or simply have the track monitor the input ('In'). I generally prefer not to record all the outputs as Audio, but simply monitor the 'In' while recording so I can hear the results of playing the MIDI part. After the part is recorded, I switch



Impulse, the drum module included with *Live*, provides up to eight individual audio outputs (one for each drum sound). This screenshot shows channel nine feeding *Impulse* MIDI data, and each sound appearing over its own audio track.

Monitor to Auto.

Of course, when using instruments which have multiple outputs, you need to make sure that you've assigned sounds properly to output channels within the instrument itself.

Creating Splits & Layers

A basic MIDI function is creating a split, so that a master controller can be divided into regions, with each region triggering a different sound. The stereotyped application is a bass sound played by the left hand and a lead sound with the right hand, but splits are good for special effects as well — high 'C' might trigger a special sample, for example.

Thanks to MIDI effects, *Live* allows setting up as many splits as you like. We'll use the example of splitting a keyboard so that playing C2 or below plays through one instrument, while playing C#2 or above plays through a different instrument.

Set up two MIDI tracks with the instruments that will play the two different sounds. Set each of their MIDI From fields to respond to your MIDI controller, or to the same MIDI data track.

The *Pitch* MIDI plug-in can be set up to create keyboard splits.

Drag the *Pitch* MIDI effect from the Browser into the instrument that's intended to play C2 and below.

The *Pitch* effect passes notes in a certain range while rejecting all others. To edit it the way we want, set the Lowest parameter (the lowest note of the range) to C-2. Set Range to 48st, which means that the range will extend 48 semitones above C-2. This places the range's upper note at C2. Do not adjust the Pitch parameter itself — leave it at zero.

As we did before, drag the *Pitch* MIDI effect from the Browser into the instrument that's intended to play notes C#2 and above. Set the Lowest parameter to 'C#2'. For Range, as we want to play anything above C#2, we can just choose the maximum range of 127st. As before, leave the Pitch parameter at zero.

And now we have a keyboard split. Note that the *Pitch* plug-in has a small light below the Pitch control that glows yellow when the plug-in is receiving a note outside of its specified range. This is very thoughtful, because if you don't hear anything, you can check here first and make sure the problem isn't just that you're playing out of the specified range.

Layering two (or more) synths so that they play together is easy. All you need do is to have the two MIDI instrument tracks monitor the same MIDI data track at their inputs. And as expected, you can combine splits and layers by simply creating an overlapping range with the *Pitch* plug-in. For example, you could place the highest note of the lower split above the lowest note of the upper split, thus creating a range where the two sounds overlap.

If you're still using *Live* strictly as a digital audio-oriented machine, that's all well and good — but you're missing out on a lot. Start by exploring the *Impulse* and *Simpler* instruments included with *Live*; you might be surprised at how getting MIDI instruments involved can expand your musical horizons. SOS

In these two screens you can see a second pass of automation data being recorded on top of a previous pass where the level is higher during the second pass than the first. In the first screen the Automation Return Time (ART) parameter is set to 100ms, while in the second screen ART is set to 1000ms: notice the how this affects how *Cubase* joins the point between the two automation passes.

Using Automation

In *Cubase SX*

Mark Wherry

This month we continue to explore *Cubase*'s automation features with a look at the different modes available to *SX* users, and the issues you'll face when using automation and MIDI Controller data on MIDI Tracks.

In last month's *Cubase* workshop we looked at how to write automation data on a track. By default, *Cubase* writes automation data in a mode that's known as 'Touch Fader' in *Cubase* speak. This basically means that *Cubase* will write automation data (when in write mode) at any time you're touching the fader, even if you aren't actually moving the fader, overwriting any pre-existing automation data for the parameter being 'touched'. Obviously it's not actually possible to 'touch' an onscreen control, so if you're using a mouse, 'touch' means to click and hold an onscreen control without releasing the mouse button. However, if you're using a hardware control surface with *Cubase* that offers touch-sensitive controls, 'touch' literally means touch.

The ARTful Dodger

In *Cubase SX* and *SL*, Touch Fader automation is used in conjunction with the Automation Return Time setting, which is part of the Automation Mode toolbar on the Project window. If you can't see this setting, right-click on the Project window's toolbar and make sure Automation Mode is checked.

The Automation Return Time (ART) is the time it takes from when you stop overwriting automation data for the value of the parameter to return to the value of the next Automation Event of that same parameter, and can be between one and 2000ms.

For example, say you have automation data written for volume that keeps the level roughly at 0dB and you overwrite this with a fade-out in one part of the track. If the ART setting is 1000ms and you release the fader at around -64dB, *Cubase* will automatically write automation data so that the channel smoothly returns to the level of the pre-existing automation data (from -64 to 0dB) over a period of a second. This prevents the volume suddenly jumping from -64dB to 0dB, which would happen if there was no ART feature, or if the ART was very short.

In *Cubase SL* (and *SE*), Touch Fader is the only way in which automation can be written, but *Cubase SX* users have four other modes available to them: Autolatch, X-Over, Overwrite, and Trim.

The Automation Mode pop-up menu in *Cubase SX* enables you to select different modes to use when writing automation data.

- In Autolatch mode, as with Touch Fader, Automation Events are recorded (overwriting existing data) from the minute you 'touch' a parameter. However, in this mode Automation Events will continue to be recorded once you have released the parameter and until you stop the transport.
- X-Over is similar to Autolatch mode, except that once the parameter you're automating is released, *Cubase* will stop recording Automation Events as soon as existing Events are encountered on the track.
- Overwrite mode is again similar to Autolatch mode, with two exceptions: firstly, Overwrite mode only works with Volume Automation Events, and, secondly, Automation Events are recorded (overwriting existing data) the instant the transport is running, until you stop the transport.
- Finally, Trim mode, like Overwrite, also only works with Volume Events; but rather than writing completely new Automation Events it allows you to proportionally adjust existing volume automation data.

When you start the transport running in Trim mode, the volume fader moves to a central position where moving the fader up increases the level of all Volume Events relative to each other before the Project Cursor, and moving the fader down decreases the level of all Volume Events ahead of the Cursor.

Usefully, you can also use the Trim mode offline as well. With Write Automation enabled on a given track, selecting Trim mode and moving the volume fader for that track will trim all the Volume Automation Events on that track between the Left and Right Locators. The last part of that sentence is the most crucial, as it's easy to forget to set the Locators appropriately, although it is really neat to be able to trim certain sections of a track by setting the Locators to specific locations. To ensure you're trimming all the Automation Events on the track, right/Control-click in the Volume Automation track and choose 'Select All Events', followed by 'Transport / Locators to Selection', which will set the Locators to encompass all of the Volume Automation Events on the track.

MIDI Automation

In the examples discussed in last month's *Cubase* workshop, we were looking at automating parameters on audio-based tracks; but it's also possible to read, write and edit automation data for parameters on MIDI-based tracks in exactly the same way, via a track's [R] Read and [W] Write buttons, along with the Automation sub-tracks. The parameters that can be automated for MIDI-based tracks include Volume, Pan, Mute, the Send and Insert Enables (along with the controls for any MIDI Insert plug-ins you might be using), and most of the controls from the Track Parameters Section (including a global on/off toggle), such as Transpose, Velocity Shift, and the Random and Range settings.

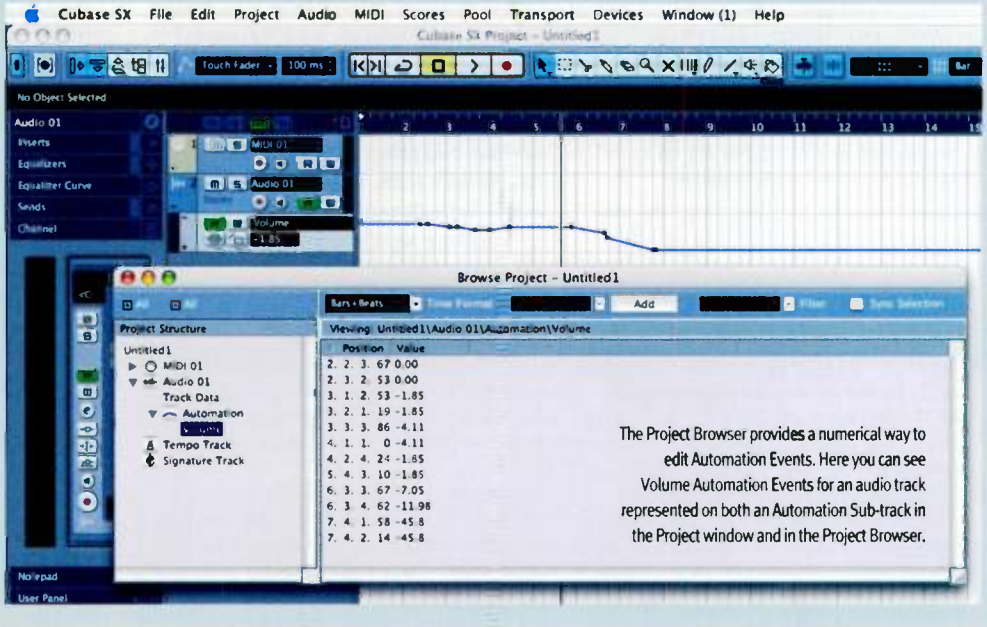
The Add Parameter window for a MIDI track shows which parameters can be automated. Parameters for MIDI Insert plug-ins will also appear here for any Insert plug-ins you're using on a MIDI track.

Being able to automate some of these parameters might not seem especially useful at a first glance, but being able to automate pitch- and velocity-modulating parameters such as Transpose and Velocity Shift can lead to some interesting creative ideas. Back in the *Cubase VST* days, for example, I was once asked to hack together a makeshift transpose fader to add variation to some taiko samples. Because the taiko samples were laid out on the keyboard with a different (but similar drum) sound assigned to each pitch, you could take some very basic rhythmic patterns and move the transpose fader to create more complex, varying patterns. This would work especially well with fast, straight patterns, and is effective when there are random notes on the keyboard that have no sample assigned, so as to create gaps. Creating these types of effects is now much easier in *Cubase SX* with the ability to automate a track's Transpose parameter.

Standard MIDI Controllers can also be added as Automation Parameters, but this where things can get a little complicated because working with Controller data as Automation Events is completely separate to working with Controller data as MIDI Events in one of the editor windows, such as the ▶

Editing Automation Events In The Project Browser

In addition to using Automation tracks (and sub-tracks) to edit Automation Events graphically, it's also possible to edit this information numerically in *Cubase*'s Project Browser window, which you can open by choosing Project / Browser or pressing Control/Command-B. In the Project Structure list, select the track containing automation data you want to edit and click the triangle beside it to reveal an Automation sub-folder, which has an entry for each of the Automation tracks (or sub-tracks) used for that track. Selecting the appropriate Automation track in the Project Structure list will display all the Automation Events for that track in the main Event Display area, where you can edit the Position or Value of Automation Events.



The Project Browser provides a numerical way to edit Automation Events. Here you can see Volume Automation Events for an audio track represented on both an Automation Sub-track in the Project window and in the Project Browser.

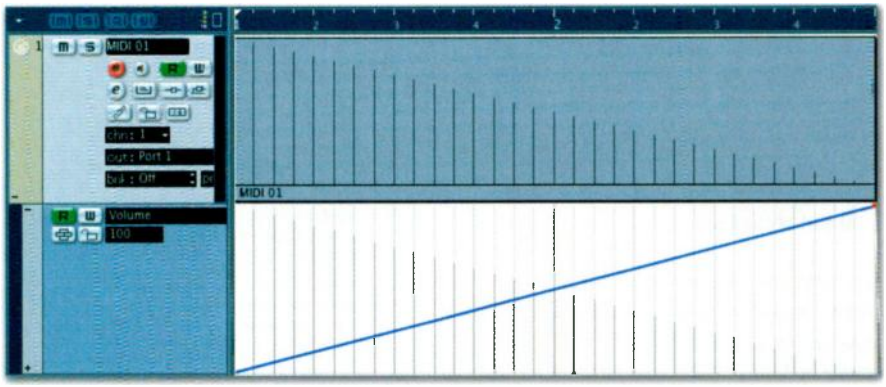
Controller Lanes in the Key editor. This is a problem because both types of data can co-exist and produce conflicting results that are confusing for the user and for *Cubase*.

If you're not completely sure what I'm talking about, let's consider the Volume parameter as an example: when you move the volume fader on a MIDI track in *Cubase*, the program will send MIDI Controller 7 (Volume) data to the appropriate MIDI device. To save the conflict of two automation parameters producing the same data, you'll notice that Controller 7 isn't available from the Add Parameters window, since it's effectively the same as the track's Volume parameter. But what if you have Controller 7 data programmed in your MIDI Parts and you have recorded volume automation data on the MIDI Track as well?

MIDI Controllers Vs. Automation

To take a worst-case scenario, imagine you have MIDI Controller 7 data recorded that represents a fade-out over two-bars. What would happen if you had Track Volume Automation Events in the same two-bar period that were performing a fade-in? On playback, the MIDI output of the track would contain both the MIDI Controller 7 fade-out data, and the Automation Events, which *Cubase* would also translate into MIDI Controller 7 data. Because we can have two sets of volume data represented in two completely different ways, that end up sending the same type of message to our MIDI device (be it an external or VST-based instrument), the result is completely garbled.

One easy solution is to only use one area of *Cubase* to work with MIDI Controller data: either work with this data as Automation Events or MIDI Events. However, the problem here is that if you choose Automation Events, you can only work with the data on the Project window, and if you choose MIDI Events you can work in one of the MIDI editor windows. Since both sets of tools have advantages and disadvantages, it's no wonder that users naturally gravitate towards mixing and matching to get the best of both worlds, which leads to the type of conflicts described earlier.



Here you can see MIDI Controller 7 (Volume) data represented by the vertical lines in the MIDI Part, with Volume Automation Events recorded over the same period of time. When played back, both sets of data send MIDI Controller 7 to the MIDI device, which produces a rather unintelligible result.

In terms of volume, one good approach is to think about how, in a musical sense, track-based automation is used compared to how MIDI Controllers are used. Track-based automation is intended, at the most basic level, to record the movements of *Cubase*'s internal mixer. While MIDI Controllers can and have been used to remotely control mixers, they also have a second purpose, which is to provide articulation data for a MIDI instrument. By articulation data, I mean parameters such as modulation, or even volume in the dynamic sense, such as when you record MIDI Controllers to mimic the way someone would produce louder and softer tones while playing an instrument. In this case, we're talking about a performance and not a mix parameter.

The good news is that when the list of MIDI Controllers was drawn up, somebody was obviously thinking about this problem. One Controller often forgotten is Expression (despite the fact that most instruments, hard and soft, support it), which is basically a volume fader that sits before the regular MIDI Controller 7 volume fader. This means you can record your dynamic volume as Expression MIDI Controller data and leave *Cubase*'s track-based volume control as a trim for the output of that channel. You can have two different types of volume data playing back at the same time without interfering with each other.

The above workaround is obviously only suitable for avoiding volume conflicts, and right now there isn't a really good workaround for other types of MIDI Controller data. On the plus side, though, a command called 'Extract MIDI Automation' was added in *SX 3* that allows you to convert Controller data represented as MIDI Events in MIDI Parts into track-based Automation Events. To use this command, simply select the MIDI Part (or Parts) in question, choose MIDI / Functions / Extract MIDI Automation, and MIDI controller data will be converted into automation data.

Last But Not Least

A final Preference that's handy if you use a hardware MIDI controller for recording Controller data is 'MIDI Controller Input to Automation Tracks', which is found on the MIDI page in the Preferences window. When this is disabled, MIDI Controller data is recorded as normal (if you're in record mode), as MIDI Events in MIDI Parts; but when it's enabled, incoming Controller data remotely controls automation parameters on the selected MIDI track instead. In this mode, you can record Controller data and some track parameters (volume and pan) as Automation Events in the same way you would normally write automation data, but using your regular MIDI hardware controller. However, note that this only works with MIDI tracks (you can't write volume data for a selected audio channel in this way, unfortunately), as the purpose of the feature is to prevent you from recording Controller data as both MIDI and Automation Events.

Understanding automation can greatly improve the quality of mixes produced entirely in *Cubase*, and, once you've got the hang of it, you'll start to see why hardware control surfaces are so useful. Although they don't actually add anything sonically to your work, one area where they really are useful is writing automation data — especially if they have touch-sensitive controls. Stay tuned for more about this in a forthcoming *Cubase* workshop article. SOS

Multiple Track Trickery

When you have multiple adjacent tracks in the Track List that you need to rename (such as after adding several MIDI tracks via the 'Add Multiple Tracks' command), renaming each one individually can be a real pain. Fortunately, there's a quicker way of carrying out this task: double-click the name of the first track you want to change and type in the new name as normal, but instead of pressing Return (or clicking outside the text field) to exit, you can simply press Tab to jump to the next track's name in the List, or press Shift-Tab to move to the previous track's name.

When you have finished renaming tracks, press Return or click outside the text field to exit as normal. The track that was selected when you started renaming tracks will still be the track that's selected in the Track List.

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Region Looping In *Pro Tools 7*

When your project includes repeated MIDI or audio parts, the new Region Looping tools in *Pro Tools 7* enable you to work faster and with greater flexibility.

Mike Thornton

In this month's Pro Tools workshop, we are going to look at a new feature in *Pro Tools 7*: Region Looping. Whether I am working in 'post' mode or 'music' mode, I often need to fill a gap between two sections. In previous versions of *Pro Tools*, I would first create a Region to loop, and then use the Duplicate command from the Edit menu (Command / Ctrl+D) to add extra copies until the hole was filled. Region Looping takes this process and makes it so much easier, whilst enabling lots of additional options.

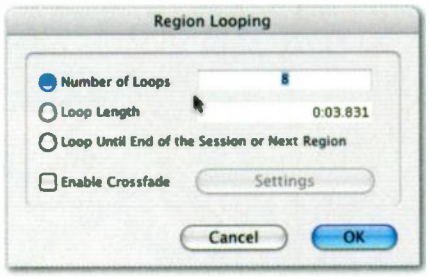
You can consider a Looped Region as consisting of a master loop with a number of 'aliases' that follow it. Nothing special about that, you might say — isn't that in effect what the Duplicate command does? Not quite. For one thing, the aliases created using Region Looping automatically change to reflect any modifications made to the master loop. Region Looping also brings in another new *Pro Tools 7* feature — Region Grouping — in such a way that the master and aliases work together as one region. Finally, automation attached to a Looped Region doesn't get repeated, as would happen with the Duplicate command.

Every instance of a Looped Region displays the Loop icon in the bottom right-hand corner so that you know it is not a simple duplicate of the original.

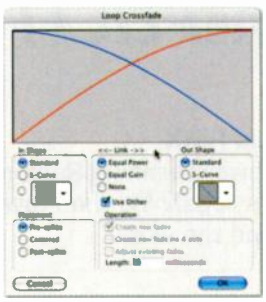
Looping The Loop

To create Looped Regions, first select an audio or MIDI Region, then choose the Loop option from the Region menu or press Command+Alt+L (Windows: Ctrl+Alt+L). The Region Looping dialogue box will open, with three options available.

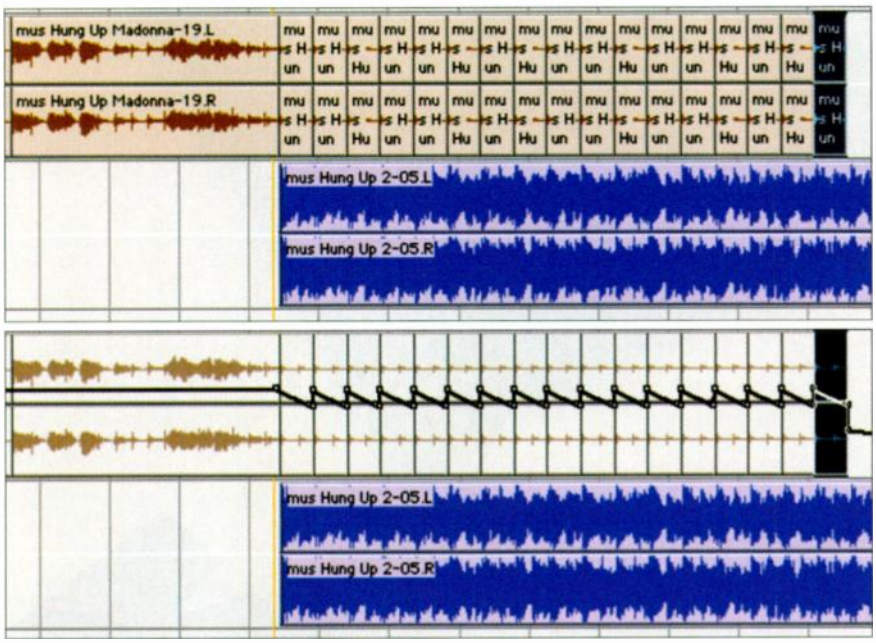
If you choose the Number of Loops option, you can simply enter the number of times you want the Region to be looped. The other options are designed to provide ways for *Pro Tools* to do the maths when it comes to working out how many times the Region needs to be looped to fit into your Session. Loop Length allows you to enter the duration required in the format of the main timebase ruler (in this screenshot, Minutes & Seconds). Note that if the duration is not an exact multiple of the loop length then the last alias will not be complete. Alternatively, the 'Loop Until End of Session or Next Region' option repeats the selected Region until another Region on that track is reached, or until the end of the Session if there are no further Regions on the track. Again, the last alias will be a fragment unless the gap is an exact multiple, in length, of the Region being looped.



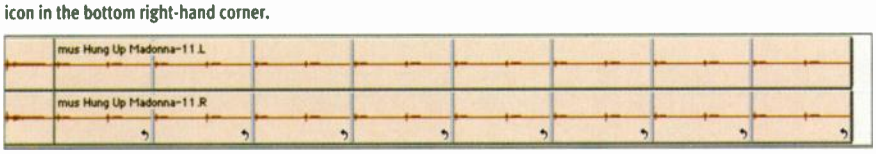
If desired, you can also select the Enable Crossfade option which will create a crossfade at each loop point. To do this click the Settings button and set the crossfades as required, before clicking the OK button to return to the Region Looping window. Finally, click the OK button in the Region Looping window and *Pro Tools* will create the appropriate loops.

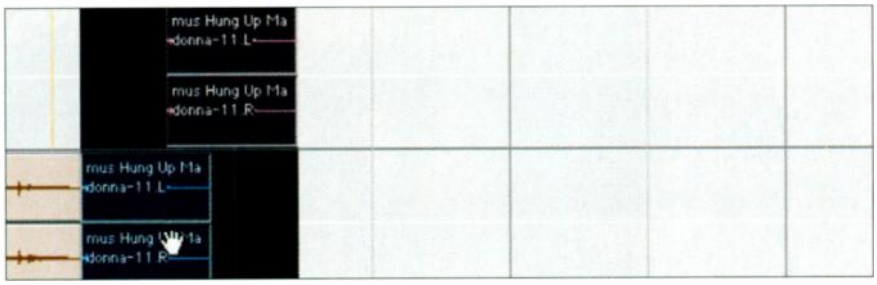


You can loop multiple Regions across several tracks: simply select all the Regions you want to loop and then run the Region Looping command as above. Region Looping also works with Region Groups as well, but if you select more than one ungrouped Region ▶

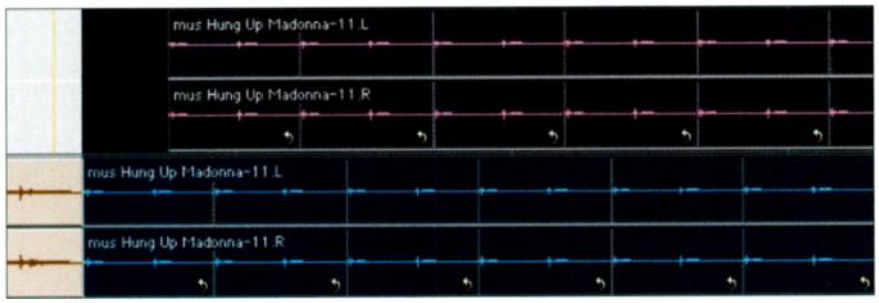


Using the Duplicate command to copy a Region creates lots of independent Regions, and copies any automation that was attached to the original (above). The new alternative in *Pro Tools 7* is Looped Regions (below), which display a loop icon in the bottom right-hand corner.





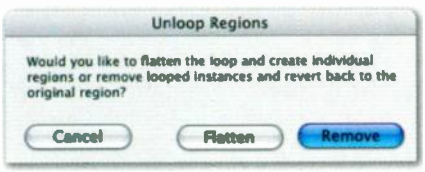
Multiple Regions on separate tracks can be looped.



▶ on the same track, *Pro Tools* will only loop the last region.

Unlooping The Loop

Just as it is possible to ungroup Region Groups, so it is possible to unloop Region Loops. Select your Looped Region and choose the Unloop option from the Region menu to see the Unloop Regions dialogue box, where you can choose from two options. Remove simply deletes all the aliases, leaving you with just the original 'master' loop. Flatten retains all of the aliases, but converts them to conventional Regions in their own right — in other words, what you would have had if you'd used the Duplicate command to create all those Regions in the first place. If you want

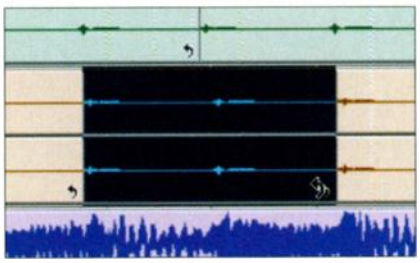


to unloop and ungroup a Looped Region Group, you can use the Ungroup All option in the Region menu.

Editing Region Loops

Looped Regions can be edited as a group or individually. Single-clicking with the Grabber or double-clicking with the Selector tool anywhere on the Region Loop will select the whole group. Now you can move the group to the desired location with the Grabber. To select an individual loop within the group, you can use the Smart tool: as you hover close to the loop icon you will see that the cursor changes to the loop icon, and a single-click will select that loop. If you are using the Selector tool, hover over the loop icon and wait for the cursor to change to a 'loop and selector' icon, then click and drag left until *Pro Tools* highlights the individual loop.

The normal tab feature where the Tab key on the keyboard moves the cursor to the next Region boundary to the right works differently in Region Loops. The conventional tab feature treats a Region Loop as one complete Region,



If you hover the Smart tool over its loop icon, you can select an individual Region within a loop.

so using the Tab key will make the cursor go the end of the Region Loop, missing out all the individual loops. However, when Tab to Transient is enabled, it will still function within a Region Loop.

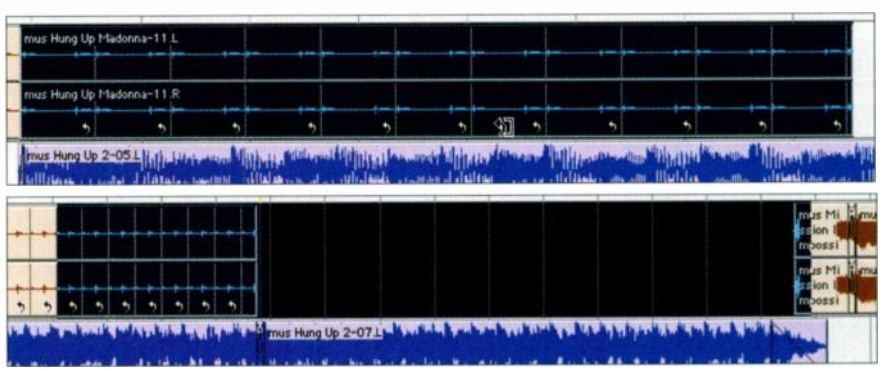
Using the Trim tool in the conventional way will trim the Region Loop as if it was one single Region, cutting the block of loops to the desired length and leaving a fragment of a loop at the end if needs be — holding down the Ctrl key (Mac) or the Start key (Windows) forces the Trim tool to trim only to the nearest Region boundary within the loop, so leaving no fragment at the end. Using the TCE Trimmer will produce a single consolidated Region rather than a master plus aliases.

Alternatively, you can trim every individual Looped Region within the group (it is not possible to change the length of just one of the Regions without flattening the loop first). Select the Trim tool (this doesn't work with the Smart tool enabled) and hover over the loop icon on one of the Regions. You will see that the cursor changes to an icon combining the loop and trim icons, and now you can adjust the length of the individual Region. Note too that the number of repetitions will automatically be increased or decreased such that the total length of the Region Loop stays the same.

You can also use the Trim to Fill feature to extend a Looped Region to fill an available 'hole'. Use the Selector tool to make a selection that includes some or all of the Looped Region, choose Trim Region from the Edit menu and in this case select End to Fill. *Pro Tools* will add loop aliases to fill the hole rather than the exact space you've highlighted. If the hole is before rather than after the Looped Region, *Pro Tools* will move the original loop back and put the aliases after it, so that the original 'master' Region is always the first loop in the Looped Region.

Automation And Region Loops

As I said earlier, automation doesn't get copied when you use the Region Loop command. If you want the automation to be repeated, you need to use the Copy Special and Paste Special commands from the Edit menu. Select the source Region, which will contain the automation, and use Copy Special from the Edit menu. Then select the Region Loop, ready to paste the automation, and use the Paste Special — Repeat to Fill Selection command. This will paste the automation over the complete Region Loop. SOS



Trimming an individual Region within the loop changes the length of all the Regions. *Pro Tools* will also add or remove additional Regions so that the loop is still the same length.

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apple notes

In just a few weeks, Apple have completed the transition to Intel processors for their entire Mac portable line, starting with a 17-inch Mac Book Pro and ending with the introduction of the Mac Book family of laptops to replace the iBook. Apple Notes assesses the new machines.

Mark Wherry

As Apple try to gain a larger share of the video post-production market, the NAB (National Association of Broadcasters) show, held annually in Las Vegas, has become increasingly important. This year's NAB show saw Apple announce the 17-inch Mac Book Pro, which makes sense given the growing number of professionals (in all media worlds, including video and audio) favouring laptop-based systems. And to show just how feasible this really is, Apple ran the majority of their demos of *Final Cut Studio* on the new 17-inch Mac Book Pro.

Does Size Matter?

The 17-inch Mac Book Pro is functionally almost identical to the original 15-inch Mac Book Pro, which we discussed briefly in March's Apple Notes, but with a few important improvements. Obviously, from a physical perspective the 17-inch model is so-called for its 17-inch display, with a resolution of 1680 x 1050 pixels — the same as the previous 17-inch Powerbook and Apple's 20-inch Cinema Display. Since this is only 1.6 inches larger on the diagonal than the 1440 x 900 15.4-inch display on the smaller Mac Book Pro, the 17-inch's dimensions of 15.4 x 10.4 inches make this new model not so much bigger than the 14.1 x 9.6-inch dimensions of the 15-inch, and both Mac Book Pros are one inch thick. For those, like myself, who always felt the 17-inch was a bit on the big side, maybe it's time to reconsider.

However, if the 17-inch Mac Book Pro's display doesn't convince you, its connectivity might, as Apple have included a Firewire 800 port on the 17-inch (acknowledging the criticism levelled at the 15-inch model for lacking this feature), along with three USB 2.0 ports (the 15-inch, like all of Apple's most recent portables, has two), followed by all the usual suspects: one Firewire 400 port, Gigabit Ethernet, an Express Card/34 slot, a dual-link-capable DVI port with support for the 30-inch Cinema Display, two mini-jacks providing both analogue and digital audio input and output, and the new Magsafe connector for power.

Like the 15-inch model, the 17-inch features a built-in iSight camera and *Front Row* media software with an Apple Remote (although 15-inch users might be jealous of the eight-speed dual-layer Super Drive included with the 17-inch Mac Book Pro, compared to the four-speed single-layer Super Drive featured in their model). Unlike some previous 17-inch models and the 15-inch Mac Book Pro, the 17-inch Mac Book Pro is available in just one configuration: 2.16GHz Intel Core Duo processor, 1GB 667MHz DDR2 SDRAM, ATI Mobility Radeon X1600 graphics with 256MB GDDR3 SDRAM, and a 120GB 5400RPM or 100GB 7200RPM SATA drive.

The 17-inch Mac Book Pro builds on the spec of the 15-inch model by adding a larger display, along with a Firewire 800 port, an extra USB 2.0 port and all the optional extras. Could this be the ultimate mobile music system?(Photo courtesy of Apple.)

When it was first introduced, the 17-inch Mac Book Pro worked out cheaper than the 15-inch model, since the 17-inch version features as standard all of the optional extras that were available for the 15-inch Mac Book Pro: namely, the faster processor and the larger or faster hard drive. Apple have since revamped the Mac Book Pro pricing and processor speeds so that the 17-inch model retails at the original £1899 price tag, with the entry-level now-2.0GHz 15-inch Mac Book Pro costing £1399 and the high-level model (which now includes the previously optional 2.16GHz Core Duo as standard) priced at £1699. The faster or larger hard drive is an extra £80 if you purchase the high-end 15-inch Mac Book Pro.

In US pricing, you end up paying an extra $200 for the 17-inch Mac Book Pro over the 15-inch model, if you take into account the hard drive option, and this gets you the larger screen, along with a Firewire 800 port and an extra USB 2.0 port, which isn't bad value for money. It's perhaps arguable that the price differentiation between the two models should be greater. The Firewire 800 port might be a clincher for some, but it's really frustrating for those who would prefer the 15-inch form factor, especially given that the previous generation of 15-inch Aluminium Power Books all had the elusive 800 port. (As a footnote, this month I've been working on a review of the 15-inch model from the perspective of an audio engineer and musician, but unfortunately it didn't make it into this issue. So for a full, in-depth analysis of the Mac Book Pro, be sure to check out next month's *SOS*.)

Mac Book: The New iBook

With Apple's professional line of laptops being christened 'Mac Book Pro', it was widely assumed that the consumer version would be known simply as the 'Mac Book', which indeed turned out to be the case when Apple replaced the iBook line with a new Mac Book family this month. Although the design of the Mac Book is clearly a natural evolution from the iBook, there many rather more radical ▶

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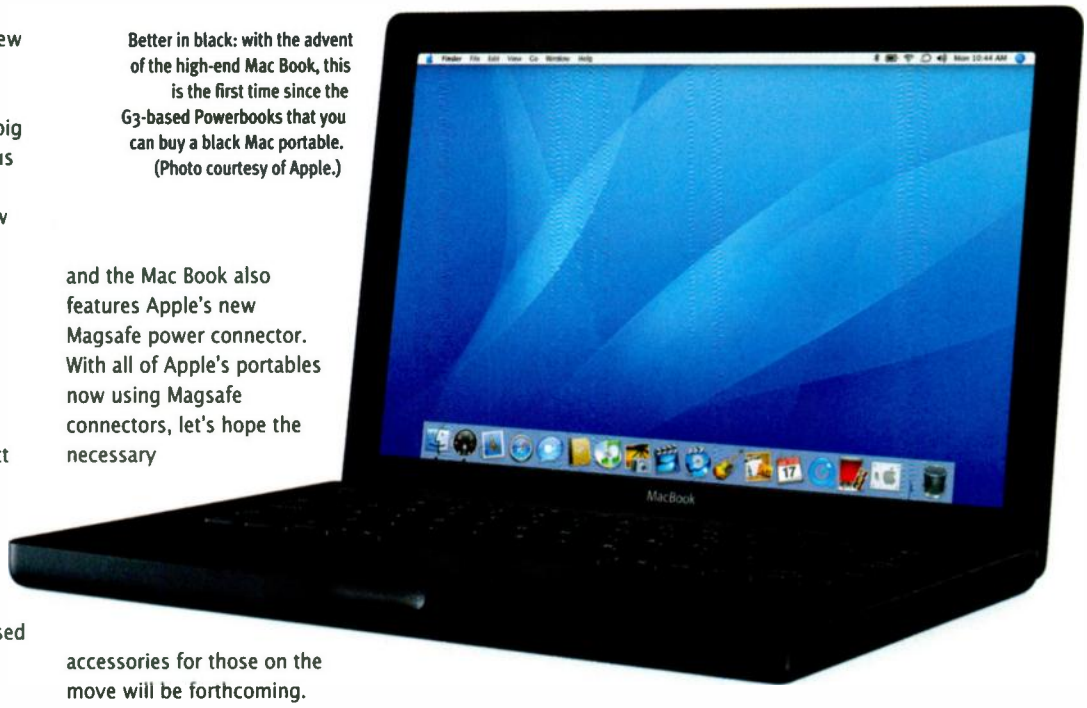
departures: for starters, the new Mac Books feature a 13.3-inch widescreen display with a 1280 x 800 resolution — a big improvement over the previous 12.1-inch 1024 x 768 display. And the TFT panel used is now of the 'glossy' variety, which, according to Apple, "lets you view graphics, photos and videos with greater colour saturation." In practice, this means that the screen is noticeably more reflective compared to the previous matt screens. Interestingly, those buying a Mac Book Pro can now choose between a matt and a glossy screen for no extra charge. Personally, despite the increased reflectivity, I think I might actually prefer the better contrast in the glossy screen.

The new Mac Book case looks quite appealing, at just 1.08 inches thick, and is available in a traditional iBook-style white finish or a new Nano-like black finish, although the colour finish you get depends on which model you purchase. Instead of the previous hooking latch system that kept the iBook shut, the Mac Book uses a much slicker magnetic latch reminiscent of the clamshell iBooks of yesteryear. The keyboard is a radical departure from any current or previous Apple portable, and although it looks as though the keys are spaced further apart than on the Mac Book Pro, it turns out that the dimensions are identical. The keyboard felt alright to type with (although I think I prefer the Pro's keyboard), but since I only had a brief play with the Mac Book I can't really comment on how comfortable it is for extended periods of typing.

Other nice touches in the Mac Book (especially when you consider the pricing of these models) that are directly inherited from the Mac Book Pro include the built-in iSight camera and *Front Row* software with the Apple Remote. You also get the two-finger scrolling-capable trackpad, which is much larger than the one used on the iBook,



Better in black: with the advent of the high-end Mac Book, this is the first time since the G3-based Powerbooks that you can buy a black Mac portable. (Photo courtesy of Apple.)

and the Mac Book also features Apple's new Magsafe power connector. With all of Apple's portables now using Magsafe connectors, let's hope the necessary accessories for those on the move will be forthcoming.

New Model Laptop

There are three models of Mac Book available, with the entry-level and mid-range models coming in white and the high-end model being sold with a highly desirable black finish. The entry-level model comes with a 1.83GHz Core Duo processor, while the other models feature a 2GHz Core Duo, the same as on the entry-level Mac Book Pro. All models offer a 2MB Level-2 Cache and a 667MHz system buss (also the same as on the Mac Book Pro) and ship with 512MB 667MHz DDR2 SDRAM. But a big difference (compared with the Mac Book Pro and previous iBook models) is that the Mac Book uses an Intel GMA950 processor for graphics, with 64MB of memory that's shared with the main computer memory.

The use of shared graphics means that the Mac Book won't be the best machine for demanding games such as *Quake 4*, but for most music software that draws 2D graphics without hardware acceleration, it probably won't make a dramatic difference initially. I say "initially" because eventually 2D graphics will be fully hardware accelerated, and at that point those using systems with shared graphics memory might achieve significantly less performance. However, while shared graphics memory is often best avoided by those looking for the best all-round system performance, early reports indicate that it may not be a deal-breaking feature in the case of the new Mac Book, for those seeking a low-cost Mac portable that's music and audio capable.

The low-end model features a slot-loading Combo drive, while the upper two models have a slot-loading four-speed Super Drive. Turning to hard drives, the lower two models feature a 60GB 5400RPM SATA drive, with the high-end model offering an 80GB device. A variety of options is available when purchasing, including 100GB and 120GB drives.

In terms of connectivity, the Mac Book offers one Firewire 400 port, two USB 2.0 ports, built-in Airport Extreme and Bluetooth 2.0+EDR (Enhanced Data Rate), Gigabit Ethernet (for the first time in an Apple consumer laptop), two mini-jacks for combined analogue and digital audio I/O, and a mini-DVI port supporting DVI, VGA S-Video or composite output. One nice thing about the mini-DVI port is that it now supports either mirroring of the built-in display (as before) or extending of the built-in display across an external display.

The low-end Mac Book costs £749, which is really not a bad price, while the mid-range model is priced at £899 and the high-end black Mac Book is yours for £1029. All models are shipping as I write. SOS

Universal *Reason* & Rewire After All

Propellerhead Software (www.propellerheads.se) officially released *Reason* 3.0.5 this month, after several months of beta testing. In May's Apple Notes we reported that *Reason* 3.0.5 would be an Intel-only release, after Propellerhead commented that the performance of the Power PC code in *Reason* suffered in the Universal Binary version compiled with Xcode. However, *Reason* 3.0.5 is in fact a Universal Binary after all, with Propellerhead claiming that this version "packs some serious Rewire performance gains for owners of older Macs" — so they must have resolved the previously discussed performance issues.

For users of Rewire-based applications other than *Reason*, Propellerhead also released Rewire 1.7, which is also a Universal Binary, meaning that Intel Mac users can now start using Rewire in earnest with Intel-native applications such as *Logic* and Ableton's *Live*.

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Martin Walker

With 40 stereo audio channels, each running a 4-band EQ, VST *Dynamics* and, potentially, a *Magneto* plug-in, the new 'DAW Bench' provides a tough but very informative test for modern dual-core computers.

A promising new PC system benchmark test has emerged that shows up a previously elusive audio interface problem. PC Notes investigates...

Back in *SOS* November 2003, I discussed benchmark testing of music PCs, pointing out that the traditional tests used by most mainstream PC magazines don't shed much light on how a particular computer will perform when running music applications.

For several years I've tested music PCs using the *Cubase* 'Five Towers' test, because its needs are relatively modest, so you can run it successfully on a wide range of PCs and compare their audio performance. However, its processor drain is so low on many modern PCs that while its results are still valid they are beginning to bear little relation to most real-world songs.

However, various more strenuous benchmark songs are now available that run rather more plug-ins and soft synths, and, in most cases, a clutch of audio tracks, bringing them more into line with what the majority of musicians are doing with their computers. Examples include the *Nuendo/Cubase SX3* 'Thonex' song, the more demanding *Nuendo/Cubase SX3* 'Fudd' test, with a different mix of plug-ins and soft synths plus loads of audio tracks, and two versions of Scott Reams' *Sonar 4* benchmark song. You can download all of these from the ADK Audio web site at www.adkproaudio.com/downloads.cfm.

'DAW Bench'

There's now a new benchmark test around that has already caused quite a stir amongst manufacturers and users alike. 'DAW Bench' (http://dawbench.vze.com) can be downloaded and run in *Nuendo/Cubase SX3* on both Mac and PC (although it's hoped that in future it can also be adapted for other sequencers).

Developed by Australian Vin Curigliano of AAVIM technology, it's been in development since August 2005, with the periodic involvement of DAW vendors, some audio manufacturers and various end users. The aim was to produce a test that could be reliably used by all parties as a reference for measuring DAW performance. What actually happened was rather more revealing, as we shall see.

The benchmark started life with the well-known 'Blofeld's Return' demo song that can be found on the *Nuendo/Cubase* CD-ROMs, which was then modified to keep the audio (hard drive) loading more constant and had 25 30-second audio tracks added to it, each containing a sine-wave tone. Then a *Magneto* plug-in (initially disabled) was added to each of the audio tracks.

Overall, this 'Blofelds DSP40' test runs 40 stereo audio tracks, 40 4-band EQ plug-ins, 40 VST *Dynamics* plug-ins, one *Multiband Dynamics* plug-in and up to 40 *Magneto* plug-ins. The idea is to run the song and then start enabling the single *Magneto* plug-in on each track in turn until you hear audio breakup, and then report the number of *Magnetos* your system can manage.

There are two strengths to this approach, compared with running a static test containing a fixed number of plug-ins and then simply reading the *Cubase* or *Nuendo* CPU meter. First, you avoid inaccurate meter readings (particularly at low latencies), but second, you also find out exactly how many plug-ins you can run on a particular system. Moreover, because even the slowest dual-core PC can manage nearly 40 *Magnetos* at 12ms latency, and you're enabling them one by one, listening for obvious audio breakup rather than relying on your interpretation of a CPU meter reading, you can reliably measure performance increments to about 2.5 percent (one in 40). Faster dual-core PC results become even more accurate because they can run even more *Magneto* instances. Such incremental accuracy not only makes it easier to quantify the relative performance of different dual-core processor families, but even to spot fairly small differences between different PCs with identical dual-core processors, which may be due to different BIOS settings, RAM speeds, and so on.

ASIO Driver Issues

I've never seen much point in testing how many instances of a plug-in you can run on a PC before it falls over, as I feel this bears little relation to the real world of the musician. However, in the case of 'DAW Bench' the balance of audio tracks, plug-ins and soft synths is a very reasonable representation of many real-world songs. It has also exposed some fascinating facts.

The most contentious of these is that while most PCs will manage to run a certain number of *Magneto* plug-ins without problems, if you save the test song at this point and then

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re-load it, it won't run without glitching unless you disable some of them again. The lower the latency, the more pronounced this effect. To give you an idea of the scale of the problem, one system managed 42 *Magnetos* at 6ms latency, but would only run properly with 29 when the same song was re-loaded.

This phenomenon finally confirms anecdotal evidence from various musicians who've reported that a particular song ran perfectly one night, but wouldn't run without serious glitching when they booted up their PC next morning. Tracking down the cause could have been an extremely frustrating exercise, but for one fact — the Save/Re-Open 'droop' didn't happen on any system running a Lynx Two soundcard. The Lynx Two not only managed to run the tests with remarkable consistency between sessions, but was also the only product across a handful of audio interfaces from different manufacturers that managed to run 'DAW Bench' right down to 3ms latency at 44.1kHz (most others refused to run it without severe glitching).

Further tests confirmed that the Lynx Two was immune to the above issues across a selection of PCs running AMD and Intel hardware, whereas both PCI and Firewire interfaces from other manufacturers, including MOTU, Presonus, RME and Terratec, had large variations reported between sessions. This appears to be an ASIO driver issue, and that impression is reinforced by the fact that RME quickly released a beta driver for their popular HDSP-series interface that hugely improves its low-latency performance in this test, on both AMD and Intel systems.

System Results To Date

The test results so far have now been gathered together into graphic form on the DAW Bench web site, and some initial conclusions drawn. Using the Lynx Two as the reference audio interface, for consistency, systems based on Intel's Pentium D940 dual-core and AMD's X2 4200+ processors turned in fairly similar results. After the poor performance of Intel's D800 series, this is good news for Intel enthusiasts and those who need maximum compatibility with audio hardware.

In this test, at least, PCI interfaces also performed better than Firewire ones with AMD systems, but this interface variance was less pronounced with Intel-based PCs. On the other hand, Intel dual-core system results proved to be quite dependent on memory bandwidth — so the most appropriate RAM modules and BIOS timings are important. Results for both AMD and Intel overclocked systems have also been published that don't display a simple proportional increase in audio performance with clock speed, which suggests that other system variables are involved.

The Future

Overall, 'DAW Bench' already has a lot of potential for exposing issues with ASIO drivers, and I suspect that it's already being used by interface manufacturers keen to improve this aspect of their product's performance. No doubt driver revisions will emerge as a result, especially now that this test is available to end-users, who will probably want to investigate how their own interface performs at low latencies, compared to others.

I suspect that the test could also prove to be a great way for both professional and DIY DAW builders to find the optimum BIOS settings for each system they assemble, since with a suitable audio interface you can reliably push any system to the edge, reboot, tweak various aspects of the BIOS, such as memory timings, and try again.

There will always be those who claim that a single test can be biased to favour certain systems (Intel or AMD, and so on), via the choice of particular plug-ins, but if you suspect this it's easy enough to replace *Magneto* with something else and check for yourself. This benchmark was specifically designed as a torture test for the audio card, its drivers, memory bandwidth and hard drives. However, it doesn't specifically stress the CPU (the component that's often the limiting factor for many musicians), so I'm looking forward to the next benchmark test in the series, which will concentrate more on gradually loading soft synths into a project. I suspect that this test may be just as revealing in different ways. SOS

PC Snippets

• HG Fortune Synths: I've not come across this German developer before, but judging by the impressive PC-only VST Instruments I've downloaded, we're likely to hear a lot more of the name in the future. Anyone who likes evolving Wavestation-style pads and textures will be tempted by the *Protoplasm TSM* synth and the spacey wave-transitions of the *STS24* (*Space Transition Synthesizer*), while those into more cutting sounds should investigate *Laserblade* and the intriguing *X-WOF II* (*Wheel Of Fortune*) which features five-part algorithmically-generated music. The free demo versions are fully functional but limited to two oscillators, while the full ones feature more voices, extra features and much larger patch collections. They retail at between 29 and 35 Euros. Sounds like another bargain to me!

W www.hgf-synthesizer.de

• New mLAN Drivers: Yamaha have released a new 64-bit version of their mLAN musical networking protocol drivers compatible with Windows XP Professional x64, with a view to making sure that their Windows Vista drivers are ready well ahead of Microsoft's own schedule for the Vista operating system. Other good news for PC users of mLAN is that by the time you read this another mLAN driver version will have been released that finally fixes the annoying random latency offset error experienced in some systems, which foxed the automatic latency-compensation features of applications such as *Cubase SX*.

W www.mlancentral.com

The *Protoplasm TSM* synth from designer HG Fortune should excite any Wavestation-loving musician with a PC.

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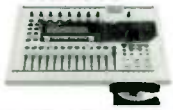

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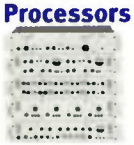

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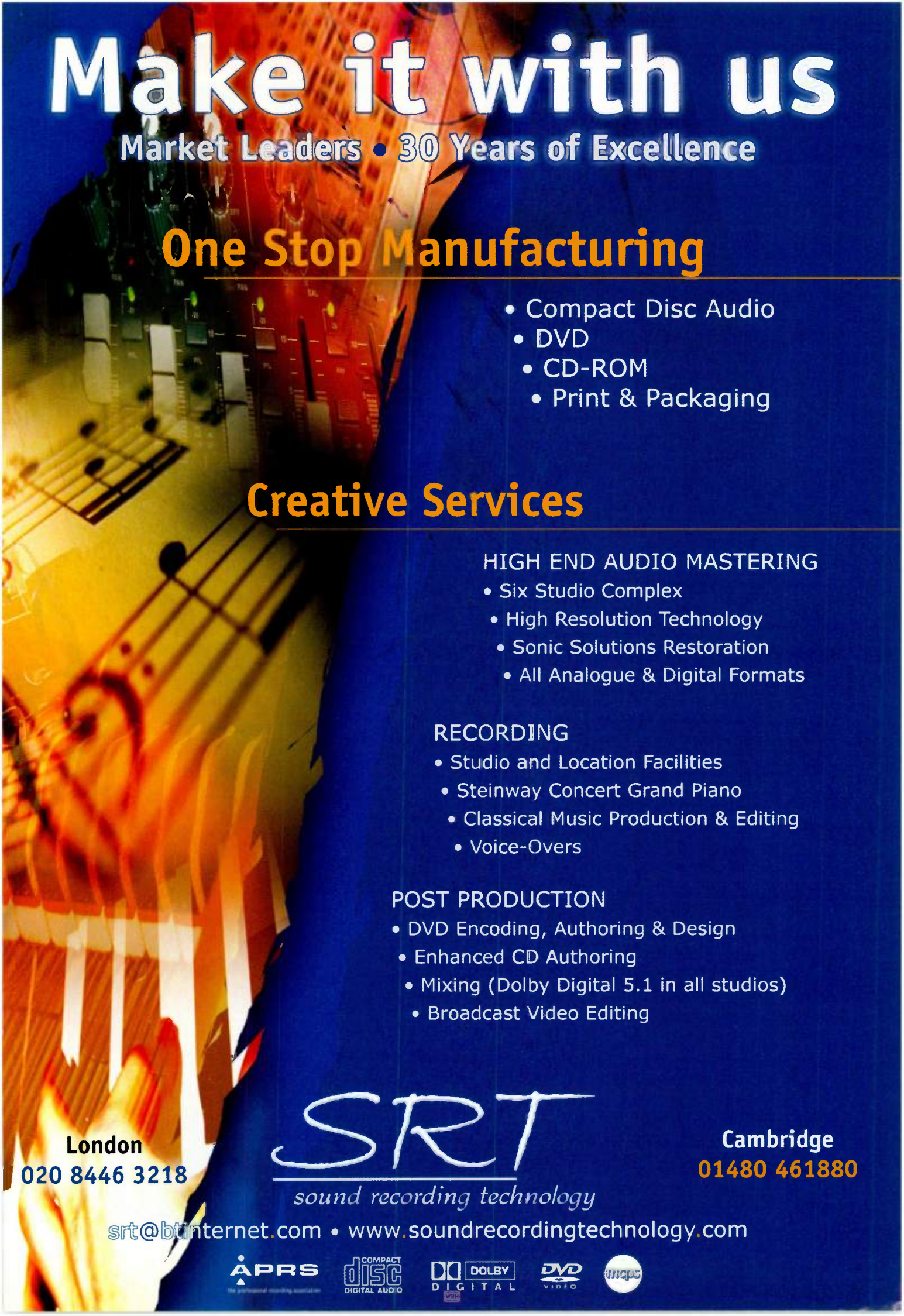

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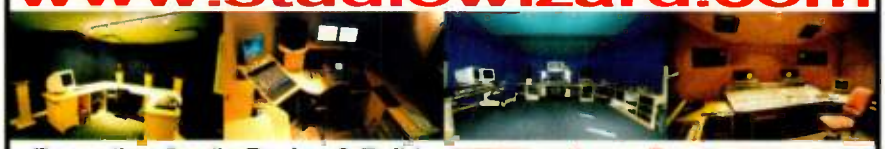

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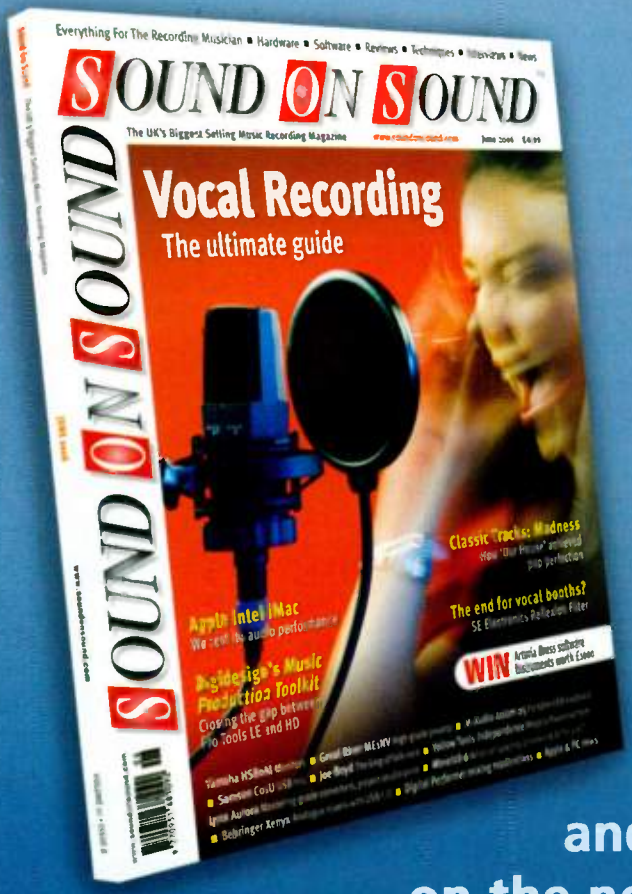

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Simplicity: for a brighter tomorrow.

David Glasper

People seem to spend an awful lot of time trying to figure out why they're unhappy with the music they've made; but it seems that often they're looking at every reason except the music itself.

You can understand why — it can be much easier to think about tangible, straightforwards things like musical equipment or problems with monitoring, than it can be to think about what is essentially abstract and emotional. The truth is that these are usually diversionary tactics — if you're not happy with your music, it's most likely that the music itself is at fault not the production or the equipment made to use it.

Pleas not to become overwhelmed by the number of possibilities offered by modern musical equipment (especially computer-based studios) are not uncommon within the pages of *SOS*. This is good advice, and the same principle can be applied to composition — remember, just because you can do something, it doesn't mean you have to do it.

You might find that your music improves if you can strip it down to its core elements. This will force you to concentrate on what's really important, and learning to recognise what's really important in a song will make you a better songwriter. The simple approach also has the knock-on benefit of making recording and mixing a lot easier.

There are no end of things you can do to simplify your music. Take the idea of rhythm and lead guitar in a standard guitar band for example — just because you have two guitarists, there's nothing to say you have to have two guitar parts, at least not all of the time. If one guitarist has a really good riff, do you really need the other one to distract from it by playing something different merely for the sake of it? Why not have them both play the same thing and use the additional guitar to bolster the sound?

Bass players are also notorious for making things unnecessarily complicated — yes your lightning-fast slap-bass scales are very impressive, but wouldn't it give the song more momentum if you just hit the same note eight times a bar and stopped mucking about?

Drummers can bring their own problems, typically through collecting far more drums, cymbals, cowbells and whatever than they could ever reasonably need. They're sods for it, basically. Sadly they often suffer from a crippling lack of self-esteem, and, although they know they only *really* need two or three drums, a hi-hat and a ride cymbal, they assemble vast collections of percussion to make themselves feel more important. This behaviour should be discouraged for three reasons: firstly because it will make them think about (and therefore hopefully improve) what they are actually playing, instead of trying to figure out how to use every last piece of the kit in every song; secondly, because it will make it a lot easier to record and mix; and thirdly because it will mean you will only need to use the one van when you play a gig.

There's also the question of what's actually being played. Just because you know tons of chords, it doesn't mean you have to use them all. Not in every song anyway.

Lou Reed said "one chord you're alright, two chords you're pushing it, three chords you've got jazz". While this is an obvious exaggeration, there's a lot of truth in it; you really don't need more than one or two chords to write a good song. 'Roadrunner' by Jonathan Richman and the Modern Lovers is testament to this — two chords all the way through (A and D, I think) but easily one of the most recognisable and driven songs you can hear. 'Tomorrow Never Knows' by the Beatles is another good example — one of the simplest structures imaginable, but still one of the most innovative songs they (or anyone else) ever wrote.

Structure is another thing that can benefit from simplification. Most people construct their songs in the same order because that's how everyone else does it, but there's really no reason to subscribe to the standard verse, chorus formula. If you've got a riff or a sequencer pattern or even just a single chord, and you think it sounds good if you play it for five minutes straight, then that's what you should do. Likewise, if you feel that what you've got is only interesting for one or two minutes, then that's how long it should be.

'Complicated' should not always be equated with 'good' when it comes to music. The important thing is that it's your music and you're under no obligation to anyone to make it anything other than what you want it to be. The only real rule you should follow is that if sounds right to you, then it is right. SOS

About The Author

David Glasper is a member of psychedelic electro-noise agitators, the Resistance. During his spare time he is Assistant Editor for *Sound On Sound*.

If you would like to air your views in this column, please send your submissions to soundingoff@soundonsound.com or to the postal address listed in the front of the magazine.

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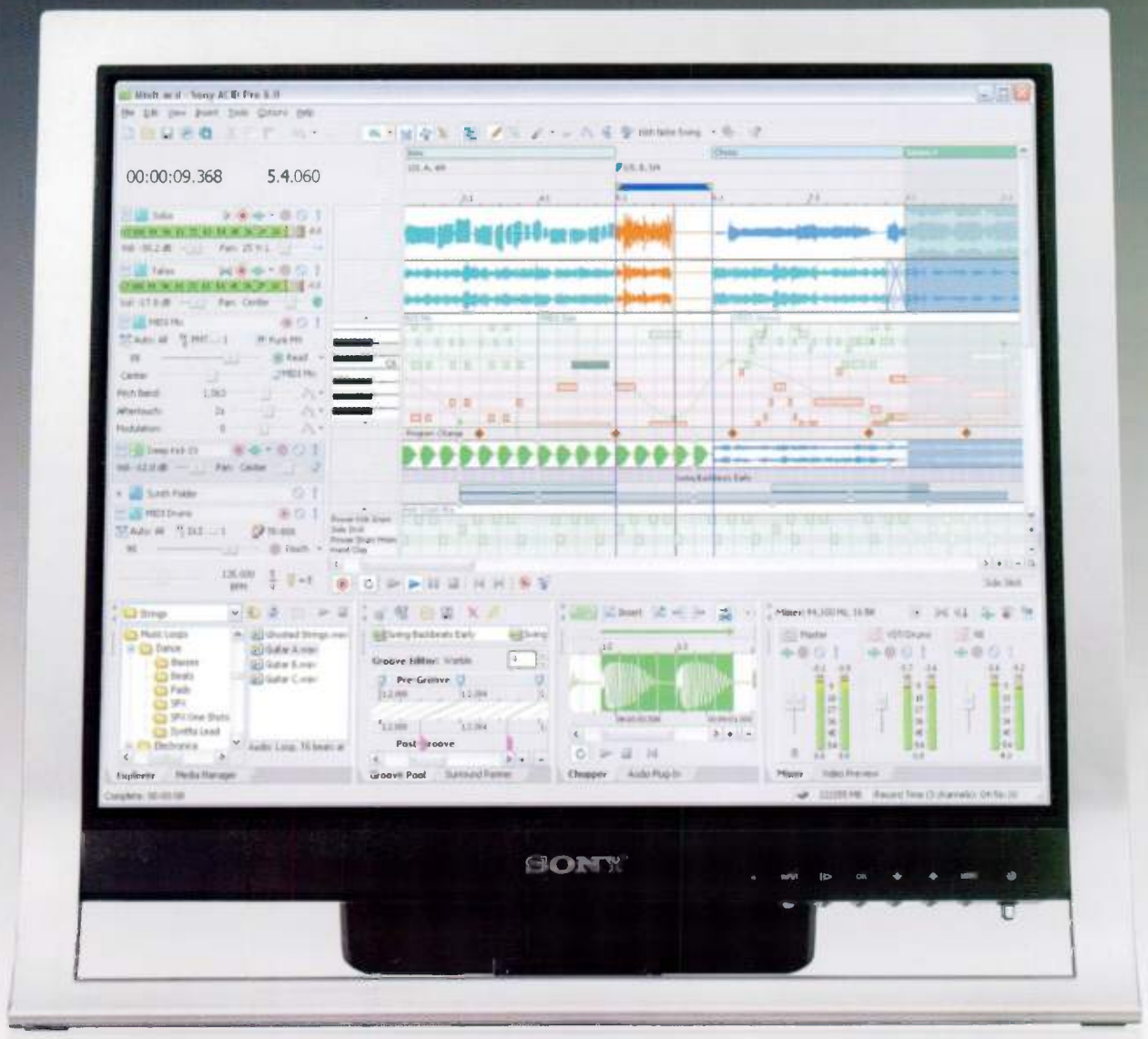

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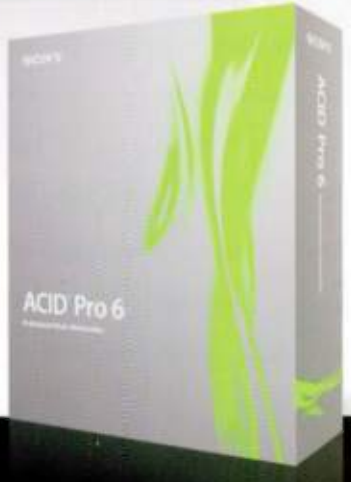

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