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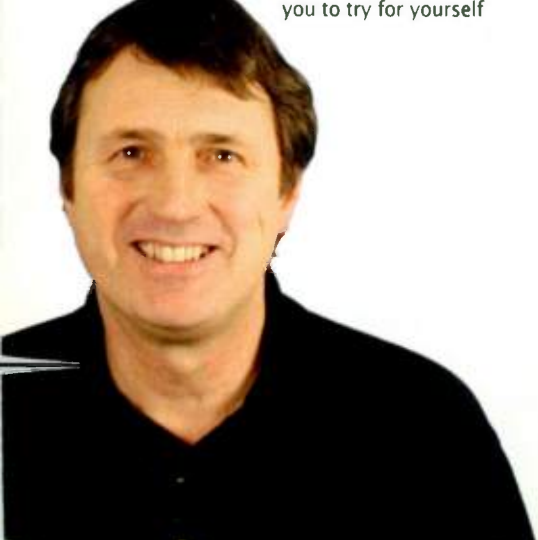
01

Listen Up!

It never ceases to amaze me that studio monitors sound very different to each other, even though their specifications claim similar performance. Our monitors are the only means we have of judging mixes, so how are we supposed to do a good job when no two models sound the same and, to further compound the situation, no two identical models sound the same in different rooms?

The reason we manage as well as we do is that the human hearing system is very adaptable, so if you get used to the sound of your room/speaker combination playing back commercial recordings that are deemed to be of a high quality, you soon develop a feel for the sound, and that becomes your new 'normal'. However, it is still very important to minimise the effect of the room on the sound, as our ears are less good at compensating for an uneven bass response, which is what we tend to end up with when using speakers in small, acoustically untreated rooms. Even moving the monitors by a few inches can change the situation. These are factors we covered in detail with our studio acoustics feature last month, but it really pays dividends to attend to them before even thinking about mixing in the room, so I make no apology for returning to the subject here.

One way of checking the way bass behaves in a room (which is often referred to in our Studio SOS series and was also mentioned in the acoustics feature) involves playing a chromatic scale of bass sine-tones, spanning a range of about three octaves. This soon shows up whether some notes are extraordinarily loud or quiet compared to the average, so we made an MP3 of this scale and put it on the *SOS* web site for you to try for yourself (www.soundonsound.com/sos/jan08/audio/sinesweep.mp3).



Where you sit in relation to the monitors can affect low-end performance, so perhaps you should evaluate your studio layout before using the tones in earnest. Experience has shown that in domestic-sized rectangular rooms it is invariably better to have the speakers firing down the length of the room, rather than across it. Small, square or cube-like spaces are generally the worst, as the listener usually ends up sitting close to the centre of the room, where the bass response will fluctuate the most.

So what do you do if, after trying your best to acoustically treat your room and optimise your speaker position, the result is still less than acceptable?

One approach is to not apply any EQ at all below the frequency at which your room starts to sound uneven, which may be as high as 150 to 200Hz in smaller rooms. That way you can get the music mastered somewhere that has decent monitoring and let them EQ the low end for you. Another tactic is to check your mixes on as many different sound systems, in as many different locations, as possible and keep adjusting the mix until it sounds acceptable on all of them. Back in the days of tape, this often meant making a cassette copy and playing it in the car, but now you are more likely to make an MP3 to play back on your iPod, or a CD to play on a standard hi-fi system.

Personally, I use one of those cheap radio transmitters that have now been legalised for use in the UK, which are designed to allow you to play your MP3 player through the car radio (where no physical input jack is available). These have a range that covers a typical house, so if you plug one into the headphones output of your audio interface or mixing desk you can tune into your mix on any FM radio in the immediate vicinity. There's still no substitute for accurate monitoring, but by making your mixes sound good on a number of audio systems, you can at least minimise the problems.

Paul White *Editor In Chief*

SOUND ON SOUND

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

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

editorial

E sos.feedback@soundonsound.com

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

Publisher Dave Lockwood

advertising

E adsales@soundonsound.com

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

production

E graphics@soundonsound.com

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

marketing

E marketing@soundonsound.com

Marketing Manager Andy Brookes

administration

E admin@soundonsound.com

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

subscriptions

E subscribe@soundonsound.com
W www.soundonsound.com/subs

Circulation Administrator Lisa Pope

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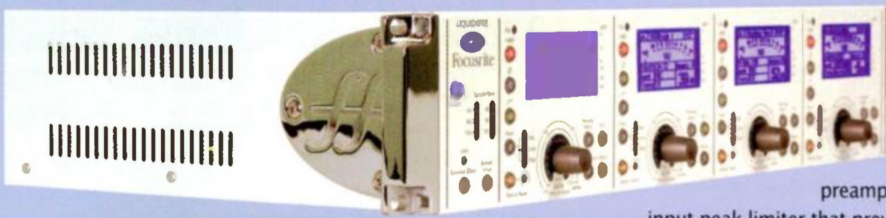
Focusrite Liquid 4 Pre Four channels of flexibility

The latest product in Focusrite's Liquid range of dynamic convolution processors was released at the New York AES show in October 2007. The Liquid 4 Pre, as its name rather cryptically suggests, is a four-channel preamp, based on Focusrite's highly regarded Liquid technology, which can be used



to emulate the sound of classic equipment. As avid gear lovers will know, there are already two Focusrite Liquid units on the market: the Liquid Channel, a mono channel strip with convolution-based mic preamp and dynamics processing emulations, and the Liquid Mix, a desktop unit that essentially functions as a DSP engine for dynamics-only convolution.

The Liquid 4 Pre offers none of the dynamics-processing features of the Liquid Mix, but has four of the acclaimed preamps from the Liquid Channel. The unit comes with 40 classic preamp models, giving the user the choice of what amounts to a whole rack full of processing gear, but in a 2U device, for £2113 including VAT.

Each channel has phantom power, phase reverse and a high-pass filter, as well as controls for selecting line, mic, or instrument inputs. The rest of the functions are digitally controlled using a multi-purpose knob and a generous LCD on each channel. As expected, the knob controls the preamp gain by default, but it can be used to alter other functions, including the amount of harmonic distortion introduced by the preamp. Each channel has a 'Session Saver' input peak limiter that prevents fast transients clipping the output.

On the rear panel, there are analogue inputs and outputs on XLR, as well as ADAT and AES in and out. But there are also Ethernet ports for remotely controlling the device, and for transmitting audio over Cat 5 or Cat 6 cable, using the Ethersound protocol (for which an optional card is required, costing £699). Included with the Liquid 4 Pre is a plug-in for Pro Tools, so you can remotely control the device's functions, and you can link up multiple Liquid 4 Pres in a system, using the Ethersound connections.

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

New cans and keyboard from M-Audio

In the past, M-Audio's main area of product development was that of controller keyboards and audio interfaces, with notable examples being the Audiophile and Delta soundcard ranges, and Oxygen and Keystation controllers. But in recent years M-Audio have broadened their areas of expertise into domains such as monitors and earphones. The latest step in this phase of development is the launch of a pair of closed-back headphones, the Studiophile Q40s.

These pro-quality cans are designed for users who need to monitor accurately on the move, but also for those who need significant isolation from external sources. The Q40's ear cups are large enough to enclose the wearer's ear, and the backs of the cups are fully sealed, to minimise the amount of sound entering — and escaping from — the phones. M-Audio claim that they can reproduce sounds from 10Hz to 20kHz, and they have an impedance of 64Ω, meaning that they should be able to be driven happily by most headphone amplifiers.

The Q40s come with a detachable cable and threaded quarter-inch to eighth-inch adaptor. They fold up neatly into a storage bag, also included, preventing damage and making them easier to carry around. They cost £109 and are available now.

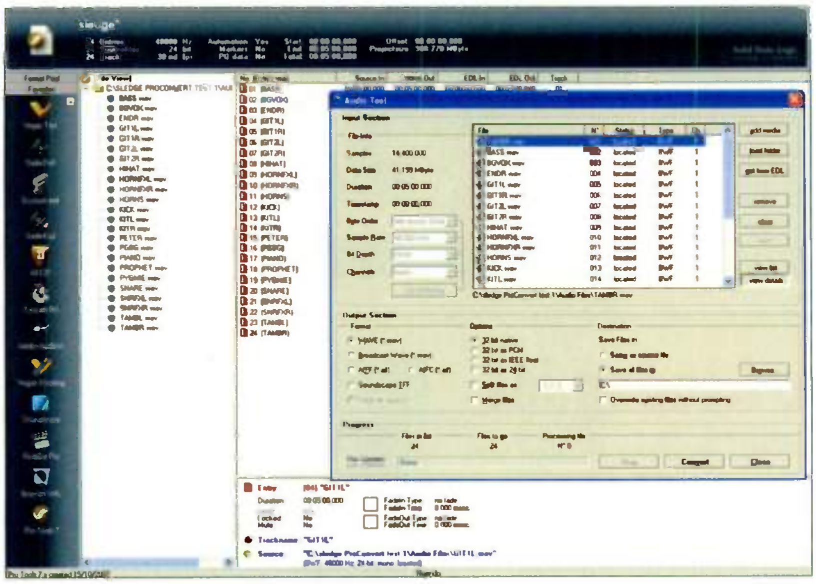
Also new from M-Audio is the Keystudio 49i, an interesting new MIDI controller keyboard with a few tricks up its sleeve. Apart from its USB/MIDI controller functionality, the new device also has a two-input, two-output audio interface, plus (and this is the interesting part) a General MIDI sound bank, and what M-Audio call a "premium" piano synth built-in, apparently sampled from a Steinway. This means that, with the Keystudio 49i, you can carry all the input and output hardware necessary for a basic portable laptop rig in just two bags: one for the laptop, and one for the keyboard, with just one cable required to connect the two! Then, on the tour bus after your gig, you can whip out your keyboard, dial up the piano sound, and write your next song! Because it contains an M-Audio interface, the Keystudio 49i is compatible with Digidesign's Pro Tools M-Powered, and it comes with Ableton Live Lite, enabling you to get writing immediately. The Keystudio 49i is also available now, costing £179.

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

SSL Pro Convert Levels the field

In the news section of last month's *Sound On Sound*, we reported the announcement of Solid State Logic's Pro Convert v5 utility. However, we only briefly mentioned it in the AES show roundup, so here are more details on what looks to be a very promising piece of software.

Pro Convert is a translation application that converts DAW projects from one format to another — for example, from Nuendo to Pro Tools, from Soundscape to SADiE, or vice versa. Although this doesn't sound terribly exciting, it means that users who employ different software platforms can move their projects between applications without having to go through multiple stages of manual export.



Pro Convert will translate audio files, region data, and pan and level fades. But if you have a source session with EQ and compressor plug-ins on each channel, for example, settings won't be transferred to the destination format. However, in the future, one can speculate that such facilities could become available.

Pro Convert was acquired by SSL when the UK-based company bought out German software engineers Cui Bono Soft, just before the New York AES 2007 show, in late September 2007. Cui Bono Soft had been developing Pro Convert under the name EDL Convert for seven years, aiming at post-production and professional users. Now SSL hope to further the development and bring it to a wider audience. Anthony David, SSL's Managing Director, commented, "the Cui Bono guys had a vision of levelling the DAW playing field that we want to pick up and run with".

SSL's Pro Convert is expected to become available during the first quarter of 2008. Keep your eye on SSL's web site for the latest news.

www.solid-state-logic.com

Aphex fill your ears With Headpod 454 headphone amplifier

Hardware experts Aphex have released a new four-channel headphone amplifier called the Headpod 454, which we first saw almost a year ago at the 2007 Winter NAMM show. It's a slightly-larger-than-palm-sized unit with four independent headphone amplification circuits, each with its own volume control and quarter-inch jack socket output. Inputs are also on quarter-inch jack, and there's a switch to select between a single, unbalanced stereo jack input and a pair of balanced mono TRS sockets. A master volume control trims the source signal.

Aphex say that the Headpod 454 can power all types of headphones, and the per-channel output power versus load impedance graph they supply certainly seems to back up the claim. Providing nearly three Watts per channel to an 8Ω load, dropping to around 100mW on a 1kΩ load, the Headpod 454 should be able to go very loud — hence the caution notice on the top panel — so will be perfect for drummers in the studio or on stage. This means, of course, that the device should also be able to deliver cleaner amplification at higher levels than some standard headphone outputs, such as those found in typical audio interfaces. The Aphex Headpod 454 is available now, at a cost of £170.

SCV London +44 (0)20 8418 0778

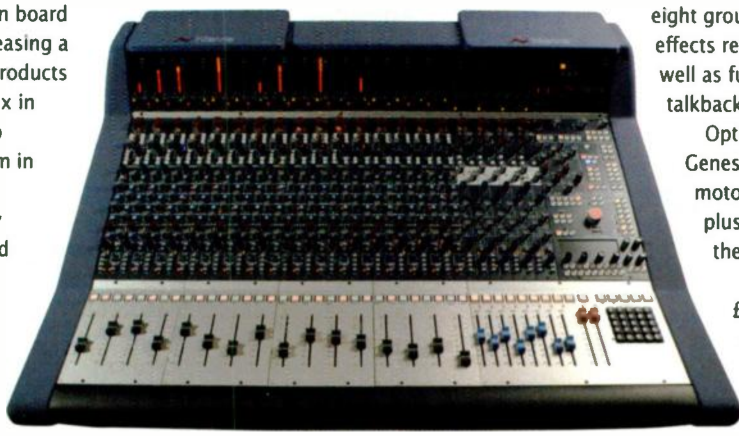
www.scvlondon.co.uk

www.aphex.com

Invisible touch Neve Genesys announced

With the decline of major commercial studios over the past decade, the need for expansive analogue consoles the length of an average family car is decreasing. Studio mixer giants AMS Neve have taken this on board over the past few years, releasing a number of rackmountable products that appeal to those who mix in the box, but who still like to retain a little analogue charm in their signal path.

Now the British company have gone an extra step, and created the Genesys, a console that should turn heads in the small professional studio market. The Genesys is available with up to 60 channels, but it's the base configuration, a 16-channel compact console, that may prove to be the most popular. It's got a total of 32 inputs, with 16 of Neve's revered mic preamps, and 16 tape return inputs. Of course, the Genesys acts as a summing mixer, and it has DAW control for driving software applications such as Logic, Pro Tools and Nuendo. What's more, there are eight group buses, eight aux buses, four effects returns and per-channel metering, as well as functions for monitoring in 5.1 and talkback.



Optional extras can be added to the Genesys, and buyers can specify motorised fader automation and recall, plus additional EQ and dynamics for their console.

The Genesys base model costs £33,546 including VAT, but excluding shipping.

AMS Neve +44 (0)1282 457 011
www.ams-neve.com

Roland open Music Academies

Roland UK have launched a scheme to promote the music industry in education, with the opening of two Music Academies at Walsall College, West Midlands (www.walcat.ac.uk) and the Rotherham College of Arts and Technology (www.rotherham.ac.uk). The scheme is designed to offer specialist training facilities to students who want to widen their career opportunities in the music industry. It runs alongside each college's existing BTEC National Diploma or equivalent NCFE courses in Music and Music Technology, and is classed as 'enrichment'. Students get a course handbook written by Roland, and assignments are delivered and monitored by college staff, but moderated by Roland's education division. The two-year course culminates in a *viva voce* interview, which is assessed by Roland UK's Head of Education, David Barnard.

Roland have also announced a new certification, called the Certificate in Music and Interactive Technology, which was developed in conjunction with the Rock School examination board.

Students attending courses at the Roland Music Academies will be taught about career development, product knowledge, marketing and communications, and there will be presentations by Roland artists and guest speakers. A third Music Academy is to follow at the University of Glamorgan in Spring 2008.

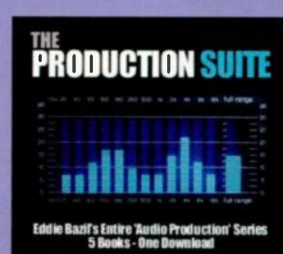
Roland UK +44 (0)1792 702701 www.roland.co.uk

Customs seize fake Shure mics

Customs officials in China, working closely with mic manufacturers Shure Inc (**www.shure.com**), have seized a shipment of counterfeit microphones — mainly SM57s and SM58s — which were *en route* to Indonesia. The shipment, which was being handled by the Shen Qiao Xing Trading Company, was detained by officials in Huangpu, Guangdong Province, and was found to contain not only Shure rip-offs, but fake Pioneer, Sony and Hitachi goods as well. Sandy LaMantia, President and CEO of Shure commented, "we are committed to fighting it [*counterfeiting*] so that our customers are confident every time they purchase a product bearing the Shure name". The company recommend that you purchase their products from authorised Shure dealers only, and they offer a counterfeit-checking service.

Samplecraze E-books

Web-based audio specialists Samplecraze (**www.samplecraze.com**) have released a new title in their range of E-books on music production, which are created by Eddie Bazil. The latest is the *Beat Production Bible*, which takes the user through the ins and outs of creating beats, covering topics such as quantisation, sample manipulation, processing and mixing. It costs just £11.99 and comprises a PDF file and an audio download, so you can hear examples while reading the tutorials. The *Beat Production Bible* is included in Samplecraze's *Production Suite*, a collection of five E-books. Costing £34.99, the *Production Suite* covers drum layering (at beginner and advanced levels), EQ, and mixing, as well as the topics in the *Beat Production Bible*. To purchase any of Eddie's E-books, head to the Samplecraze web site. If purchasing the *Production Suite*, a broadband connection is recommended, as it's around 200MB in size.

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New Samson power distributors

Samson have unveiled a new range of their power distribution and rack lighting devices, comprising three new products: the Powerstrip PS10, the Powerbrite PB10 and the Powerbrite PB10 Pro.

The Powerstrip PS10 costs just £49 and features eight switched AC outlets, which distribute mains power to the equipment in your rack. The Powerbrite PB10, which costs £79, adds a pair of bright LED light clusters, with dimmer knob, to the spec of the PS10, while the PB10 Pro (£129) also features these, plus an ammeter and a voltmeter with dedicated readouts on the front panel.

All three models have a 10-amp circuit breaker built-in, with front-panel reset button, and an additional unswitched AC outlet on the front for convenience. The LED-equipped models also feature a USB port on the back, for connecting an included gooseneck LED that illuminates the back of your rack.

The three new models are all available now, through Sound Technology in the UK.

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

Competition winners

Get the Music Mill treatment

In September, the lucky winners of the Music Mill/PMI Audio equipment competition featured in *SOS* August 2007 made it down to Devon to collect their prizes and attend the weekend studio engineering course at the Music Mill.

Darrell Walker (pictured far left), from County Down, and John Simmons (far right) from Tunbridge Wells, Kent, spent two days in the studio with Music Mill co-owner and seasoned engineer Malcolm Toft (centre), learning what Malcolm refers to as "real skills", such as the principles of microphone placement and using a large-format analogue console. Malcolm also handed over the hardware that was up for grabs: a Toft Audio ATC2 twin channel strip, which went to Darrell, and a Studio Projects C1 large-diaphragm condenser mic, which went to John. Both winners were understandably happy with their prizes, and no doubt they'll be creating great results in their own studios, thanks to Malcolm's training.

Sound On Sound would like to thank Malcolm Toft, PMI Audio and the Music Mill for supplying such great prizes. For more details on the Music Mill and PMI Audio, check out their respective web sites.

Music Mill +44 (0)1626 361999 www.themusicmill.co.uk
PMI Audio UK +44 (0)1803 215111 www.pmiaudio.com

A-T's lifetime warranties

Audio hardware manufacturers Audio-Technica (**www.audio-technica.co.uk**) have announced that they will be offering lifetime warranties on products in their '40' series of studio condenser microphones. The warranty is free of charge, and comes with all 40-series mics purchased from authorised resellers from 1st November 2007. The scheme, which shows the company's confidence in their products and commitment to their customers, is running worldwide, and includes respected models such as the revered 4033, the 4040 and the 4050 large-diaphragm condenser mics.

Schertler's UK distribution

Swiss audio transducer manufacturer Schertler (**www.schertler.com**) have appointed Shropshire-based sound specialists Systems Workshop (**www.systemsworkshop.com**) as the new UK distributors of their products for the pro audio market. Schertler's range includes PA equipment, guitar amps and musical instruments, but they're best known for their dynamic and electrostatic pickups, of which there are a number, designed for use with a variety of different instruments. The Schertler range should be in stock at Systems Workshop by the time you read this. You can call them on +44 (0)1691 658 550.

Helping hand for SAE students

The London branch of global audio engineering school SAE (**www.saeuk.com/london**) have teamed up with American student loan provider Sallie Mae to offer prospective students another source of much-needed funds. SAE London are among the first to be approved for Sallie Mae's new UK offshoot, which offers similar financial support services to students of other independent educational institutions. Applicants can borrow up to £20,000 per academic year, although the loan amount is derived from the full cost of the course (approved by the education establishment), plus the estimated cost of living. The London initiative follows successful partnership between SAE and Sallie Mae in the USA, where thousands of students have used funding from the loan provider.


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CYCLE
ADD
REC
STEREO
9/10
11/12

Double take
HHB CDR882

HHB, whose purple-faced Burn It range of CD recorders has proliferated in studios over the past few years, have announced a new 'Dual Burn' CD recorder, the CDR882. It's a 2U rackmountable device with two CD drives, each capable of playing and recording CDs. The CDR882 has some neat recording modes, which enable the user to seamlessly spread a long recording over two or more discs, record to two discs simultaneously or duplicate CDs — at speeds limited only by the media in use. It has digital inputs in AES-EBU, S/PDIF and Toslink formats, with on-the-fly sample-rate conversion that can convert rates between 32kHz and 96kHz to 44.1kHz, for the CD standard. The device also has balanced and unbalanced line inputs and outputs, as well as a word clock input. Usefully, the user can plug a PS2 computer keyboard into the front or back of the unit, allowing CD Text entry.

HHB's CDR882 Dual Burn CD recorder is set to cost £645 including VAT when it becomes available in January.

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

MXL V88 & Mic Mate

Mic and audio accessory manufacturers MXL have added to their line of products with a new large-diaphragm condenser microphone and a USB-powered in-line microphone preamplifier. The new mic, the V88, features a 32mm capsule with a fixed cardioid polar pattern and a self-noise figure of 14dB (A-weighted). MXL claim that the mic can handle frequencies between 20Hz and 20kHz and SPLs of up to 138dB, with, according to the manufacturer's literature, Total Harmonic Distortion (THD) of 0.5 percent at these levels. The mic comes with a case and a shockmount.

The Mic Mate is the second new product from MXL. It's a one-way, single-channel audio interface with mic preamp, designed for people on the go. As such, the only connections on its marker-pen-sized body are an XLR input and a USB output. It will supply phantom power for condenser microphones, and it has a three-position gain control with options for high, medium and low. Its A-D converter runs at 16-bit, at sample rates of either 44.1kHz or 48kHz. At the time of going to press, pricing details were unavailable for both the V88 and the Mic Mate.

Sound Control +44 (0)870 067 2922 www.soundcontrol.co.uk
www.mxlmics.com

Vocal booth builder wins big!

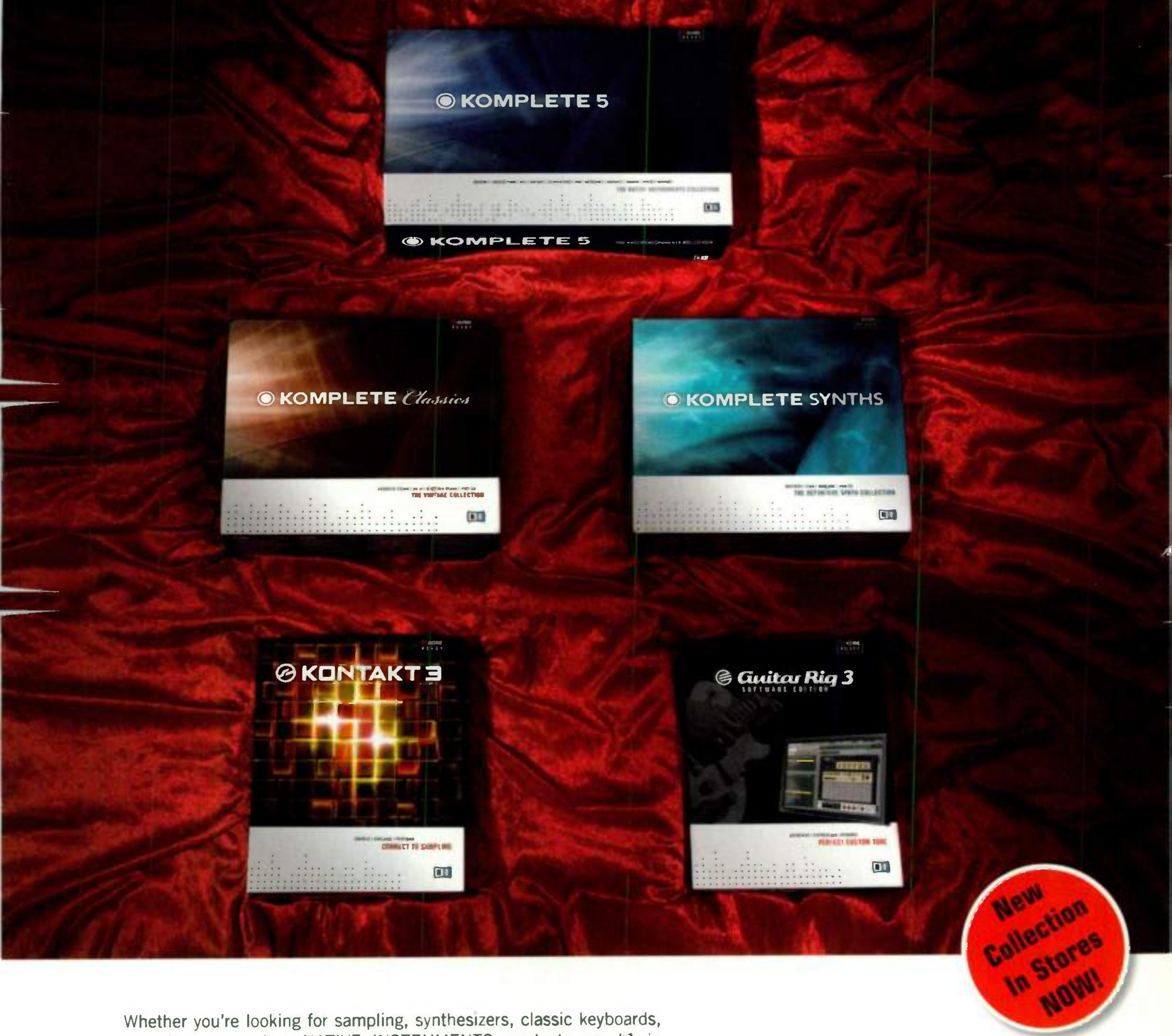
Al Cherin's London home studio suffers from a considerable amount of noise from aeroplanes and trains, making clean audio recording difficult. So last year he bought an E-book from on-line acoustics resource Project Studio Solutions (**www.projectstudiosolutions.com**) and set about researching, planning and, eventually, building an isolation booth. Coincidentally, Project Studio Solutions were running a 'build your own vocal booth' competition, the grand prize being a two-year on-line subscription to *Sound On Sound*. So Al entered, and won!

Al's booth is double-walled and has a 10cm cavity filled with Rockwool. Rubber buffers are used to decouple the structure from the existing walls and the floor. The window is double-glazed, with two different thicknesses of laminated glass, and the door features a rubber-sealed compression latch. Inside, the walls are lined with acoustic foam, and there's a wall box into which to patch a mic and headphones. Amazingly, the whole project — including acoustic foam, home-made sound absorbers, mic box and wiring — cost under £700!

City Lit offer new courses

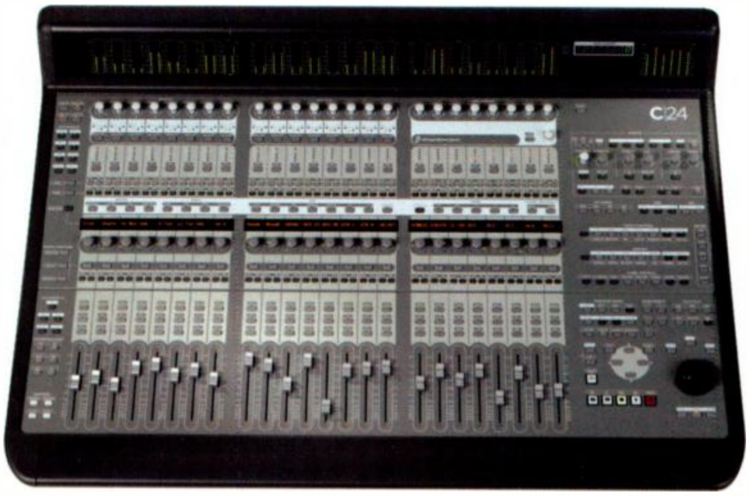
London adult education centre City Lit have announced two new courses that will start mid-January. The first, An Introduction To MAX/MSP, needs no explanation, really. The course is aimed at beginners, and will teach them the main features of the program, as well as instilling good working practices in students. Normal cost for the course will be £183, although there are concessionary and senior rates of £58 and £112 respectively. It runs for 10 weeks, with one session per week. The second course is titled New Music and covers modern classical composition, graphical scoring and the use of technology in compositions. It also runs weekly for 10 weeks and costs £183 as standard, although there's a concessionary rate of £58. Check out **www.citylit.ac.uk** for details and an enrolment form.

EXPERIENCE FIRST CLASS.



Digidesign C24

New mid-sized control surface for Pro Tools



Digidesign have launched their first medium-format control surface for Pro Tools since the acclaimed Control 24, which was released back in 2000. According to the company, the new C24 "draws upon the legacy" of the Control 24, offering a "powerful and affordable solution" that adds the ability to control Pro Tools in a tactile way.

The C24 has 16 analogue inputs that can accept mic, line or instrument signals, and a versatile monitoring section offering facilities to control 5.1 monitoring setups. Also built into the C24 are talkback and listen-back control, and an analogue submixer that mixes up to eight stereo line inputs to a stereo pair. Control surface functions include 24 full channel strips, each with a 100mm fader and dedicated, illuminated buttons for standard functions such as mute, solo and select, as well as for activating EQ, dynamics and automation in Pro Tools. Also on each channel is an assignable rotary encoder and a two-row LCD scribble-strip display.

The new console has a well-equipped meterbridge, with 24 pairs of LED bar-graph meters for displaying signals on the channel strips, plus six meters for monitoring each of the audio streams of a 5.1 surround mix. Other functions on the console include full transport control, a scrub/shuttle wheel, and buttons for direct access to many of the functions inside Pro Tools.

The Digidesign C24 is available now, at a cost of £7044 including VAT. For a limited time, Digidesign are offering two attractively priced bundles that include the C24. Firstly, there's a Pro Tools HD package that comprises the console, an HD 2 Accel system, a 96 I/O A-D/D-A converter, and a bundle of plug-ins, for a total price of £13,390 including VAT. A second, Pro Tools LE-based setup includes a 003 Rack, DV Toolkit 2 and plug-ins for just under £9200. Visit your local Digidesign retailer for further details.

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

Native Instruments

Join forces with Thames Valley University

London College of Music, part of Thames Valley University (TVU), have announced a partnership with software developers Native Instruments, resulting in the birth of a new training facility. LCM's existing Mac labs have been renamed NI labs, and are now equipped with Native Instruments' Kore 2 systems and the latest Komplete 5 bundle of software. A total of 166 workstations on both Reading and Ealing campuses have been configured with the NI gear, and the use of new software is now featured in numerous modules on a variety of courses. For example, students creating experimental music may be encouraged to use Reaktor for its powerful programmability, while those following a more contemporary composition route may decide that one of the more conventional software instruments, such as B4 II or Elektrik Piano, best suits their needs.

In mid-November, TVU hosted an event to celebrate the opening of the new facilities that saw a live set from experimental drum & bass producer Tim Exile, along with product demonstrations from NI.

Thames Valley University +44 (0)800 036 8888
www.tvu.ac.uk

Korg update KP3 and Radias

Users of Korg's Radias synth workstation and the KP3 Kaoss Pad can download updates for their machines. The Radias update, available from Korg's support web site, brings a number of new features to the synth, such as additional destinations for the Virtual Patch function, support for Korg's Komponent system (which allows connection to the M3 sampling workstation) and Windows Vista and Intel Mac compatibility for the Radias Sound Editor. Also available is a collection of sounds created using some of the new features in the update.

The KP3 download, also available from the Korg web site, adds new sample-triggering options, such as a gate function, which only plays back the sample when the sample pad is being pressed down. Other additions include a polyphonic sample mode, so four samples can be played simultaneously, then mixed on the fly; and new loop modes, which enable the user to change the starting point of the sample by running their finger across the KP3's Pad. For full details on both these updates, visit **www.korguksupport.co.uk**.

Cambridge Music Technology launch new courses

A new range of day courses has been launched by Cambridge Music Technology (CMT). Designed to provide convenient and practical training for home and project studio users, the courses cover topics such as vocal production, electric-guitar recording, drum miking, piano recording, and mixing, with special emphasis on exploring the techniques employed by top-name producers. Attendees of the courses will, say CMT, be given live demonstrations of equipment and techniques, so that they can compare different studio gear and recording approaches first-hand. In the vocal production course, for example, participants will be able to take part in a shootout of mics costing between £80 and £2000, allowing them to make their own mind up about what yields the best results (and value for money).

Cambridge Music Technology was founded by prolific *SOS* contributor Mike Senior, author of cover features including Recording Guitars (*SOS* August 2007 and on-line at www.soundonsound.com/sos/aug07/articles/guitaramprecording.htm), and the piano recording piece that you can find starting on page 106 of this issue. The course content is based around in-depth research on popular techniques that are used by professionals for hit records.

The courses, delivered by Mike and his hand-picked team, cost £145 per person, and run from 10am to 4pm, with time for Q&A. They take place on Saturdays in selected locations around Cambridge, a 45-minute train ride from London, making it convenient for those who work or study during the week. For more information, or to enrol on one of CMT's courses, visit the Cambridge Music Technology web site. The first workshops will take place in March 2008.

www.cambridge-mt.com

Lahaina Studios re-establish themselves

Devon-based studio and record label Lahaina UK are getting back to normality after studio owner Chick Holland was involved in a serious car accident in 2002. Back then, Lahaina had just completed the building of a new studio and were ready to begin kitting it out with new gear. It has taken five years for this to happen, as Chick was out of action with non-stop headaches and acute tinnitus.

Now, Lahaina are up to full speed with their recording, mastering, CD production and promotion services. Chick is currently looking for new artists to develop, using his contacts with major labels, radio stations and music pluggers.

In another development since the accident, Lahaina UK have been appointed as resellers for the new Drobo data storage system (**www.drobo.com**). For full details on Lahaina's services, check out **www.lahaina-studios.com**.

Marantz on the move

With pocket-sized PMD620

The latest professional product from hardware manufacturers Marantz is the PMD620, a portable solid-state stereo recorder. Like its competitors, it has a built-in stereo condenser microphone and mini-jack connections for an external mic input (so you can bypass the built-in ones), a line-level input and output, and headphones. There's also a 'remote' socket, which can be used alongside the Marantz RC600PMD remote (not included), with which you can start and stop recording from a handheld trigger.

Costing just shy of £300, the PMD620 is a worthy option for professionals and amateurs alike, and it has certain features that other similar products do not have. For example, there's a mono loudspeaker on the back of the unit, so you can play back audio, as you can with a dictaphone, and there's the facility to edit audio inside the machine, without having to copy files to a PC or Mac. Should you wish to bring your recorded audio into a computer-based editing package, however, you can, by simply plugging the unit into the computer and dragging files off the machine. The Marantz PMD620 can record PCM WAV files at up to 24-bit/48kHz, but it can also record MP3 files at rates as low as 32kbps, allowing for longer audio-recording capacity. Data is written to an SD card, and the PMD620 is compliant with the SDHC format, which allows for high-capacity cards. As such, the maximum data capacity is limited only by the installed memory card.

Bundled with the device is a collection of useful peripheral equipment, including an audio cable, a USB lead, an AC adaptor, a strap, a tripod adaptor — enabling the PMD620 to be mounted on a stand — and an SD card. It's available now.

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Q&A

Your technical questions and queries answered

Q How should I compress a classical recording?

I recorded a performance of Handel's *Messiah* recently, and I was wondering what the common practice is when compressing the whole mix in classical recordings. During the session, I put a compressor over the master outputs, just to catch any stray peaks, and when fiddling around with the settings I found that a low threshold and a low ratio helped to blend the mix as a whole and fattened it up. Is this acceptable?
***SOS* Forum Post**

***SOS* contributor Mike Senior replies:**
The question is what you're trying to achieve. With a piece like this, which has a wide dynamic range (between the quieter Recitatives and the full-scale Hallelujah chorus), I'd certainly recommend reducing the dynamic range a little to make the CD more suitable for home listening. The most transparent way of doing this would be to use simple fader automation, riding up the quieter sections to make them more audible. I wouldn't go for much more than about a 6dB increase to the quietest sections if you're unsure how far to go. The advantage with this approach is that a human engineer can intelligently anticipate changes in the signal in a way that no compressor can.

Another thing you can also deal with using fader automation (or even audio editing) is ducking any brief signal peaks which are unduly loud, which allows you to achieve a louder final CD. Some might suggest limiting or even soft-clipping to achieve a similar effect, but neither will sound as transparent, so I'd stick with fader automation myself.

If you're wanting a little more detail and ambience to the sound, by all means try the low-ratio, low-threshold compression you mentioned, as this will usually work fine on most types of music. Don't stray over a ratio of around 1.1:1 for classical recordings, though, if you want to play things safe, and if you're getting gain-reduction of more than about 4-5dB, you've probably got the threshold set too low. I'd personally set the attack time fairly fast to track the signal levels pretty closely, and then go for faster release times for more detail/ambience and longer release times for less detail/ambience, but this will inevitably be a matter of taste. Any isolated accented chords will be particularly revealing of potentially unpleasant compression artefacts, so listen out for how those sound.

You might be tempted to use multi-band compression with similar settings, as many people do when working with more modern music styles, but I'd steer clear of this, to be honest. The fluctuating tonal changes that arise from this kind of processing are likely to upset the delicate balance of the performance.

A more transparent approach to compression is to use a compressor as a send effect, mixing the compressed signal in with the unprocessed one — this is often referred to as parallel compression. For this to work, you need to make sure that the compression processing doesn't also introduce any delay, otherwise you'll get a nasty kind of static phasing sound. That said, most software DAWs now have comprehensive plug-in delay compensation, so this is becoming less of a problem for people these days.

When working like this, you can usually get away with slightly heavier compression, but I'd stay below a ratio of 1.3:1 to be on the safe side. What some engineers do is automate the compressed channel's fader, rather than the main channel's, adding in more of the compressed signal during quieter sections. This can work really well, as it's often when the music is quietest that it benefits most from added detail.

Recordings of classical music typically have an incredibly wide dynamic range. In this example waveform (above), for instance, the audio level is generally very low, except for the explosive section about three-quarters of the way through. Sometimes the best way to reduce this dynamic range is to manually ride the fader on mixdown, or draw in volume automation to drop the level of the loudest parts by a few decibels. Once you've done this, you can make up lost gain by turning the signal up at the output stage.

Q How can I reduce hi-hat spill on my snare mic?

I've always had trouble with hi-hat bleed through my snare drum mic. I use a Shure SM57 on the snare, and if I try to boost the snare around 15kHz, the hi-hat stands out like a sore thumb. What's the best possible mic placement for the snare to reject the hi-hat as much as possible?
***SOS* Forum Post**

***SOS* contributor Mike Senior replies:**
This is a perennial problem, and is the reason I rarely bother recording a separate hi-hat mic most of the time — you've usually got too much hat in all the other mics already anyway! At the risk of stating the bleeding obvious, first have a word with the drummer, because it may well be easier for him to rebalance his sound a little than for you to faff about with mics and processing for hours.

In terms of reducing spill while recording, any cardioid-pattern mic, such as the SM57, in theory should reject the spill best if you aim the back of the mic directly at the hi-hat. However, the SM57 actually becomes something like a hypercardioid at high frequencies, so you'll need to experiment with angles a bit to get the best rejection in practice. If there's any way you can get the hi-hat further away from the snare that may also help, if only by giving ▶


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Q&A

some 192dB. That can be arranged to provide a lot of headroom above the 24-bit input signals, and equally a very low noise floor below them. In many cases, more complex calculations (such as for EQ) are done to higher resolution than 32 bits (so-called double or treble precision), but the result is then reduced back to 32 bits for onward processing. If properly engineered, this fixed-point approach works well in practice, but if you try hard you will eventually find the headroom limits!

The floating-point approach tackles the problem from a different direction, and I'm sure you'll remember the idea from school maths lessons. Instead of writing a large number as, say, 12,345,600,000,000, we can write is as 1.23456 x 10^13 (10 to the power of 13). The first part of the number (the 1.23456) is called the mantissa, and the second part (10^13) is called the exponent. The exponent describes how far to move the decimal point to get the full number — hence the term 'floating point'. The obvious advantage to us of this nomenclature is that it makes very big numbers much easier to write down, but it also makes the maths easier when multiplying numbers, as you have to do when changing the gain or volume of a digital signal.

In most practical applications, floating point is still used within a 32-bit framework, but with 24 bits allocated to the mantissa and eight bits allocated to the exponent. If you do the maths you'll find such an approach provides the utterly ludicrous theoretical dynamic range of 1500dB, which means you will never run out of headroom inside the processing, and never lose signal in noise floor.

So, as you found, you can increase or decrease the level to the most ridiculous extremes inside a floating-point system, and as long as you restore the gain to something more appropriate to feed the output converter's dynamic range, you will not suffer from the noise or clipping that a more conventional fixed-point or analogue system would. Which is very impressive.

Floating-point maths isn't quite the perfect audio saviour that it might appear, though, and really shouldn't be seen as a handy excuse not to bother with traditional gain-structuring practices. The mantissa is still restricted to 24 bits, which imposes some limitations on ultimate resolution when mixing multiple signals together, and that probably lies behind the raging arguments about quality differences when mixing 'inside the box' as opposed to outside. But I hope this somewhat long-winded explanation has quenched your thirst for an explanation as to why you can crank a signal up to +100dBFS and still get it back without it having been clipped.

Q Do I need a professional setup to record on my laptop?

I want to mic up my bodhran so that I can record what I am doing in Audacity, then play it back in order to hear where I'm going wrong, with the hope of improving my technique. One of the mics I've been recommended is an AKG C418 clip-on model, but I'm told it needs phantom power. What's this phantom power thing, and do you think I could achieve it without a fully professional recording setup?

I'm intending to record and play back from my PC laptop, but I can record to cassette tape if necessary. I also have a hi-fi amp, which I currently use to transfer the odd bit from vinyl and cassette into Audacity on my computer, should it be of use.

John Blackwell

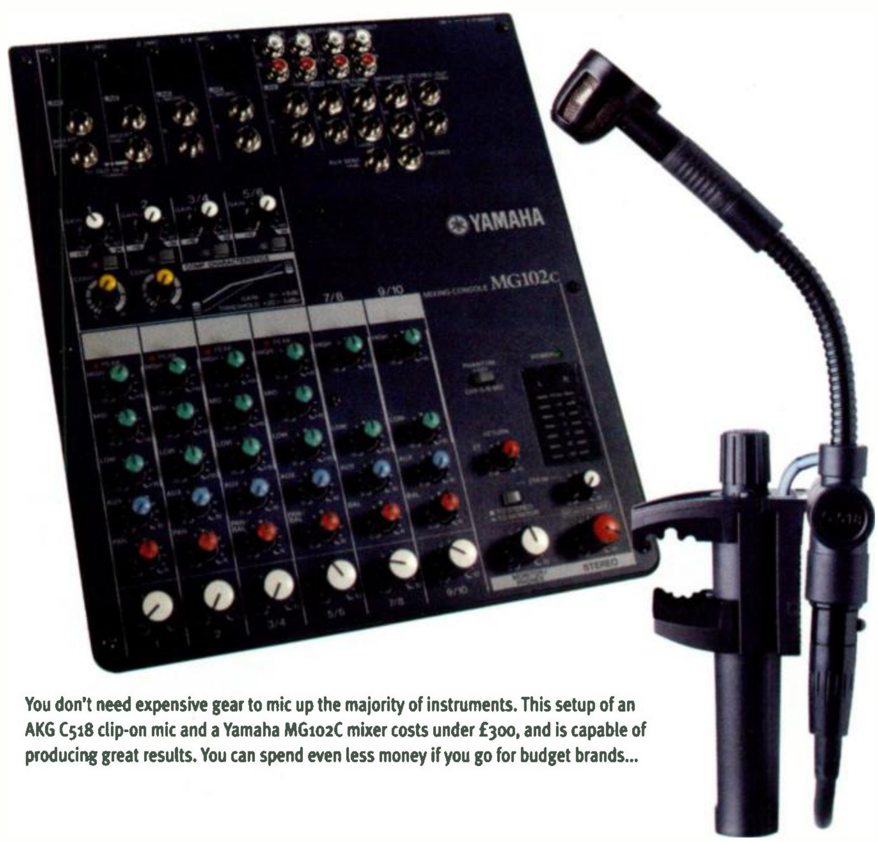
News Editor Chris Mayes-Wright replies: Phantom power is used to power specific types of microphones. Models such as the omnipresent Shure SM58 and other common handheld vocal mics don't need phantom power because they are dynamic — they operate like a speaker working in reverse, turning acoustic vibrations into electronic signals.

The AKG C418 (superseded by the new C518, pictured below) is a very capable condenser microphone that's often used for close-miking drum or brass instruments, due to its handy clip-on nature and gentle low-frequency roll-off. Its capsule (the bit that moves when sound waves hit it) is electrically charged, and requires a power supply to deliver this charge. So phantom power is sent from the mixer (or whatever the mic is plugged in to), down the microphone cable, and is used to polarise the capsule and also power other internal circuitry.

Phantom power is found on almost all modern mixing desks, professional or not. It's normally indicated by a button with '+48V' or 'phantom' next to it. The nominal supply voltage for phantom power is 48V, but some phantom power circuits only produce figures around the 30V mark.

To connect a mic to a phantom power supply, simply plug it in to the XLR inputs on the mixing desk, and switch the phantom power on. It's not good practice to leave phantom power on then plug the mic in, as this can cause surges when the cable makes an electrical contact with the microphone.

Hopefully that gives you a basic idea about what phantom power is. All you need now is some advice on how to acquire it! A cheap mixing desk is the obvious thing to go for, and you can get a usable one for under £50 and a more-than-half-decent one for around £70 — the Yamaha MG102C



You don't need expensive gear to mic up the majority of instruments. This setup of an AKG C518 clip-on mic and a Yamaha MG102C mixer costs under £300, and is capable of producing great results. You can spend even less money if you go for budget brands...


Roland
V-Synth GT

(pictured) was on sale for £69 at the time of going to press. At both of these price points, you'll get something that can be used for applications other than powering your AKG C418 (or C518, if you decide to purchase the new model), such as mixing three or four instruments and feeding their signal into a PA system. The cheapest mixers don't have phantom power, so make sure you get one that does! Any such mixer will have outputs that you can send to your cassette recorder, PC laptop, hi-fi amp, or main-stage PA system, for when you've perfected your playing technique!

If you just want to record into Audacity, and you don't need the other mixer paraphernalia, you can get an external audio interface — basically a soundcard that connects to your computer via USB or Firewire. If going down this route, make sure you choose an interface that has at least one microphone preamplifier and phantom power, for obvious reasons. All you have to do to record on to your computer is install the software drivers for the interface and make sure everything's connected correctly. Then, Audacity should be able to 'see' your incoming signal from the connected equipment. Interfaces cost anywhere from £40 to £1500, but you can get a decent two-channel model for just over £100.

As another alternative, you could get a single-channel microphone preampwith a line-level output that you could plug straight into your hi-fi amp or PC line input. Prices for these start at around £60.

Personally, I'd go for the first option I mentioned: a small mixer. It's the most flexible and cost-effective approach. SOS

Sound Advice

#7 How do I record small percussion instruments?

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

Hugh: Recording hand percussion is often more challenging than it would initially appear to be. Part of the problem is that often the sound pressure levels aren't that high — so close-miking appears to be sensible approach to take — yet a lot of physical movement is involved in playing the instrument in question, making distant mic placement seem like a better option.

Paul: Whatever you're recording — from balaphons to finger cymbals and thumb pianos — you will need a microphone that is able to deal with a very wide range of frequencies. As Hugh mentioned, some instruments are very quiet, so a capacitor microphone with a low self-noise figure (typically less than about 17dB EIN) would be a practical option in most cases.

Hugh: Agreed, percussion obviously involves a lot of fast transients, and the detail of the sound is conveyed by those transients, so a responsive microphone such as a capacitor is a must. But ribbon mics are enjoying renewed popularity, thanks to the new cheap components flooding the market. These tend to sound smoother and more natural than capacitor mics, without any resonant emphasis at the high end (which can be an issue with tambourines, for example). However, most have a figure-of-eight polar pattern, which will result in more room pickup than, for example, a cardioid mic. Some ribbons are designed with 'bright' and 'dark'-sounding sides, so some experimentation may be appropriate to see what complements the percussive sound best.

Paul: Deciding whether the sound of the room enhances or degrades the recorded signal is something that you will have to do after listening to what's coming from the mics. Generally speaking, domestic rooms tend to sound boxy and add little to the life of the sound. Therefore, you may be better off keeping the recording fairly dry and then adding ambient reverb (predominantly early reflections) when you mix. If you find that you're getting too much of the room sound in your recorded signal, you can place a broadband absorber, such as a commercially available filter or some thick duvets, behind the mic. If you're using Hugh's suggestion of a bi-polar ribbon mic, this will also help to negate the contribution from its rear lobe.

Hugh: Mic positioning varies from instrument to instrument, but my general rule for capturing a natural sound is not to bring the mic closer than the longest dimension of that part of the instruments that produces sound. In the case of a drum, this would be the head diameter, though as with all drums, you can mic them very close up if that produces a more useful sound, even though it may not be as accurate as miking from a greater distance. It's always best to search for the sweet spot, but as a fallback position, you can usually capture a decent sound by miking over the player's shoulder, providing the instrument sounds good to the player.

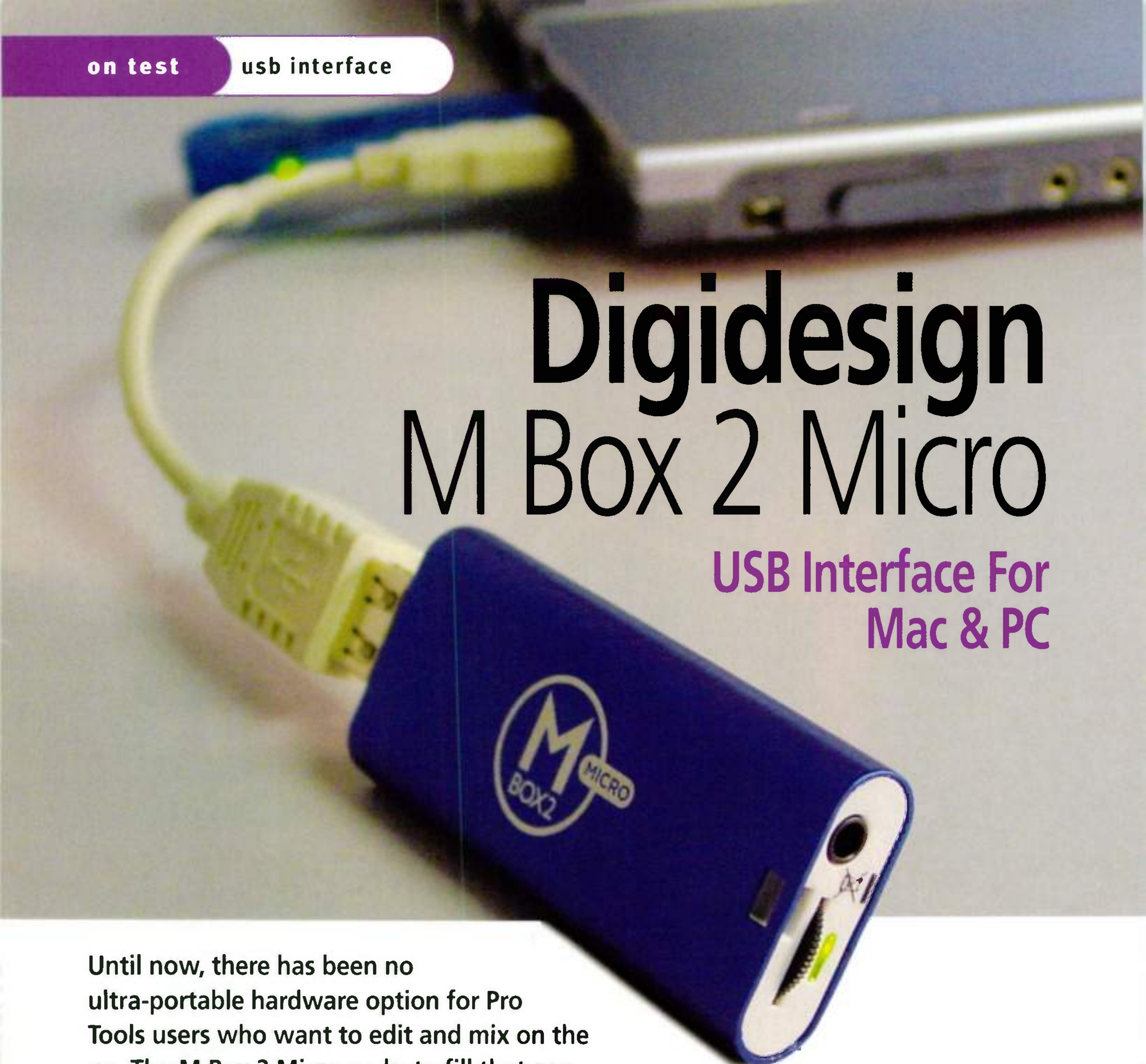
Paul: Once you're happy with the sound you're getting from your mic, there are various things to consider when actually recording it. Because of the inherent nature of most hand percussion, often involving loud and brief transients, ensure you record with a generous headroom margin — I would suggest at least 12dB. In many cases, there will be little low-frequency content, and filtering off the low end during recording can help reduce unwanted room colorations quite effectively.

Hugh: Absolute rhythmic accuracy is usually of prime importance with hand percussion, and if the

performer's abilities are limited (playing hand percussion accurately for a three-minute track is extremely difficult and tiring), then there isn't much shame in identifying a bar, or a couple of bars, that work well, and then copying and pasting them as necessary in to the track.

Paul: In the event that you record a percussion part that just doesn't stand out as we'll as you'd hoped, I recommend using an enhancer, my favourite being the Noveltech Character plug-in for TC Electronic's Powercore platform.


Focusrite



Digidesign M Box 2 Micro

USB Interface For Mac & PC

Until now, there has been no ultra-portable hardware option for Pro Tools users who want to edit and mix on the go. The M Box 2 Micro seeks to fill that gap.

Sam Inglis

Since the launch of the Digi 001 back in 1999, the LE range of project-studio equipment has become a major strand in Digidesign's Pro Tools universe. The LE range has grown steadily, encompassing everything from the simple M Box 2 Mini USB interface to the likes of the 003, which has enough I/O to record bands and multi-miked drum kits. Further options were opened up by the 2005 launch of Pro Tools M-Powered, which can work with any of a huge number of PCI, USB and Firewire interfaces from M-Audio.

However, one thing has always been lacking from the Pro Tools world. There are lots of people who want to use Pro Tools on a laptop for editing and mixing, but have no need to record audio in their portable rigs. To those users, even the smallest M Box interface represents an inconvenience, not to mention a drain on precious battery resources. I can't be the only Pro Tools user to have cast an envious eye in the direction of PCMCIA interfaces like the Echo Indigo, which add a high-quality, low-latency headphone output to your system — and nothing else. With no cables to plug in and no unwanted circuitry to drain your batteries, the only cost is a minimal increase in the footprint of your portable computer.

Mixing In The M Box

Well, Digidesign have finally produced an LE interface aimed at meeting these needs. The M Box 2 Micro has a refreshingly minimal ▶

SOUND ON SOUND

Digidesign M Box 2 Micro **£182**

pros

- Fills a long-standing gap in the Pro Tools LE product range.
- Headphone amp can go loud enough for use in any environment.

cons

- Limited to 44.1 and 48 kHz sample rates.
- Poor labelling risks accidental exposure to high sound pressure levels.
- Could be considered expensive for those who already own a copy of Pro Tools LE.

summary

The M Box 2 Micro is a product that many Digidesign users have been waiting a long time for, and as long as you don't need to work at high sample rates it offers a neat way of downsizing your Pro Tools rig for the road.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21 The Mall, Clifton Village Bristol BS8 4JG
Tel: 0117 946 7700
bristol@digitalvillage.co.uk
CAMBRIDGE
86 Mill Road, Cambridge Cambs. CB1 2AS Tel: 01223 316091
cambridge@digitalvillage.co.uk
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3 Kingsgate Centre St., Mary St. Southampton SO14 1NR
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DIGIDESIGN M BOX 2 MICRO

feature set: it provides a headphone output with a thumbwheel volume control, and that's it. The Micro is a lightweight metal device about the length of a cigarette and perhaps four times as wide, and connects to your computer via a USB 1.1 connection. You can plug the Micro directly into a USB socket, but I imagine most people will use it with the bundled extension lead. This is about eight inches long, and holds the Micro firmly enough that you're unlikely to disconnect the two accidentally in normal use. For the purposes of streamlining your portable rig, it's not quite as nice as card-based interfaces, which barely add to the profile of a computer, but it's understandable that Digidesign have taken this design route, as currently no card-slot format is sufficiently standard across all Mac and PC laptops.

"The Micro's robust output level will certainly be welcomed by those needing to work in noisy environments."

Like all M Boxes, the Micro acts as a dongle for the bundled Pro Tools LE software, but if you want to use any additional plug-ins, you will likely need an iLok key as well. In the review model, the accompanying DVD contained an installer for Pro Tools LE v7.3 and a separate updater to v7.3.1. However, the Micro will ship with v7.4 (reviewed elsewhere in this issue) as soon as it's available, and anyone who doesn't find 7.4 in the box will be entitled to a free upgrade.

Pro Tools LE was happy to live alongside Pro Tools M-Powered on my Windows machine, which made a nice change from the last time I tried this, several versions ago. It started up without mishap, recognised all my existing plug-ins, and generally worked exactly like you'd expect Pro Tools LE to work, although busy Sessions sometimes required higher buffer sizes to play back successfully with the LE/Micro combination than was the case in M-Powered with my M-Audio Firewire 1814. A graphical glitch which will be corrected by the v7.4 upgrade is that the I/O Setup and Hardware Setup dialogues appear to be talking to M Box Mini and not a Micro, but this caused no problems in use.

The Mighty Micro

I got an uncomfortable surprise the first time I used the Micro, because there are no markings by or on the thumbwheel volume control, and nothing in the printed documentation to show which way round it works. I left it in the default position, and it took me about a tenth of a second to rip my headphones off in agony, because this thing is loud. It seems ironic that the Micro comes with a leaflet that details the terrifying health and safety hazards posed by such things as inedequate ventilation in the studio, but says nothing about the risk of exposure to high SPLs. That said, the Micro's robust output level will certainly be welcomed by those needing to work in noisy environments.

In terms of sound quality, there is plenty of detail and mid-range punch on offer, although the Micro's headphone output was a little lacking in low-end weight with my Sony MDR 7509s. However, this is not an uncommon problem with headphones that have such a low (24Ω) impedance, and the Micro sounded fine with an 80Ω pair of Beyer DT250s, and other widely used headphone models.

96 Tears

Some of those Pro Tools users who have been asking for something like the Micro are working with 88 or 96 kHz Sessions in their HD rigs, and would love to be able to transfer them to a laptop for editing. Unfortunately, like Digi's other USB interfaces, the Micro is restricted to 44.1 and 48 kHz sample rates. It's true that USB 1.1 offers limited data bandwidth, but other manufacturers have managed to make output-only USB 1.1 devices that run at 96kHz — including Digidesign's sister company M-Audio.

I asked Digidesign about this, and they cited two reasons for the limitation. One is that, according to their own research, only 15 percent of existing Pro Tools users work at rates higher than 48kHz. The other is that the Micro's architecture and drivers are based on those of the existing M Boxes; support for higher rates would apparently have meant a complete redesign, and hence a much longer development period. Personally, I'd have thought that 15 percent of Digi's user base was a large enough cohort to be worth taking notice of, but I'm not running the company.

A related question is whether the Micro offers good value for money. Your £182 gets you a copy of the Pro Tools LE software as well as the Micro itself, which is fair enough if you don't already have it. However, as their own research suggests, Digi must be aware that the likely market for the Micro is greatest among existing Pro Tools users (after all, if it was your only Pro Tools interface, you wouldn't be able to do any audio recording). True, buying the Micro will save you the cost of upgrading your LE rig to the latest version 7.4, but even if you factor that in, it looks like a pricey buy for those who already have an LE system. For not very much more money, you could buy an M-Audio Transit USB interface and a copy of Pro Tools M-Powered, giving you 96kHz playback, an audio input and a digital output in a package not much larger than the Micro. You'd also be free to add other M-Audio hardware into the mix as and when you needed it, without forking out for another copy of Pro Tools. The cruel thing is that this approach is an option only for those who don't need the DV Toolkit — who are probably those most likely to want the 96kHz support.

Grumbles aside, though, the M Box 2 Micro does what it sets out to do. If you need to take your Pro Tools Sessions on the road for mixing and editing, it makes the footprint of your system significantly smaller. Job done. SOS

The Micro comes with a neat protective cover that should prevent damage to the USB connector in transit.

information

- £ £182 including VAT.
- T Digidesign UK +44 (0)1753 655999.
- F +44 (0)1753 658501.
- E infouk@digidesign.com
- W www.digidesign.com

Dual Compressor Limiter

Summit Audio DCL200

Summit Audio's high-end designs are known for their quality and longevity — so is their new compressor a classic in the making?

Hugh Robjohns

Californian manufacturers Summit Audio have a history in the pro-audio business dating back to 1979, when they started providing a brokerage service for high-end audio, but it wasn't until 1985 that the company released their first product — the TLA100 Tube Levelling Amp. I reviewed its smaller sibling, the TLA50, in *SOS* March 2003, but the TLA100 is still in production 20 years on!

The Summit Audio DCL200 reviewed here is a dual-channel compressor/limiter employing hybrid circuitry. It combines both valves (three screened 12AX7 double triodes) and solid-state technologies (principally Deane Jensen-designed 990 op-amp devices, complemented with a couple of OP297s and some traditional 5534s) to deliver reliability with a 'precise but warm' sound. Interestingly, Summit Audio use a regulated heater supply voltage for the valves, which they claim increases their useful life considerably, with more than 10,000 hours being recorded! The input and output stages, as well as all of the side-chain circuitry, are solid-state, while the signal amplification is performed by the valves.

The unit occupies 2U of rack space and extends roughly 10.5 inches behind the rack ears. The construction is to a very high standard, with a linear power supply fed from a transformer mounted on the right-hand side of the case. The majority of the circuitry is contained on a main PCB covering most of the floor area of the unit and carrying the three valves and four 990 op-amp blocks. The input and output XLRs and side-chain insert TRS sockets are cabled back to the main board, and two daughterboards mounted behind the front-panel controls carry the side-chain and gain-reduction circuitry. The mains input is via a standard fused IEC mains inlet, with a recessed switch to change mains voltage (115 or 230V).

SOUND ON SOUND

Summit Audio DCL200 £2345

pros
- Sublime sound quality.
- Solid construction.
- Hybrid circuitry gives best of all worlds.
- Easy to use.

cons
- Expensive.

summary

The DCL200 is a two-channel hybrid compressor that has vintage styling, but a modern sound. Tonally flattering and well-controlled, this is a high-end processor that is easy to use and sounds sublime.

Control Zone

The front panel of the DCL200 is a riot of vintage flair. The two channels' controls are arranged one above the other, starting with a toggle switch to bypass the channel's gain reduction (the signal circuitry remains in the path). Next up are five large, vintage-style rotary controls, with a blue, graduated legend calibrated simply 0-10. The manual explains that these arbitrary

Photo: Mike Cameron

markings are employed because the controls tend to interact with one another, and so the actual parameter values vary with the amount of gain reduction.

The input Gain, Threshold, Slope (ratio), Attack (roughly 0.1 to 100ms) and Release (about 35ms to 10 seconds) can all be adjusted with the continuous (rather than switched) and surprisingly light controls.

The Slope control allows the compression ratio to be varied between 1.1:1 and 7:1, always with a soft-knee transition. The actual gain-controlling element is described as a proprietary design and, as it is hidden away from view on the undersides of the daughterboards, I can't tell you anything more about it. However, the side-chain is of a peak-detecting type (rather than an RMS-level design), and it sounds VCA-based rather than optical to my ears.

To the right, a pair of modern back-lit rectangular VU meters each have an associated red Overload LED and a toggle switch to display either the output signal level or the amount of gain reduction being applied. The Overload LED means exactly that — it illuminates when the signal is within 0.5dB of clipping — while the 0VU mark equates to +4dBu when the meter is switched to show output level.

Finally, there is a 'jewelled' On lamp and associated mains-power toggle switch, plus a Link toggle switch to gang the two channels for stereo operation. In this mode the Threshold, Slope, Attack and Release controls of the ▶


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Neyrinck Mix 51

Surround Panning System For Pro Tools LE

You can't mix in surround on Pro Tools without shelling out for an HD rig. Or can you?

Mike Thornton

One of the major limitations of the current native versions of Pro Tools, LE and M-Powered, is that there is no support for surround formats or mixing at all. However, Paul Neyrinck has come up with a plug-in suite which has been designed to get round these limitations of working in surround on LE systems. It works by exploiting a relatively new feature in the Pro Tools format that was designed to furnish additional I/O from instrument plug-ins.

Mix 51 will work on any HD, LE or M-Powered system running Pro Tools 7.0 or later, but Neyrinck advise using at least v7.2 to get the full benefit. They also remind users that as LE systems are host-based, Mix 51 will introduce additional latency into your Session. The amount of latency is dependent on the H/W Buffer Size setting; for a typical setting of 512 samples the latency will be 10.6 milliseconds. To work in surround on an LE system you will also need to have an interface with at least six outputs, such as the M Box 2 Pro, 002(R) or 003(R).

Overview

Mix 51 is made up of two main components. First, there are Mono and Stereo Panner plug-ins that can be placed on track inserts to pan, mix and route audio to the Mix 51 Surround Mixer buses. Second, the Mix 51 Surround Mixer plug-in provides 30 channels of outputs, grouped as three 5.1 main buses and three quad auxiliary buses. The idea is that the additional 5.1 buses can be used as subgroups to manage, say, dialogue, music and sound effects, while the quad buses are designed to handle quad reverb sends, as LE doesn't support surround-format sends.

In order to work in surround in LE you will need to spend some time in the I/O Setup window. Because Pro Tools LE can only address stereo output buses, you have to allocate three stereo outputs to connect to your 5.1 monitoring system. Neyrinck recommend you adopt the SMPTE order of Front Left, Front Right, Centre, LFE, Rear Left and Rear Right. You should also add the relevant mono sub-paths as well, so that your I/O Setup looks like the one you can see in the screen on the opposite page.

Mixer & Panner

Surround Mixer is an RTAS plug-in which can be placed on any mono or stereo track. A single instance of this plug-in manages all Mix 51 surround inputs and outputs in a Pro Tools Session, so it is not necessary to have more than one Surround Mixer in any one Session. Because its inputs come from Surround Panner plug-ins and its outputs appear as Pro Tools track inputs, it is not critical where you place the Surround Mixer plug-in, as it does not process any audio on the track where it is located. The audio is simply passed through from input to output. The mixer is fairly self-explanatory, and gives you metering plus volume, solo and mute controls for each of the three 5.1 mix buses and quad auxiliary buses.

The Surround Panner plug-in operates as an RTAS plug-in on mono and stereo tracks. It will simultaneously pan to one of the 5.1 main buses and to one of the quad effects send buses, and provides panning, volume, mute, solo, LFE and divergence controls. By default, the Mix 51 Panner does not output any audio to the normal Pro Tools mixer, but hitting the Bypass button will pass the signal to the Pro Tools mixer as well, so you can set up an additional stereo mix while

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SOUND ON SOUND

Neyrinck Mix 51 **$189**

pros

- It works!
- It is a unique solution, generally well implemented and very cost effective.
- Excellent demo session included.

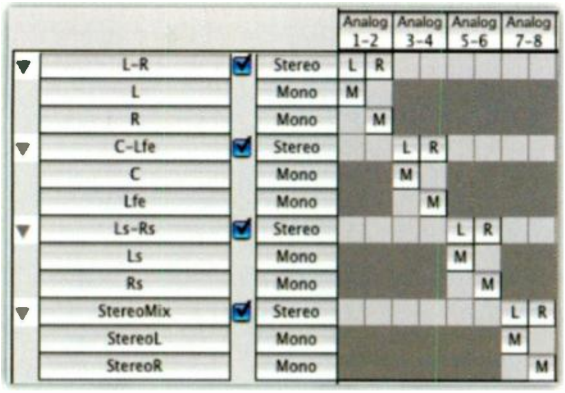
cons

- Poorly designed stereo panner.
- Panners are pre-fade.

summary

Mix 51 is an excellent idea that adds very usable surround mixing to LE and M-Powered Pro Tools systems.

To use Mix 51, you need to set up stereo paths and mono sub-paths in the Pro Tools I/O Setup window for the various physical outputs.



	Format	Analog 1-2		Analog 3-4		Analog 5-6		Analog 7-8	
▼ L-R	Stereo	L	R						
L	Mono	M							
R	Mono		M						
▼ C-Lfe	Stereo			L	R				
C	Mono			M					
Lfe	Mono				M				
▼ Ls-Rs	Stereo					L	R		
Ls	Mono					M			
Rs	Mono						M		
▼ StereoMix	Stereo							L	R
StereoL	Mono							M	
StereoR	Mono								M

also continuing normal Mix 51 operation. It is even possible to have multiple panners on a single track, so you could route a track to two different Mix 51 main or send buses.

An additional LFE Send plug-in operates as an RTAS plug-in in mono, with a multi-mono version included for stereo tracks. Its purpose is to allow you to send a track directly to the LFE channel without having to send the signal into the main L/C/R/Ls/Rs outputs. Like the Panner, the Mix 51 LFE Send plug-in does not route any audio to the normal Pro Tools mixer unless you hit Bypass, whereupon LFE Send will route the track to the Pro Tools mixer as well as the LFE bus.

Setting Up

The next step is to create enough Aux tracks in your Pro Tools Session to return the individual outputs from the Surround Mixer plug-in and route them to the interface outputs. You will need three stereo Aux tracks for each plug-in 5.1 buss you want to use, and a further two stereo Aux tracks per send bus. You can then insert your preferred stereo reverb plug-in in both channels to create a pseudo-surround reverb. Once all these Aux tracks have been set up, route their outputs to the appropriate interface outputs as set from your I/O Setup window.

The Surround Mixer plug-in can go on any track in your Session apart from one of these Aux tracks, as a plug-in cannot route audio to a track it is on. The best place to put the plug-in is on one of your normal 'content' tracks. It doesn't matter if it is a mono or stereo track.

Then you turn your attention to the input routing. Once you have the Surround Mixer plug-in in your Session, when you click on the Input Routing button for one of your Aux tracks, you will see that there is now an additional 'plug-in' option available to you. From here, you can select the appropriate ▶

Maximum Migration

The Mix 51 system has been designed to be compatible with Pro Tools HD, so that surround mixes originated in Mix 51 and LE can easily be migrated to an HD system. Sessions that use Mix 51 will open and work fine in Pro Tools HD, as long as Mix 51 is installed there too, but you can also migrate a Mix 51-based Session's automation tracks to Pro Tools HD channel panning and mixing automation tracks by using the Special Copy and Paste Automation features in Pro Tools, as Mix 51's pan and volume controls have been designed to use the same panning and volume tapers as those in Pro Tools HD.

bus to pick up from the Surround Mixer plug-in. Repeat this with all the Aux tracks, selecting the appropriate buses as required. You can then hide the Aux tracks so they don't unduly clutter up your Session.

You can now add Panner plug-ins to your 'content' tracks. Within the plug-in, you can select which of the Surround Mixer's multi-channel buses that track will be sent to.

Surround Panning

The layout and operation of the Mix 51 Surround Panner is very similar to that of the Digidesign Surround Panner. The basic panning tool is a conventional two-dimensional panner of the kind that is often called an X/Y panner. You can either click on the orange '+' control and drag it around to perform panning moves, or use the sliders on the bottom and side of the panning grid to position the output of that track in your 5.1 surround sound stage. Separate controls set how much of the output of the panner will be routed to the Centre and LFE channels.

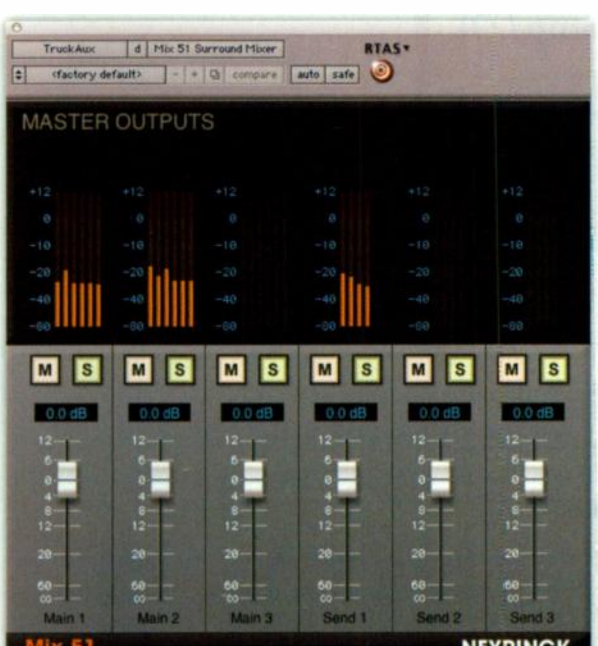
The Front, Rear and Front/Rear Divergence controls allow you to bleed some of the signal into other channels without having to pull the panner into the middle of the grid. When the Divergence controls are set to 100 they are, in essence, off, with no bleed, while 0 means 'send the signal everywhere'. I am not a particular fan of Divergence on any surround panner — if I want to route the signal everywhere, I pan it centrally!

I was disappointed to find that the stereo version of the Surround Panner is basically two mono panners put together, sharing only a common routing and fader section. This isn't too much of a problem for setting static positioning, but it makes steering a stereo track around the surround sound stage very much more difficult, as you cannot grab both orange '+' icons and move them simultaneously. I would have preferred a version of the stereo panner that had one set of steering controls, as in the Waves 360 panner.

The stereo Surround Panner is, in most respects, a dual-mono design, and doesn't allow you to pan both channels with a single movement.

Track Faders & Panners

A hurdle with the whole Mix 51 system is that all plug-ins in the Pro Tools mixer are inserted before the track fader, with the exception of Master Fader tracks, so the track fader potentially becomes redundant in the surround mix. There are two ways of dealing with this. One is to use the fader in the plug-in instead; the down side of this is that you don't see the mixer's fader move as you would normally. The other is to route each track to an Aux track, and insert the Panner plug-ins on these Auxes rather than on the source tracks. This, of course, means adding yet more channels to your mixer, and you end up with the fader and the plug-in on separate tracks. We cannot blame Neyrinck for this limitation, as it is inherent in the design of the Pro Tools mixer, but it does make for a messy solution in this particular application.



The Surround Mixer plug-in can address up to three main 5.1 and three quad send busses.

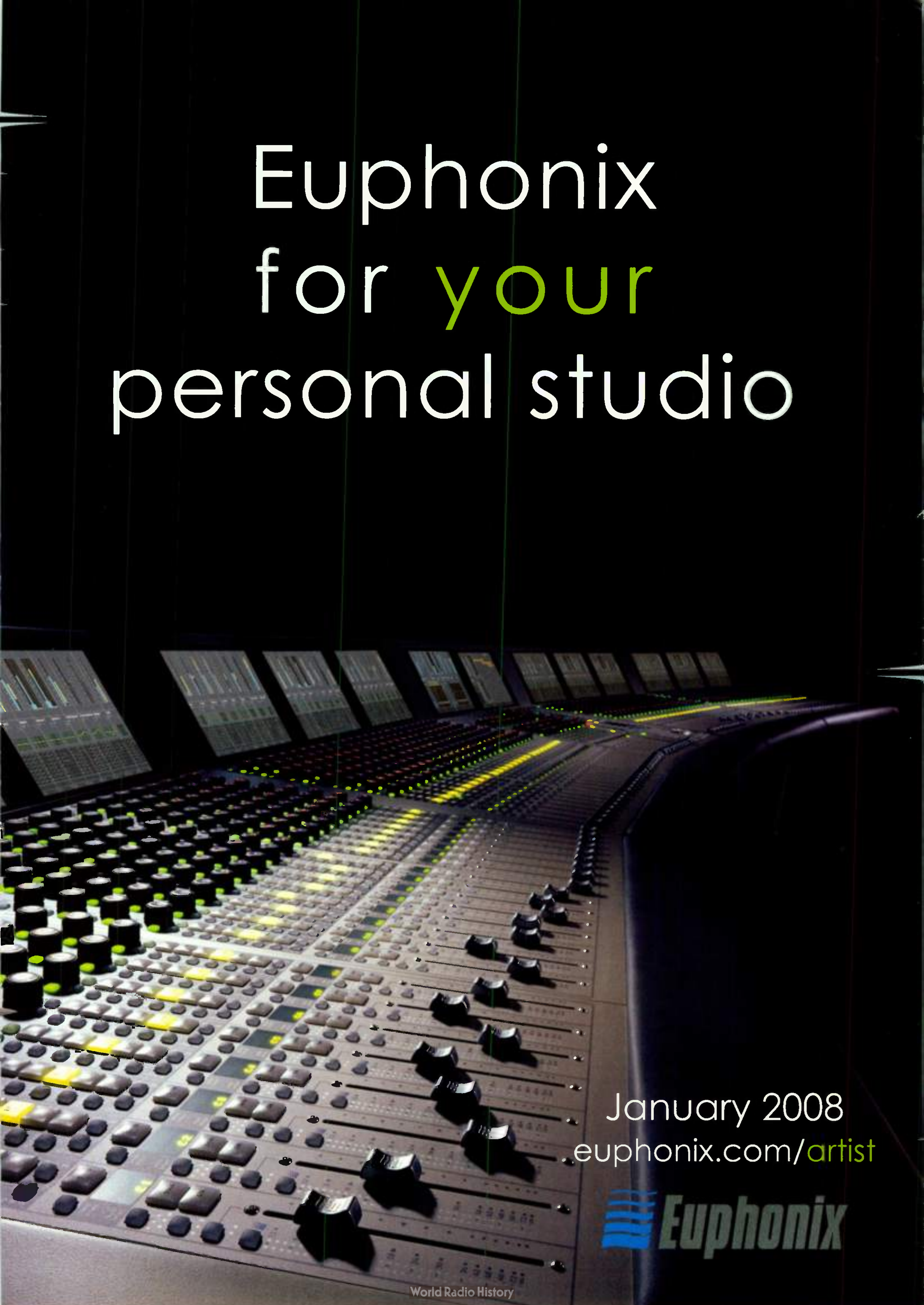
Conclusion

Paul Neyrinck should be commended for producing an excellent 'out of the box' implementation of surround mixing in LE and M-Powered systems. Mix 51 is a very workable solution that enables LE users to create and mix surround content. The idea of bypassing the conventional Pro Tools mixer routing does take a little getting used to, but the fact that multiple surround and send buses are built into the Mix 51 structure makes surround bussing and mixing on LE systems as easy as possible. There is an excellent demo session included in the installation, which demonstrates how to use Mix 51 and is well worth going through before you start using it in anger. My only disappointment in the design is the implementation of the stereo panner, where a single-steering option would make all the difference. SOS

information

£ $189

W www.neyrinck.com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Music Lab Real Strat

Virtual Guitar Instrument

Music Lab impressed us with the playability and sound quality of their virtual acoustic guitar instrument, Real Guitar — and now they've gone electric...

Nick Magnus

Before the virtual instrument revolution, producing convincing keyboard-generated guitar parts was a rather hit-and-miss affair. Although it was possible to achieve some moderately passable acoustic and electric 'lead guitar' performances, given a decent source of sampled raw material and some appropriate outboard processing, it was usually at the expense of the finer details; those 'guitaristic' articulations and techniques that add an authentic feel of spontaneity and human interactivity. Altogether much harder to emulate were convincing strummed guitar parts. Two hardware MIDI products from the 1990s, Oberheim's Strummer and Charlie Labs' strap-on Digitar, made a brave stab at the job by analysing any chord presented at their MIDI input and producing a 'strummed' MIDI output, in an appropriate guitar voicing, to drive a target sound source. Of these, the Digitar allowed for true real-time strumming and was the more successful of the two in terms of realism; nevertheless the dark circles under my eyes still remain, testifying to the many editing hours spent bullying Digitar parts into submission. Yet even after all that work, they ended up being buried in the mix to protect their patently artificial nature from detailed examination!

Steinberg provided a groundbreaking solution in 2002 with the release of Virtual Guitarist, a software instrument based on time-sliced, sampled loops of real strummed acoustic and electric guitar performances that could sit prominently in a mix. The greatly expanded and enhanced Virtual Guitarist 2 followed in 2006. Virtual Guitarist 2 is nevertheless based upon a supplied library of rhythm styles which, despite being editable and customisable in a DAW, do not allow for real-time strumming performances.

Russian company Music Lab, in collaboration with Best Service, then raised the bar in 2004 with the first release of Real Guitar, the brainchild of Sergey Egorov. (For a more detailed low-down, see the head-to-head reviews of Real Guitar 2L and

Virtual Guitarist 2 in the September 2006 issue of *SOS*.) Taking a different approach to Virtual Guitarist, Real Guitar is exclusively devoted to acoustic guitars, using discrete single-note multisamples taken at multiple velocities, driven by a dedicated engine that employs MIDI processing not entirely dissimilar to that found on Charlie Labs' Digitar. Chords played on a MIDI keyboard are re-interpreted to produce authentic guitar voicings which can then be 'strummed' in real time, using groups of trigger keys elsewhere on the keyboard. However, Real Guitar goes much further than that, offering a fully polyphonic Solo mode and four different Chordal modes, variously utilising numerous user-controllable 'guitar performance' tricks such as fret-slides, hammer-ons and tremolando effects, not to mention keyswitchable alternate articulations such as mutes, palm slaps and harmonics. At last, a highly convincing and playable 'acoustic guitar' that could be featured loudly and proudly in a mix without a hint of embarrassment or apology. Users fast became fans, and were almost immediately asking "will there be an electric guitar version?"

Enter Real Strat

It's a reasonable assumption that in deciding to develop Real Strat, as opposed to 'Real Les Paul' or 'Real Tele', Sergey Egorov settled upon that particular guitar as being a quintessentially iconic, versatile and ubiquitous example of the genre. Unlike Real Guitar, which provides eight different acoustic guitars, Real Strat currently offers only the one sample set, although we'll have to see whether this is augmented in the future with Real Strat-hosted add-on guitar expansion packs (with alternative GUI 'skins' that match specific guitars?), or perhaps Real Strat is just the first of an ongoing series of 'Real' electric guitar virtual instruments.

Real Strat requires a VST/DXi host for PC, or a VST/AU host for Mac, and RTAS support is also available for Pro Tools 6/7 users with FXpansion's VST-to-RTAS Adaptor (which is available or both Mac and PC). A stand-alone version is also installed automatically. During installation, the Real Strat Bank Manager applet asks you to choose a sample rate for the core library appropriate to your usual working environment: six sample rates are offered, from 44.1kHz all the way up to 192kHz. I installed the 44.1kHz version, which occupies 892MB of disk space; if, however, you subsequently wish to change your DAW's sample rate you will have to run the Bank Manager applet again to re-install the library of the corresponding sample rate. Once the core library is copied over to your hard drive of choice, Real Strat is ready to rock in time-limited demo mode; to fully activate the product, simply apply for an authorisation code via email, and this will be returned in the same way.

Beneath The Scratchplate

Anyone familiar with Real Guitar will feel immediately comfortable with the Real Strat interface, as the two have much in common. Real Strat occupies around 15 percent more screen space than Real Guitar, due to the virtual keyboard and additional functions required by Real Strat's Solo mode, which goes into considerably more detail than that of Real Guitar.

The GUI is divided into four areas of interest. Across the centre lies the fretboard, upon which green dots appear when Real Strat is played, to indicate which 'strings' are 'sounding'. To the right is the pick-position selector, which can be placed in any of 15 positions between the neck and bridge, providing a useful range of tonal variation, and making up, in part, for the lack of a pickup selector.

Above the fretboard, on and adjacent to the guitar body, are a number of controls that are always visible, regardless of performance mode. Strum sets the base strum speed (of chords or any simultaneously played notes) for the whole instrument. This can be

SOUND ON SOUND

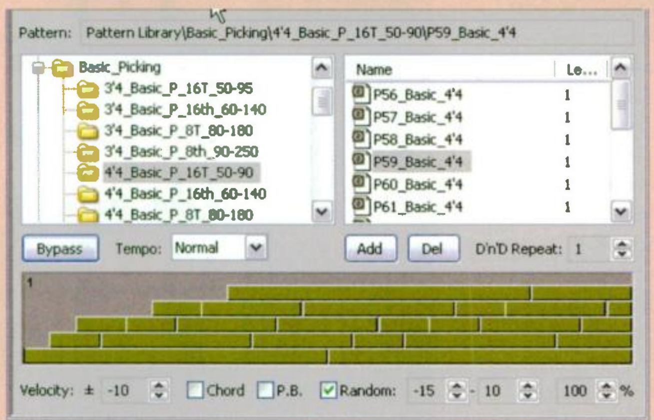
Music Lab Real Strat £133

pros
- Stunningly vibrant, articulate and realistic.
- Low CPU consumption.
- Modest RAM requirements.

cons
- None, except 'more guitar models, please!'

summary
Once you get to grips with the numerous and ingenious ways that guitaristic playing techniques can be applied to a performance, Real Strat really can produce believable results that might otherwise be impossible to achieve without using a highly-complex, dedicated sample library many gigabytes in size — or a real guitarist.

Pattern Manager



selected Pattern display. Once a Pattern is chosen, simply play a chord on the keyboard and the Pattern plays in tempo sync with the host DAW. Incorporating these within a sequence is simplicity itself; just drag a Pattern from the lower display onto a MIDI track and it appears as one bar of MIDI trigger key data that can be copied as many times as required. Being MIDI data, it can also be edited, so applying different grooves and quantise settings is totally possible. The Pattern data does not include chord information, which you add on a separate MIDI track, making sure both tracks' outputs are routed to Real Strat. It's instant, ready-made accompaniment and a potential time saver. However, the pleasures of playing Real Strat are so great that I would opt to 'roll my own' every time!

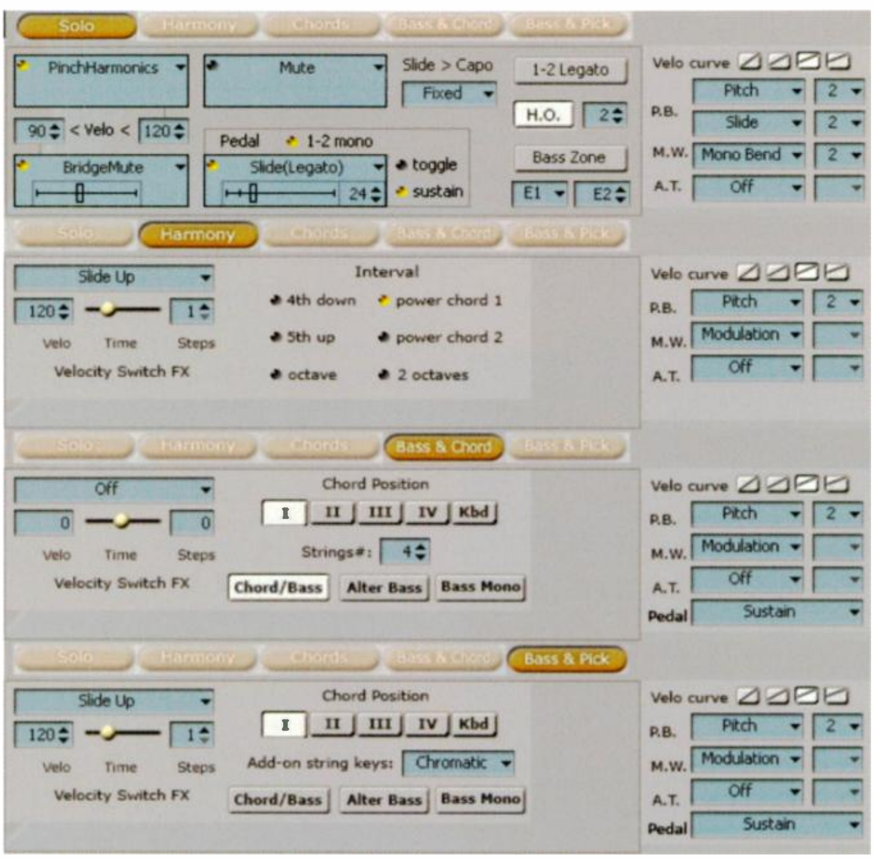
Featured on Real Guitar and carried over to Real Strat, Pattern Manager contains a sizeable library of pre-programmed rhythms and picking styles, 1250 in all, and is a derivation of Music Lab's earlier Rhythm'n'Chords MIDI plug-in. These are categorised according to tempo range, meter and playing technique and cover everything from basic picking and strumming to blues, jazz, funk, reggae and world styles, amongst others.

The PM button opens the Pattern Manager window which is divided into three panes: the folder browser, the file browser and the currently

modified (as can most Real Strat parameters) with a MIDI controller, and also overridden by longer Slow Strums whenever certain definable conditions are met. Attack has the effect of time-stretching or shrinking the plectrum noise, which naturally affects the apparent latency of the instrument. The default setting of 20 percent seems most effective; a setting of zero, while producing the fastest response, seems to detract something from the sound's 'physicality'. Release affects the rate at which the strings are damped, as you'd expect. The default of 100 percent is fine for most tasks although fast, über-metal-style passages or trills do benefit from shorter settings for cleaner, smudge-free results, especially when using high amplifier overdrive settings.

Part of the realism behind Real Strat's sound is the Floating Fret Position, which imitates the way a guitarist changes playing position on the neck. This is indicated by a 'capo' on the fretboard which automatically follows your movements up and down the keyboard. In Solo and Harmony modes, the button labelled 'Auto' lets you enable or disable this feature. If disabled, you can 'lock' the capo's position by right-clicking on the fretboard, whereupon the top five strings will only play samples above the capo position. The three Chord modes address this differently, as explained later. Two Accent Hi/Lo sliders vary the velocity threshold at which the three velocity layers will trigger, effectively extending or reducing the velocity range over which a specific dynamic layer will play. Like Real Guitar, Real Strat also features a full-time round-robin system that alternates samples for repeated notes. The Alter box offers five choices, the minimum representing three alternating samples, and the maximum being 10. This totally eradicates any hint of the dreaded 'machine-gun' effect, especially when playing tremolando or fast, *Reservoir Dogs*-style passages. In all modes but Solo mode, the Hold button substitutes the sustain pedal — in other words, all chords sound for their full duration until they either fade out naturally, or you play a new chord or one of the Mute trigger keys. In Solo mode, Hold works only while at least one key is kept held down, whereupon any subsequent notes will sustain until all keys are released.

At the top of the interface are two groups of drop-down menus. The left-hand group handles output level, EQ, tuning, modulation and general instrument setup parameters. Here you can also choose whether Real Strat will add pitch-bend and modulation to all notes, or only those keys that are currently pressed. The latter option is the default, and is the most naturalistic, as it allows you to bend specific held notes within a chord while the rest are ringing via the sustain pedal. In the right-hand group,



This composite picture shows the performance control options of the various playing modes (Chords Mode is shown in the main plug-in screenshot).

the Mixer allows you to balance fret noise, release noise, pick noise, mutes, slow-strum and velocity-switched effects against the main body sound, while the FX Mixer offers further level control of bridge mutes, harmonics, pinch harmonics, slaps and scrapes. Also found here are settings for Real Strat's own built-in wah-wah effect. This can either be set to respond automatically to your playing dynamics (like Electro-Harmonix's Doctor Q stomp box) with a choice of positive or negative sweeps; to auto-wah according to the set modulation rate; or be controlled manually via a MIDI controller. If you have a continuous MIDI footpedal that can be assigned to this task, so much the better.

The lower part of the GUI has two areas: one that contains the various performance control options (these change depending on which playing mode is active) and the other a virtual keyboard that shows the range of playable notes in the Main zone (see below), as well as displaying which keys are currently being played. The performance control display is the 'nerve centre' of Real Strat; an examination of its options for each of the various playing modes will follow shortly.

Basic Performance Technique

The MIDI keyboard connected to Real Strat is divided into three zones: the Main playing zone covers E1 to B4 and there are two Repeat zones above and below the Main zone covering C0 to D#1 and C5 to C7. The playing technique (particularly for the Chordal modes) essentially involves playing notes or chords in the Main zone, and repeating them (i.e Strumming) using the Repeat zone keys, although the exact technique differs somewhat depending on Real Strat's playing mode. The Repeat keys are subdivided into two tasks: white keys repeat the full sound (in Chordal modes, neighbouring white keys alternate between up and down strums) while the black keys play muted versions of the same notes.

Solo Mode

As its name suggests, Solo mode allows for fully polyphonic, freestyle playing of single lines, arpeggios, chords or whatever takes your fancy. This features the most detailed set of control options, enabling a vast array of different articulations, noises and guitaristic shenanigans to be activated in various ways. Of the four larger blue boxes shown in the top-left corner of the screen to the left, the left-hand pair govern velocity-switchable articulations and effects. These are selectable from drop-down menus, with independent velocity thresholds for low- and high-velocity effects. When the yellow LEDs are on, these are active; when off, their assigned functions are ignored.

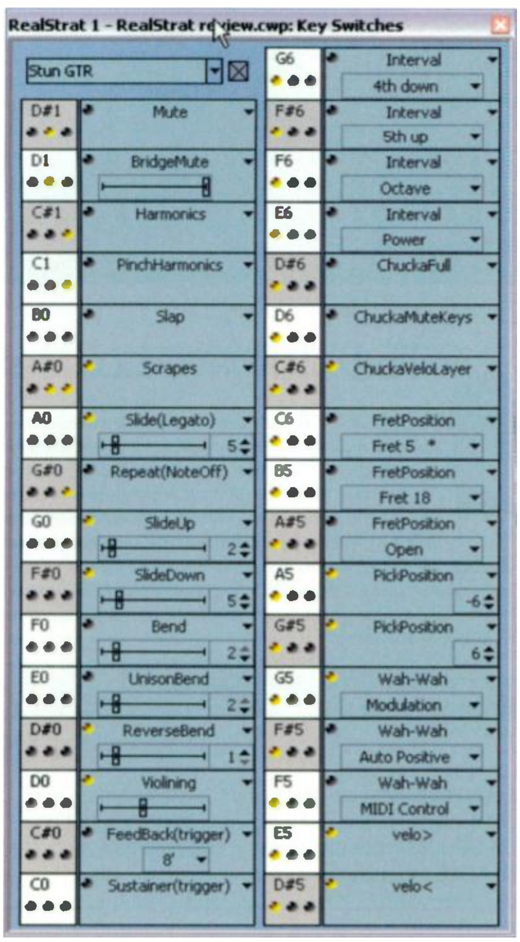
The large box to the lower right offers a substantial list of effects that can be engaged using the sustain pedal; these can either be momentary or latchable, toggling on and off with alternate pedal presses. Sustain itself can be enabled or disabled along with these effects if desired. The upper right-hand box offers a selection of alternative articulations which engage permanently when the box is turned on, and which ignore any velocity-switching settings.

Hammer-ons and legatos are well catered for too; Legato offers smooth note transitions over a two-semitone range, and is very effective for ensuring that two adjacent notes played on the same 'string' don't run across each other. Hammer-ons also include automatic pull-offs, their operational range being between one anOd 12 semitones. ▶

A separately definable Bass Zone can be toggled on or off, allowing notes within that zone to ignore velocity-switched effects and mute trigger keys, enabling notes within the zone to continue sounding while notes outside the zone respond to all the set conditions.

In addition to these, various functions for the pitch-bender, mod wheel and aftertouch can be selected, with operational ranges for each. Solo mode allows different functions and ranges for upward and downward pitch-bend movements, so you could have, for example, smooth upward whole-tone pitch-bend and chromatic downward 'fret slides' over five semitones — very cool. Included amongst the pitch-bend options is MonoBend; this bends only the lowest of two or more notes, an effect otherwise known as Unison bend. On discovering this option, I found that uncannily authentic renditions of 'Honky Tonk Women' and 'Hocus Pocus' slipped out before I could stop myself! Another nice touch is that the pitch-bend range can be set to half, quarter or even one eighth of a semitone, all especially useful for performing ultra-controllable real-time vibrato using a pitch-bend lever.

If, even after all this, you're running out of ways to add more articulations, Real Strat keeps on throwing them at you. In Solo mode, the entire range of articulations is available to you via keyswitches. The KS button on the lower far left opens a separate window, listing 33 possible keyswitches (as shown in the screen below) operating across two ranges, C0 to D#1 and D#5 to D6. Each keyswitch has a drop-down menu to select an articulation or effect, and each one can be individually enabled or disabled. Three LED switches to the left of each keyswitch determine whether that particular effect will be momentary or togglable, have sustain (hold) added or simultaneously function as a normal Repeat key. Thoughtfully, Real Strat allows any keyswitch setup to be saved as a preset, so even the most involved setups can be easily recalled. By now you're probably wondering what these various articulations are. The list is too long to detail in its entirety, but a glance at the keyswitch screenshot on the left shows the vast majority. Slaps, bridge mutes and harmonics are here of course, along with violining (swells), tempo-synced trills and tremolandos, pinch harmonics and chucka-wah noises. You can even add feedback, at any of six selectable pitches, at the press of a trigger key! The intriguingly named Sustainer extends the length of held notes by overlaying an additional swelled version of the same note each time the trigger key is pressed. The Scrapes articulation is actually a complete multisampled collection of one-shot effects including squeaks, squeals, wibbles, scribbles, divebombs, plectrum scrapes and general full-shred guitar mayhem that add a genuine sense of grunge and attitude — barking mad and brilliant! If there's an articulation not included here, you probably don't need it.



Amplitube 2 Duo

Real Strat comes bundled with Amplitube 2 Duo, a cut-down version of IK Multimedia's Amplitube 2 Guitar Amplifier simulation plug-in. This limited version provides two Amps, two Cabinets, two Microphones and two Stomp Boxes, compared to the 14 Amps, 16 Cabinets, six Microphones, 21 Stomp Boxes and 11 Rack Effects of the full version. An amplifier simulation is, of course, an essential addition to an instrument of this sort, and Amplitube 2 Duo comes as a welcome bonus for anyone lacking in this department.

Judgements on the quality of guitar-amp sounds are bound to be subjective. While the two amp models supplied seem competent enough at the more bluesy or clean end of the scale, I struggled to obtain anything approaching the creamy-smooth leads that an (admittedly non-guitarist) ageing progger like me might gravitate towards. Having said that, the full list of amp simulations and other extras in the full version of Amplitube 2 may well contain the missing ingredients, and a reduced-price upgrade to the full version is available.

Those on a shoestring budget might like to check out the growing number of freeware amp simulators on the net. Two of my faves are Voxengo's Boogex (www.voxengo.com) and BTE Juicy 77 (www.bteaudio.com), both quite different, but producing a range of tones between them that complement Real Strat very well.

Chord Mode

Identical to the mode of the same name in Real Guitar, Chord Mode is the place to come when you want to strum. Real Strat can detect 26 different chord types, the name of the current chord being displayed just above the fretboard. As hinted at earlier, the Floating Fret (neck position) behaviour is slightly different to Solo mode: it can either be set at one of four fixed positions or set to track your keyboard position. The capo does not (visibly) track the keyboard as you play, but you can manually position it by right-clicking the fretboard on any fret to override the current chord position. The capo's position can also be moved using a MIDI controller, making for a very flexible arrangement.

The two number boxes named 'Strings' allow you to restrict the number of strings sounding, so, for example, an upper setting of one and a lower setting of four will only allow the upper four strings to play — invaluable for avoiding the muddiness of full, overdriven six-note chords in a busy mix. A switchable Chord/Bass option enables major- and minor-triad chords to be rooted by any bass note; for example, a chord of Bb-C-E-G sounds like a C chord over a Bb bass note, rather than being interpreted as a C7 with the Bb at the top. One velocity-switchable effect can be assigned from a choice of slow strums, slides up or slides down. Every setting here can be altered using MIDI controllers, so many subtle variations can be programmed with precision into a sequencer.

Other Modes

Bass & Chord Mode is similar to Chord Mode, but in this case the C5 and D5 trigger keys play the root and fifth (or occasionally the third) of the chord, while only the top four strings are strummed (or fewer, if you alter the Strings# value. The Bass Mono setting prevents the root and fifth bass notes from over-running each other, which lends itself to tidier results. This is the perfect mode for country and western stylings or that wedding party version of 'Mull Of Kintyre'.

Bass & Pick Mode has a performance control panel that's nearly identical to that of Bass & Chord Mode, but requires a completely different playing technique to the other Chordal modes. Here, the six trigger keys C5 to A5 each trigger one of six 'virtual strings'. While holding a chord in the Main zone, the six trigger keys are played in a finger-picking style, just as if they were the actual guitar strings. The Add-on String Keys selector box determines the function of the black trigger keys from C#5 to A#5. These can play mutes, as in the other playing modes, or, alternatively, if you select Unison they can duplicate the 'full' note that is one semitone above, facilitating easy performance of tremolando on one string. Even more interesting is the Chromatic setting, whereby the black notes C#5 to G#5 sound one semitone down from the next-highest white note (A#5 to C6 move progressively one semitone higher), leading to some very pleasing and often serendipitous chord voicings, without changing chord shape in the left hand. Dreamy, chorused Genesis-inspired arpeggios, anyone?

And finally, Harmony Mode. This is Real Strat's simplest mode, and is essentially a one-finger power-chord generator. Six preset power-chord intervals are provided, together with the option of velocity switchable upward or downward slides with configurable velocity threshold, slide speed and range.

Conclusion

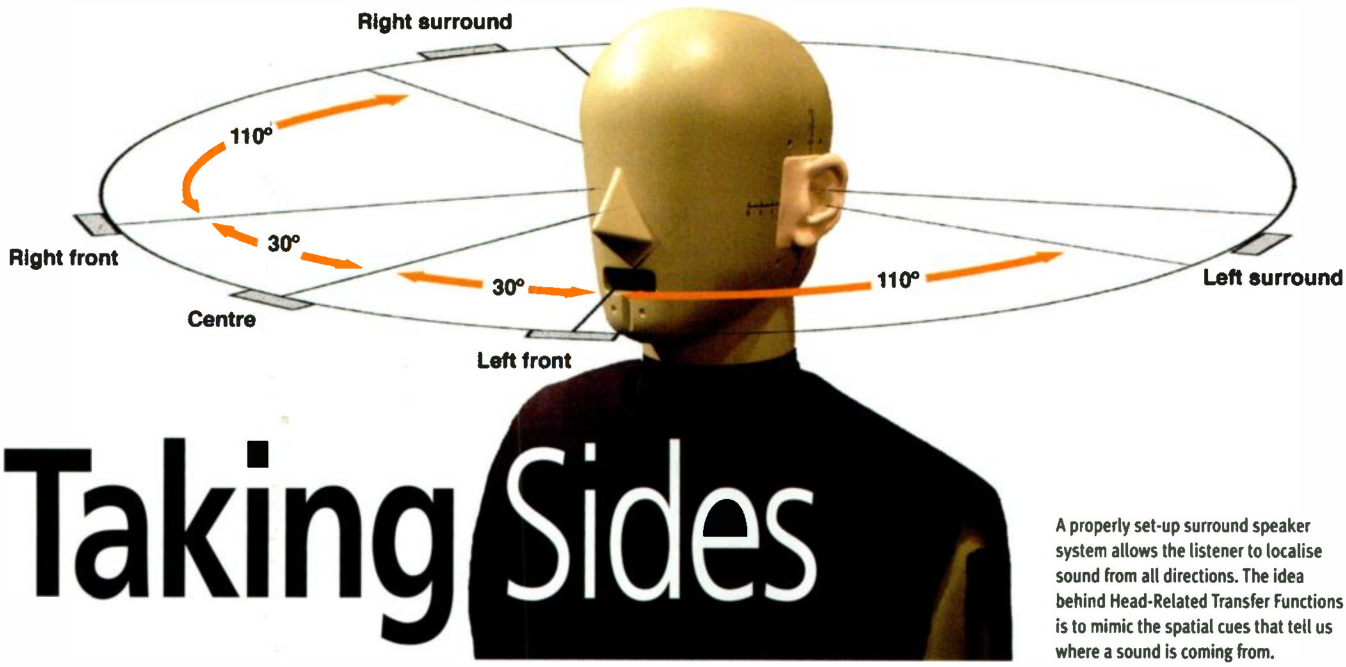
I unequivocally love this plug-in. The range of sounds obtainable using various amp simulators, effects and general guitar-oriented processes is seemingly endless. From sparkling, LA-style compressed and chorused arpeggios to full-on metal and down 'n' dirty blues, Real Strat just works with them all. Techniques such as unison bends, legato fret-slides and hammer-ons, which were so difficult and time-consuming to contrive using my former methods, are a breeze and sound totally convincing now; so much so that I feel compelled to revisit a particular ongoing album project and replace all my previous guitar emulations with Real Strat — it really will make that much difference.

If any criticism at all could be levelled at Real Strat it's that it sounds so unmistakeably like a Stratocaster that some may hanker for the earthiness of a Les Paul or the manicured tones of a Paul Reed Smith — but the label does say 'Real Strat', and that's just what it does.

So does this mean that I will no longer be needing to hire the services of real guitarists? Not at all! But when push comes to shove and budgets are non-existent, I can load up Real Strat and know that the results, although a mere caricature of what a good player would provide, will be far from embarrassing. SOS

information

- £ £133 including VAT.
- T Time + Space +44 (0)1837 55200.
- E sales@timespace.com
- W www.timespace.com
- W www.musiclab.com



Taking Sides

A properly set-up surround speaker system allows the listener to localise sound from all directions. The idea behind Head-Related Transfer Functions is to mimic the spatial cues that tell us where a sound is coming from.

Steve Marshall

All About MP3 Surround

Until now, huge file sizes and the need for dedicated equipment have prevented the distribution of surround mixes over the Internet. That could be about to change...

It's free, it can compress 5.1 surround files down to almost the size of a stereo MP3, the sound quality is excellent and there's a plug-in for Winamp. There's also a binaural option for playing 5.1 on stereo headphones, so you don't even need surround speakers. Does all that sound too good to be true?

MP3 Surround could eventually transform the way we listen to music, and it probably will. Yet since its 2005 launch, it has been largely ignored by the music industry and the media — who, quite justifiably, think they've heard this before.

Grounded Surround

The story of surround sound in the home has so far been one of heroic failure. Over the last 40 years, a succession of brilliant innovations have arrived with fanfares — then faded away to public indifference. The Quadraphonic system, launched in 1970, was implemented beautifully: to record four synchronous audio tracks onto a vinyl record is no mean achievement. But there was no agreement between manufacturers on standardising the system; technical difficulties and conflicting formats meant that very few people felt confident enough to buy into it, and so Quad died. But Quad was never going to accurately reproduce a 3D soundfield anyway, due to a basic flaw in the maths: four speakers placed in the corners of a room cannot produce the 'phantom images' that stereo or 5.1 can. At 90 degrees, the angles are too wide. It can give some good effects, but it's not proper surround sound.

Next came Ambisonics, again born in the '70s and still popular among some audiophiles and academics. Ambisonics is genuinely brilliant — it's still arguably the most accurate and versatile way to record and reproduce a 3D soundfield, and can even reproduce height. Endlessly adaptable, the system can incorporate any number of speakers, in almost any positions. (The Soundfield microphone is based on Ambisonics and can record 360-degree surround sound onto four tracks.) Ambisonics' failure has been attributed to its being promoted by the British government-funded National Research and Development Council — the same people who brought you the hovercraft. Or didn't.

From the '80s onwards came a succession of surround formats that readers may be more familiar with: Dolby Pro Logic, Dolby Digital, DTS and so on. The concept of placing the speakers in positions based on a listening circle was established; subwoofers provided the 'point one', and then innovation was simply a case of adding more and more satellite speakers to reach 5.1, 6.1, 7.1, 10.2...

Designed by the inventor of THX, the 10.2 format is claimed to be 'twice as good as 5.1' and includes two channels for height. The latest and most lavish surround format, though, is 22.2! With two entire levels of height and two subs, it is the companion to Ultra High Definition Video and I don't expect many of us will be buying it just yet.

Surround Today

So who does buy this stuff? Surround sound is regarded by the male-dominated hi-fi world as having an extremely low WAF, or 'wife acceptance factor'. How many people do you actually know who have a 7.1 system in their home? I know just one and yes, he's a bachelor. But surround sound is good value for money. As the number of surround channels has increased over the years, so the sound quality and dynamic range have improved, yet the cost continues to fall. So if surround sound has never been better or cheaper, why isn't it more popular?

It's not only surround sound that's experienced this increase in quality: digital stereo equipment has been going the same route. Super Audio CD and DVD-Audio offer much greater bandwidth and dynamic range than CD, yet today's most popular format turns out to be low-bit-rate MP3! How can this be?

The only answer is that the hi-fi approach actually leaves most people cold. Most people are not stupid; they know that MP3 is not such high quality as CD but they simply don't care. They don't want or need super-high fidelity; they want technology that fits in their pocket and is cheap, or preferably free. MP3 fits the bill precisely. The high data-compression of MP3 results in smaller files and makes it feasible to send tracks over the Internet quickly and easily, thus creating a new market for music sales. Hi-fi hasn't gone away, but it has become something of a niche market, like surround sound.

MP3 Surround

The public's unwillingness to invest in surround sound has not dented the audio industry's conviction that surround is the future. And now there is finally a chance that the industry could be proven right! For some years, surround capability has been creeping into music recording packages. Logic comes well equipped for up to 7.1 surround, and Steinberg's audio engine for Nuendo and Cubase was 'engineered from the ground up' for surround. PCs have been coming equipped with 5.1 surround cards for a while now — all of this for no particular reason, other than a commonly shared hunch that surround would somehow happen eventually. Enter MP3 Surround.

In this, the audio industry seems finally to have come up with a surround product that there is a demand for, and one that uses existing hardware. Investment in new equipment is optional but not essential: the important part is that the new medium is driven by free software. MP3 Surround was developed mainly by Fraunhofer, inventors of the original MP3 codec, and is currently available as a free evaluation download from their web site (www.iis.fraunhofer.de). The free software is available for PC, Mac or Linux. What's more, MP3 Surround is completely backwards-compatible: surround files will play on any of the previous generation of MP3 players, albeit in stereo only.

There are three main parts to the MP3 Surround system. First is the encoder/decoder or 'codec' (which means 'compress and decompress'). The encoder is the cleverest part of MP3 and is the result of detailed research into psychoacoustics, combined with some very serious number-crunching. By removing elements of the original recording that are inaudible, file sizes can be drastically reduced. In MP3 Surround the compression is astonishingly powerful, resulting in 5.1 surround files that are only about 10 percent bigger than stereo files!

The decoder comes separately and there are two versions: a stand-alone MP3 Surround player and a plug-in for the freeware Winamp that enables streaming. The second part of the system is Ensonido, a binaural simulator that allows the playback of MP3 Surround using only stereo headphones, using HRTF (Head Realted Transfer Function) technology to simulate the effect of a 5.1 soundfield. The third component of MP3 Surround is MP3 SX or Stereo eXtended. By analysing the ambience of a stereo recording, SX can synthesize a pair of rear channels and create artificial surround sound.

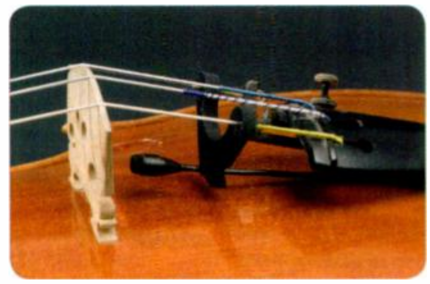
Downloading and installing the MP3 Surround software takes only a few minutes, and already there is some free music to listen to on the download site. Fraunhofer's business partner, the US Thompson company, has a much wider variety of tracks and styles on their site at www.all4MP3.com.

For best results you'll need a PC or Mac with a 5.1 soundcard, running five speakers and a subwoofer. If you don't have a suitable soundcard you can still listen in binaural surround by using Ensonido with headphones, but to appreciate how good the codec is, you should use speakers. The interface for the MP3 Surround player is as simple as it can be: you just drag and drop files onto it and select a playback method from 5.1 surround, Ensonido or stereo. Sound quality is extremely good, particularly when the amount of data compression is taken into account. I expected a thin, grainy sound but heard quite the opposite: the sound is solid and full, but with far more subtlety and detail than I would have thought possible.

MP3 Surround does have weaknesses, but they're really only apparent when directly compared with systems such as DTS or Dolby. MP3 Surround sounds impressive, but follow it with a DTS ▶

Making HRTFs

I visited the acoustics department of Salford University to see a typical setup for measuring Head Related Transfer Functions. It's not cheap or easy to do! An £8000 Bruel & Kjaer dummy head, fitted with measuring mics, sits in the centre of an anechoic room that is fully floating and acoustically isolated from the rest of the building. The foam wedges on the walls have to be as long as possible, to absorb the lower frequencies (they absorb at quarter wavelength). Even the floor is absorbent and has thick wire mesh suspended above it to walk on. This room is totally anechoic down to 100Hz and cost almost a million pounds to build!



track and you'll immediately notice the rather deeper bass and brighter top of DTS. Dynamic range is also affected, but again, this is not obvious until systems are compared. It is only to be expected, though: MP3 Surround runs at a constant bit-rate of 192 kilobits per second (kbps) but DTS runs at 1411kbps, while Meridian's 'lossless packing' system (MLP) used on DVD-A discs runs at a variable bit-rate of between 6000 and 9000 kbps! Take all of that into account and MP3 Surround is all the more amazing.

Making Surround MP3s

The MP3 Surround encoder is very easy to use. Easy, that is, when you've already made a multi-channel WAV. Then it's simply a matter of dragging and dropping the WAV onto the MP3 Surround interface and naming the new file. Initially, I objected very strongly to this method, because I couldn't actually make a multi-channel WAV and had to get a friend to do it for me! I would have much preferred it if the encoder could just be fed with six mono WAVs, but this is the demo version, and it is free.

My gripe is that multi-channel WAV is only supported by the latest version of most music packages, and that users (such as myself) may not want the disruption or expense of upgrading. For simply turning mono WAV files into multi-channel WAV's, Fraunhofer technicians recommended Copyaudio, part of the AFSP-library, which is freeware. I would regard it more as 'boffinware', as I found it completely unusable, but you might fare better. However, if you have the latest version of Logic or Soundscape, or any of the Steinberg products, making multi-channel WAVs should be very simple.

Unlike other surround codecs, MP3 Surround uses 'Binaural Cue Coding', in which the signal is represented as a single mono sum channel plus some difference data that is used to reconstruct the other channels.

Transfer Your Own Head

A great deal of research is going into Head Related Transfer Functions at the moment, largely because of the mobile phone industry. The advent of stereo Bluetooth means that some people will soon be wearing stereo headphones most of the time! Personal gadgets are all rapidly merging into all-purpose 'devices' that combine phones, music and video players, Internet browsers, games consoles and so on. With HRTF technology comes the capacity for adding surround sound and 3D 'mobile environments'. The idea is that you can wear a stereo headset and take phone calls, listen to music, chill out in a 3D tropical rainforest — that kind of thing. Some systems will also incorporate noise-cancelling to remove unwanted ambient sound.

All of this is possible even without the use of headphones. Spatial sound environments can actually be projected into the air, simply from a pair of micro-speakers an inch apart, mounted in one end of a phone. The old but effective 'transaural' technique has been enhanced by incorporating HRTF data: a combination of phase cancellation and time delays is used to eliminate crosstalk between two speakers, and to simulate the effect of wearing headphones. Once each of our ears is receiving only the signal that is meant for it (and none of the other channel), we're into binaural territory. HRTF coding can then be used to make the sound appear to come from much further away, and even from bigger, virtual speakers.

Amazingly, spatial sound is even possible from only one loudspeaker! A 'dipole' transmits sound from both the front and the back surfaces of a driver, radiating in a figure-of-eight pattern (the two outputs will naturally be out of phase with each other). It's then possible to process the resulting output using HRTF algorithms, simulating the effect of several speakers placed around a room.

It seems likely that some form of HRTF-based transaural technology will eventually become the most convenient way of listening to surround sound at home. All those messy satellite speakers, stands and cables could be eliminated by only using one or two speakers inside a TV to simulate a 3D surround system. With HRTF processing the effect can be spectacular: a convincingly solid soundfield is projected out into the room, which now appears to be full of correctly positioned surround speakers. Such systems have been tried in the past without much success, but as the technology continues to get better and cheaper, we can expect to see affordable systems that actually work.

Creating HRTFs is a complex and expensive business. Here at Salford University, a dummy head is placed in an anechoic chamber and test tones are recorded from strategically placed speakers.

But if all these techniques depend on Head Related Transfer Functions, don't we have a potential problem? Our ears are all different. Ears come in many shapes and sizes, and, like fingerprints, none are exactly the same. One approach is to search for the universal HRTF that suits most people, and there are already many patented variations on this theme.

Alternatively, technology is available that can tailor a unique HRTF to each individual. Either by a series of simple listening tests, or by inspecting a photo of your ear, it is already possible to generate custom HRTFs, and we can expect to see some radical developments in this area in the near future.

It will eventually be possible to modify Ensonido with custom HRTFs, although this has not been implemented in the evaluation version. Fraunhofer do stress though, that Ensonido is not a pure HRTF processor, and that: "It combines the acoustic reception of a human head, room acoustics measurements and simulations and equalisation. The individual HRTF measurement is only one element in the design of Ensonido."

▶ Comparing DTS & MP3 Surround

For the past few years I've been producing my own work in surround sound, using binaural location recordings made with a dummy head (or, more recently, with mics in my ears). I convert the binaural material into DTS 5.1 and make a kind of hi-fi musique concrete with extremely vivid spatial imaging — sounds come from every direction, even from overhead where there are no speakers. I have to sell it on disc because although the quality is superb, the file sizes are very big.

In order to put binaural samples on the 'net, I've already converted binaural material to stereo MP3, and with great success, even at low bit-rates. MP3 Surround will only code at a constant bit rate of 192kbps, which is considered to be hi-fi by most MP3 fans (MP3 can run as high as 640kbps, but rarely does). Fraunhofer's published listening tests indicate that most people can barely discern a difference in quality between their compressed version and the original audio, so I wanted to hear for myself, and directly compare MP3 Surround with DTS, using my own material.

On my Bilocation web site is a short demo mix in DTS 5.1 that can be downloaded and burned to an audio CD for playing on a 'home cinema' surround system. To me, the DTS compressed version sounds exactly the same as the six original WAVs, but it's a big file: at three and a half minutes in length, the DTS file is 36MB. I set about making a surround MP3 of the same mix so I could compare the two.

The first step was to go back to the original six WAV files — one for each channel of the 5.1 surround mix and all of exactly the same length.

The next step was to interleave the six WAVs by converting them into a multi-channel WAV. This must be done in the right order: Left Front, Right Front, Centre, LFE, Left Surround, Right Surround. This produced a big uncompressed file of 107MB (all my work is done at 44.1kHz but MP3 Surround will also accept 48kHz). The final step was to drag the multi-channel WAV onto the MP3 Surround window and marvel as the file was compressed in seconds, down to an amazing 4.86MB! And that's all there is to it.

Comparing the two formats was now simply a matter of switching between the DTS version playing from disc and the MP3 Surround version playing from the PC's hard drive. The Bilocation mix contains a lot of spatial movement, all of it 'real' rather than panned. MP3 Surround reproduced this quite well, with all the imaging pretty much as it is in the DTS version. There were differences, though, mostly with the 'above' effects. There was still a sense of 'above', but it was a bit vague, and not always in the right place. I assume this was due to the MP3 process removing too many 'unnecessary' frequencies. (I do realise this test is a bit unfair, as 5.1 isn't even supposed to have height!) One part of the track was recorded in a huge dome in India, where children clap and shout into a natural repeat echo. Although this still sounded good in MP3 Surround, it was here that the limitations showed up. The echoes that faded to silence were OK, but at one point there is a max-level (0dB) handclap, right next to the mics, that in the DTS version opens up the acoustic and explodes into ambience. The MP3 Surround version of this is disappointing and sounds very obviously 'compressed'. But really, that is my only criticism! Also, my material is hardly typical. If you'd like to try this experiment for yourself, both of the files are still available at www.bilocation.co.uk.

Look, No Speakers

One very exciting addition to the MP3 Surround system is Ensonido. Based on HRTF technology, it can simulate the effect of a 5.1 speaker system binaurally, for headphones. Head-Related Transfer Functions have been around for a long time (at least 30 years), but thus far they have been used mainly as an acoustics research tool. They are algorithms that mimic the contribution of the pinnae — the shapes and folds of the human outer ear — to our hearing. As sounds approach us from different directions, they are 'coloured' by the pinnae, in a way that the brain can decode as spatial cues. This is the principle behind binaural recording: the HRTF information doesn't have to be artificial, it can be obtained simply by stuffing a small pair of mics in your ears!

Laboratory HRTFs are made using a special dummy head with microphones inside it. The head is placed in an anechoic chamber to eliminate room ambience, and

MP3 Stereo eXtended

When the world gets used to MP3 Surround, ordinary stereo will seem far too flat and boring for us to listen to, right? I guess that's the reasoning behind MP3 SX. Simple to use, it can create new surround mixes from stereo: you just drag MP3 files onto the interface. Allegedly "the ambience of the original track is analysed and two rear channels are created to match". I still have lots of old mono records and have never felt the need to convert them to artificial stereo, but I tried to get enthusiastic about MP3 SX anyway. The word 'ambience' is very tantalising here, but misleading. I imagined that dry sounds would remain unchanged in the stereo mix, and that reverbs would become three-dimensional: quite an exciting prospect. So I made some MP3s to test it with.

David Bowie's 'Heroes' is well known for its innovative vocal treatment. The vocal starts off quite dry, but when Bowie starts to belt it out after about three minutes, the room acoustics open up and he seems to be louder. It's a great effect in stereo, but converting the track to MP3 SX had the exact opposite effect to what I'd imagined. The whole track was generally 'smaller' and there was a very unpleasant effect like a very slow, pumping compression where the overall level would suddenly drop for no reason. Tony Visconti would not be happy.

I tried some other tracks. Medeski Martin & Wood are a Hammond organ trio with a trademark sound that is rough and sounds very 'roomy'. Here again, the ambience was reduced by MP3 SX. Only cymbals seemed to be expanded into the rear channels, and they seemed to be coming through a cheap '80s chorus pedal. Brian Eno's 'Shadow' from the *On Land* album was the most successful track I tried, but only because it features a sound like tropical insects that happens to be at the right frequency to get copied into the rear channels. Tracks by Joe Meek and Abba, featuring spring and plate reverb, were very disappointing: hardly anything came out of the rear channels except a bit of hi-hat.

I give it two out of 10... See me later.

Point One

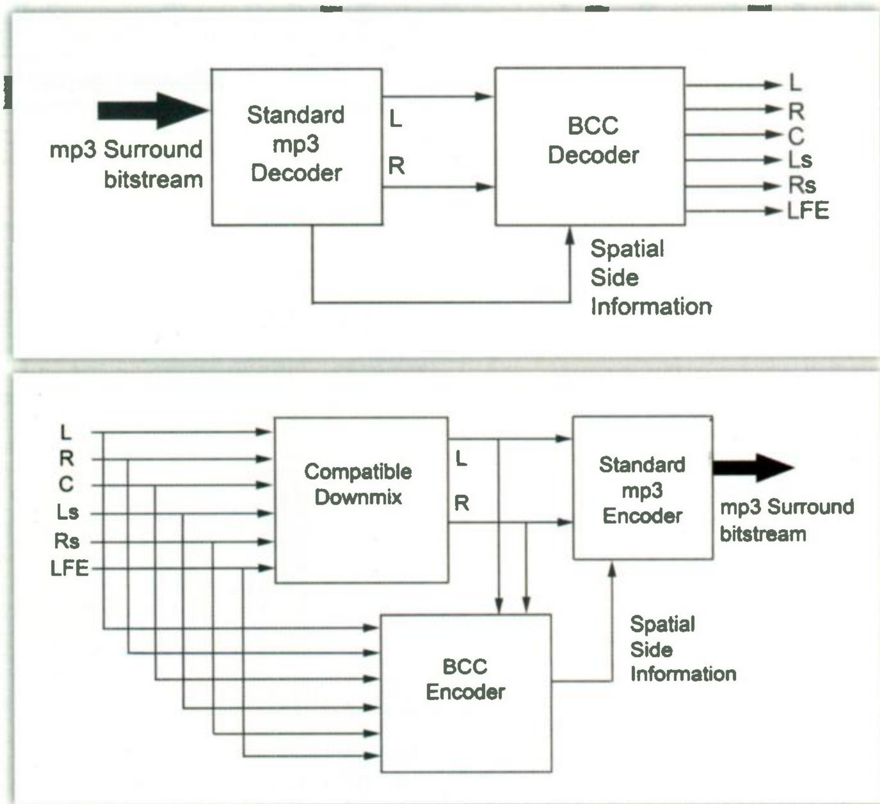
The 'point one' of surround systems is the LFE channel, so-called because of its very limited bandwidth compared to the other channels. LFE actually means 'low frequency effects' and it is supposed to be used for occasional film sound effects, such as explosions — not kick drums. When I had originally made my Bilocation demo mix I didn't have an LFE track, as the Minnetonka software that I'd used for DTS coding doesn't require one. The MP3 Surround encoder expects an LFE track whether it's used or not, so I had to record a WAV of silence that matched the other five in length and sample rate.

test tones or broadband noise are played in many different positions all around the head. The resulting set of recordings can then be analysed and subjected to Fast Fourier Transformations to extract the spatial information. This process is not cheap, quick or easy; until quite recently it was almost impossible to obtain HRTFs, and none were in the public domain until 2001.

Ensonido comes ready-fitted to the MP3 Surround player and the Winamp plug-in. There are four HRTF sets to choose from, and users are advised to experiment in finding which of the four best suits their ears. In my case, none of the four options gave very good results. I don't think my ears are particularly unusual, but although I could detect a difference in timbre between the four, I didn't really feel that I was hearing surround sound with any of them. Although the sides were good, I felt the front and back imaging was very shallow, only extending out by a foot at the most. I've heard far better imaging with old-fashioned stereo binaural, though friends have reported better results and I don't know why that should be.

I asked Fraunhofer how the four HRTF options differ: are they differentiated by size, sex or ear shape? It appears that the modes derive from composite HRTF measurements taken from dummy and real heads, combined with different room acoustics, and were "selected in listening tests to provide a good combination of localisation and spectral neutrality".

And what type of headphones are meant to be used? Open, closed or ear-buds? This is always a problem area for surround sound listening. Closed headphones can physically distort the pinnae. Open headphones are preferable, but the pinnae will colour the sound to some extent. Ear-buds, on the other hand, bypass the pinnae altogether, but tend to be rather compromised in sound quality due to their size. And although it's rarely mentioned, virtually all headphones, even the most expensive hi-fi ones, have a notch at 5kHz as part of their design. Fraunhofer had the difficult task of optimising Ensonido for all these factors, but they say that Ensonido works best for open headphones and ear-buds with as flat a response as possible.



The evaluation versions of the MP3 Surround encoder and decoder are simple to use.

How Does MP3 Surround Work?

The MP3 format actually dates back to 1992, and has been in common use for over 10

SoundTech LightSnake
Guitar-to-USB Cable


The Lightsnake USB offers just about the most straightforward way you could imagine of getting a guitar signal into your computer.

Described by manufacturers SoundTech as a 'soundcard in a cable', the LightSnake USB comes in instrument and microphone versions — there's a quarter-inch jack or XLR plug on one end and a USB connector on the other — and you can also now get an RCA and an iPod version.

The LightSnake works with Mac (OS 9 or 10) and PC (most Windows versions from 95 to Vista). A Sony 30-day demo software disc (including Sound Forge, Acid and Vegas) also comes as part of the package, to help you get started. An ASIO driver should be available by the time you read this, though it wasn't shipping at the time of this review.

I tried the guitar version and it couldn't be simpler: the USB end plugs into your computer and the jack into your guitar (or bass). The jack end includes an additional socket, piggy-backed onto the plug, allowing you to take a direct 'thru' feed into a guitar amp, for latency-free source monitoring — although if you do this you have to remember to disable software monitoring, or turn down the level.

A green LED, which glows to show that the cable is connected and powered up, is embedded in each end of the cable, and it flashes to indicate when it has been recognised by the host software.

The 16-bit A-D converter operates at sample rates of 48 and 44.1kHz, and HSDL (Host Side Data Loss) Noise Reduction is included. I'm unfamiliar with this system: it is said to minimise noise when recording, but there's no explanation of how it works and it may be as simple as a software noise gate.

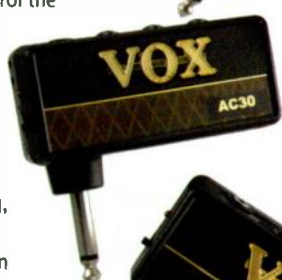
The 'soundcard in a cable' claim is only half true, as you only get the input part of a soundcard, not the output — you have to use your existing system audio output for that, and when I tested it with Apple's Logic Express this entailed setting up an Aggregate Driver combining the LightSnake and the on-board I/O, so I could play back what I'd recorded.

A gain stage precedes the converter, so the recording level is adequate for most types of electric guitar and bass, without risking overload on signal peaks. However, because the gain setting is fixed and needs to accommodate hot humbuckers without distorting, the recording level from my Strat was fairly low, with peaks at around -20dB — which meant that my recording resolution was equivalent to 13 of the 16 available bits. Normalising the recording to bring it to maximum level produced a subjectively decent sound without undue noise, so it shouldn't be a problem for basic demo work or recording.

Although the LightSnake, at £30, is definitely a budget device, it is capable of very respectable results. I wouldn't choose it for serious recording, but for no-hassle demos or working on a laptop it offers a very practical solution, and the 'thru' jack is a good idea too. Because the cable draws its power directly from the USB socket, it is very simple to set up — you literally plug, then play. *Paul White*

SUMMARY
An easy way to get your guitar sound into your computer. Though cheap, it is capable of good results.
Synergy +44 (0)121 270 6506.
www.synergydistribution.co.uk

Vox AmPlug Headphone Guitar Amplifiers



The Vox AmPlug: a big sound from the smallest of amps!

Vox are well-known for their amp-modelling technology, but I didn't expect them to come up with anything like this! Priced at £35, making them the perfect impulse purchase, the AmPlug range comprises three models of headphone guitar-amp, each of which has a specific amp voicing. They plug directly into the output jack of your guitar (and yes, they do fit a Strat-style jack), providing a mini-jack headphone out and a separate jack input for your MP3 player, just in case you'd like to play along with external music.

The three styles are Vox AC30; Classic Rock, which is based on a popular UK-made amplifier head; and Metal, which models a US high-gain amplifier. You might expect such a product concept to be based around digital modelling, but in fact the circuitry is all analogue, which means that you can expect a long battery life (up to 15 hours) from the two included AAA cells that power the unit.

The AC30 unit has two separate gain stages, while Classic Rock has four. Metal combines two ultra-high gain sections with a mid-cut filter, and Vox claim that the circuit response of the original amp is simulated to a high degree of accuracy, even down to the way the tone circuits sound and the way the sound cleans up when you back off the guitar volume control.

Each model has three thumbwheel controls for gain, tone and volume (you'll need to control the MP3 input level at source). There are no effects, but the basic sound is actually very authentic and musically satisfying, unless you turn the level up too far — in which case some clipping is evident and the sound tends to get a bit gritty. There's a bit of background hiss, especially at higher gain settings, but with care you could actually record directly from the output and get a usable result.

My favourite amongst these units is, quite predictably, the AC30 — which sounds very much like the AC30 top boost model of my Vox AD30VT amp — but they all do exactly what they are supposed to do and are a lot of fun to play. There's not a great deal of flexibility, but for general practice or even demo recording (where you can add your own effects later) the concept is excellent.

There are, of course, now numerous headphone-amp solutions out there, but this range embodies the distinctive characters of the amps being simulated. If you want all the bells and whistles, you'll have to spend around twice as much on a Pocket Pod or something similar, but the AmPlug is perfect if you're looking for a gizmo that you can put in your guitar case as a means of filling those quiet moments. *Paul White*

SUMMARY
A range of very small and surprisingly good-sounding headphone amps based on popular guitar amplifiers.
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on test processors

BEHRINGER SX3040 & SU9920

manages to beef up the bass very cleanly and naturally when applied sensibly.

Interestingly, it also enhances mixes in other, less predictable, ways. In particular it seems to boost the effect of reverbs and other low-level detail, thereby increasing the apparent depth of a mix and enabling the listener to experience hitherto hidden subtleties. Other reviews have said similar things about SPL and Aphex products, so the SX3040 offers a chance for those on a budget to get a taste of what they've been missing.

The only real issue, for me, is the lack of a high-pass shelving filter, which would be ideal for cleaning away those boomy sub-bass frequencies that get out of hand during processing. Clubs and certain dance mix engineers might not worry too much about this, but in general, having too much energy in the very low-frequency range detracts from the important parts of a mix. I tested this by severely filtering out the lower end on the stereo buss feeding the SX3040 and found that the sound cleaned up nicely.

Of course, the processor can be used on individual sources as well as mixes, but it would be easy to overdo things in isolation. Obviously, not everything can be super-energetic and up-front in a mix.

The overall operation is pretty idiot-proof. High drive settings usually require a moderate mix setting, and vice versa. The Tune control, when set at its minimum position, provides fairly unappealing sub rumbling, and it begins to interfere with the guts of a track when turned all the way up. Between those extremes the active bass frequencies are located and can usually be enhanced rather nicely.

Alternatives

The SX3040 has undoubtedly been inspired by Aphex's 204 Aural Exciter and Optical Big Bottom, which has the same number of controls, all labelled with the same names. The Aphex offers two channels, each with an exciter for enhancing the high end and a parallel compressor for improving the punch of the lower frequencies. Behringer's product shares almost exactly the same I/O connections too, but it lacks the operating level switching. It is also significantly less expensive.

SPL are renowned for their numerous Vitalizer products and, although they are more expensive than these Behringer units and generally offer a slightly different set of features, they are well worth auditioning to get a perspective on this market niche.

In the last few years Phonic have also pitched in with their T8300 enhancer, which differs in that it is a 2U valve processor. Like the rest of the alternatives, it costs more than Behringer's products. Phonic's A6100 is closer still, though the latest model appears to be discontinued.

The SU9920 is most obviously in competition with BBE's Sonic Maximizer. BBE actually have four current Maximizer products on sale, several of which are slight modifications of well-established members of the range. Of the current batch it is the semi-pro 428i model which is closest in terms of its features and position in the market. On the face of it, the features are very similar, but the BBE has been designed with particularly high-quality components, as is reflected in its higher price.

The SX3040 Sonic Exciter has two identical, independently operated channels. Each is divided into two sections: the first filters off and processes the bass (using compression and automatic phase-shifting); while the second is used to enhance the higher frequencies by adding extra harmonics.

SU9920 Sonic Ultramizer

Like the SX3040, the SU9920 processes both the high end and low end of a signal, but it takes a slightly different approach — in fact, it does not add any harmonic distortion at all. Exactly what it does to the audio, in technical terms, remains uncertain, as the manual notes are brief but, as Behringer explain, its brief is to do the processing without damaging "the relationship between the fundamental tone and the harmonics," which can often occur through equalisation. Simply put, it enables the user to increase the impact of the bass and treble frequencies without adversely changing the character of the audio. As mentioned earlier, the concept and design is similar to BBE's Sonic Maximizer, which splits audio signals into several frequency bands for dynamic equalisation processing. It then delays each band by a differing amount, in order to maintain their relative phase relationships, resulting in a psychoacoustic effect that makes the processed signal appear louder, clearer and more detailed. It seems that Behringer's Ultramizer is doing roughly the same thing, according to the specification sheet, which describes the enhancer section as a three-band phase delay and dynamic filter.

Feature Set

The rear panel is identical to that of the SX3040, as is the design of the knobs and


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the style of the chassis screen-printing, and, although it is even cheaper than its stablemate, it too has an internal power transformer. There are only four controls on the front panel, not including the global processor In/out button and, of course, the power switch.

Each channel has a knob labelled Process and another labelled Low Contour. The Process knob might have been better titled High Frequency Processor, as it is the control responsible for adjusting the upper part of the audio spectrum. More specifically, it applies up to 12dB boost at 5kHz. The more informatively titled Low Contour knob provides up to 12dB at 50Hz (although the manual claims it operates at 50kHz!).

To keep a check on levels, each channel has been given a horizontal meter comprising five LEDs. The range begins at -20 dBu and goes up to +10, before reaching the red Clip level, which presumably lights at +20. Apparently, the red LED illuminates 3dB before clipping occurs, but 0dB is the optimal output level.

In Use

Operating the Ultramizer is as simple as it can possibly be and requires no explanation, really. The only slight stumbling block is that the controls are reversed on each channel, like a mirror image of each other — it would, in my opinion, be easier to operate if they were in the same order on each side, as is the case with the SX3040.

The processor is surprisingly good at adding more bass and top without making a mess of things or somehow altering the fundamental character of the material, but it is best used in moderation. It's easy to get carried away and find yourself dialling in more bass and more high-end sizzle, and producing a brutally fatiguing mix. Describing the Ultramizer's character is tricky, because by its very nature it manipulates the source signal rather than adding anything of its own. What can be said, though, is that it seems to work in a way that traditional EQ does not, managing to boost the selected frequency range without drastically changing the balance of the piece of music.

Even more so than the Sonic Exciter, the Ultramizer is primarily intended for using as a stereo processor, as is evident from the single processor In/Out button governing both channels. For dual-channel use it would really be handy to have a stereo link button, so that the two sides are forced to adopt the same settings. On the plus side, having the two independent from one another does allow separate mono sound sources to be processed simultaneously using the one box.

Time To Get Excited?

At such ridiculously low prices, the SX3040 and SU9920 are well worth buying just for the sake of having them to hand for when that extra something is required. I can't imagine both being needed on a mix, but obviously they can also be used in the channel inserts or effect-send path of a mixing desk, or as a tool in the effects rack of guitarists and keyboard players.

Of the two, the slightly more expensive SX3040 is the more flexible and interesting. As a mix processor it can be used to improve lacklustre recordings, but it really shines when the source material contains something for it to get its teeth into. At this price it would be ridiculous to form many criticisms, but I would like to have an adjustable high-pass filter available in the circuit to eradicate troublesome low frequencies. A higher-end product of this kind should offer switchblade -10dBV/+4dBU operating levels, plus a more refined enhancer sound, but for a lot of jobs this Sonic Exciter will be just the ticket. The SX3040 is quite literally a cheap trick — but my goodness it does work!

As for the SU9920, I can see it finding a home in a large number of racks, particularly in small venues and other live spaces, where a quick tweak is needed to suit the environment. I think many people will use it to process instrument feeds, as it is ideal for speedily spicing up keyboard and synth parts, or even flat-sounding guitar rigs. Clearly, it also has its place in the studio as a mix problem-solver. Jobbing producers, for example, might find it very useful when their non-technical clients say, "Yeah, that's great but can you just make it sound better?" Tweak the knobs and hey presto, everything instantly sounds better!

As it is not particularly flexible, the Ultramizer is not a total mix-mastering solution, but it could be used as a component in the signal path for home mastering. The processor does not seem to degrade the signal with the addition of noise or any other kind of distortion, so there is no reason why it shouldn't be used across a whole mix — although it's worth remembering that mastering houses will undoubtedly be able to achieve a similar result using a more sophisticated bit of kit. SOS

Build Quality

Given the extremely low cost of these two Behringer products, many potential buyers will be wondering if the build quality has been sacrificed to save money. Actually, both units seem to be constructed reasonably well. All the inputs and outputs are securely attached to the metal chassis, so even the most vigorous plugging and unplugging of leads is unlikely to damage the more fragile circuit-board inside, and the rack ears are formed from 3mm-thick metal, which is more than adequate for the job.

Both products feel very light when you consider that they house mains transformers and have predominantly steel chassis. I removed the top panel of the SX3040 to look inside, and discovered that their lightness is because there is very little inside. In all, there are just two circuit boards mounted vertically, each measuring about an inch in height. The longest one is positioned just behind the front-panel controls, the other, a short board of less than 18cm in length, sits at the back behind the I/O. Other than that there is just a small mains transformer at one end, fixed midway between the IEC input on the back and the power switch at the front. The other end is completely empty, so there is plenty of room for air to circulate and keep components cool even when the unit is in a rack.

The only weakness in the design is that the control pots are supported by their fixing to the front vertical circuit-board, as far as I can tell, and simply poke through the metalwork instead of being secured on the chassis itself. This means that they flex easily and are not as resistant to knocks as the I/O.O

Other than the labelling denoting the model, the rear panels of both these processors are identical, with the usual power in socket, and balanced input and outputs presented on both jack and XLR.

information

£ SX3040 £86; SU9920 £69.
Prices include VAT.
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E email via web site form.
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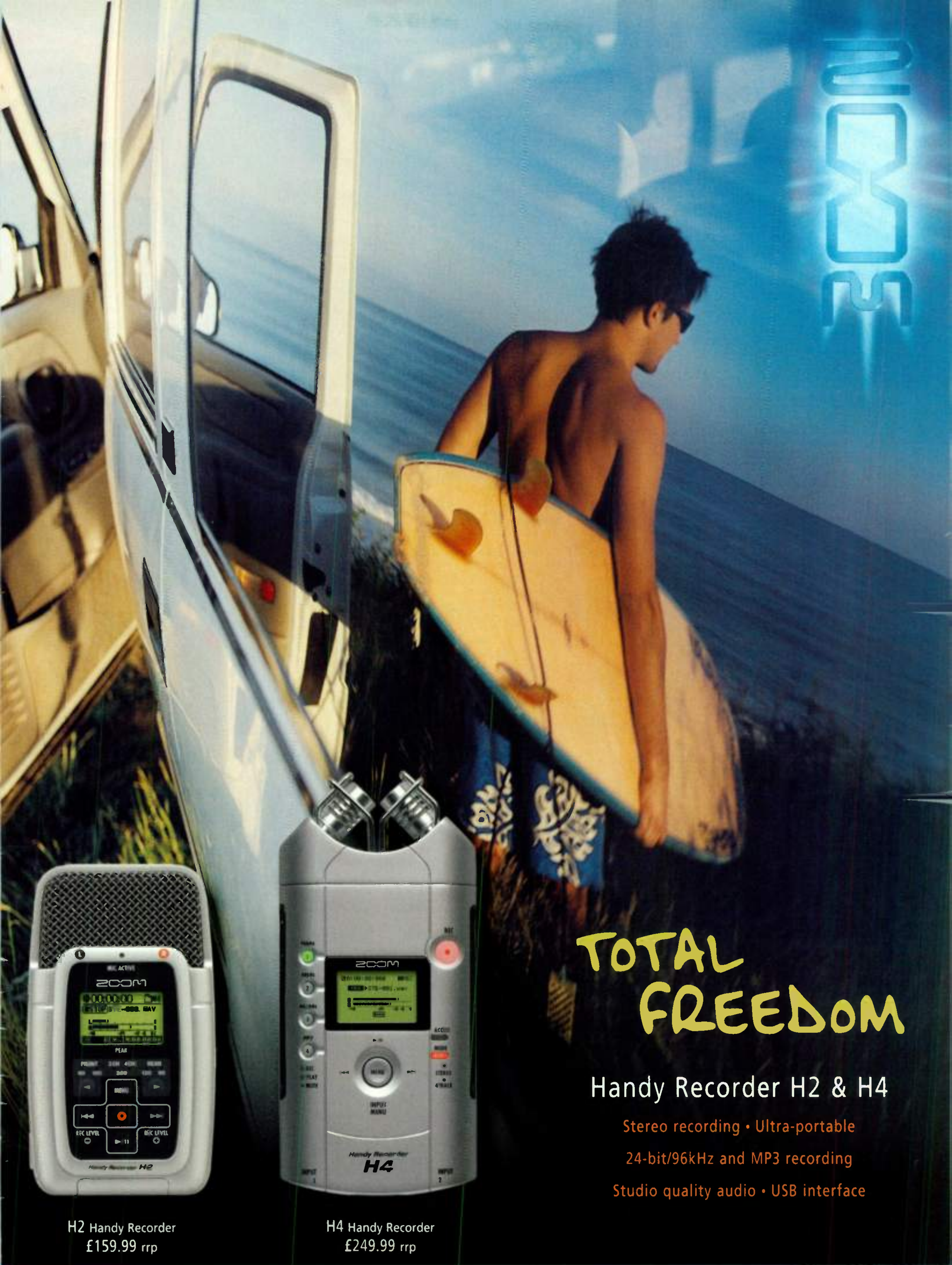
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Dual capsule design and unique three-pattern switch (cardioid, cardioid with -10dB pad and omni), handles everything from soft vocals to the loudest garage band and it's ideal for podcasting, frequency response 40Hz to 18kHz, for Mac OSX and Windows XP, USB 1.0 or 2.0, 64MB RAM recommended
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the t.bone SC450 USB
Large diaphragm microphone with USB
for use with Windows XP and MAC OS X, no special drivers needed, phantom powered via USB, external low cut and -10dB pad-switch, incl. shockmount and plastic case.
order code 195302 € 98.- £ 70.-

the t.bone SC450
Large diaphragm studio mic
Cardioid, -10dB pad, low cut filter, impedance 200 Ohm, 48V Phantom, 30Hz - 20kHz, size 50.5mm x 190mm, incl. shockmount and PVC case
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SC450 Stereo-Set matched stereo pair order code 174363 € 177.- £ 127.-

the t.bone RM700
Ribbon microphone
Sensitivity 1.8mV/Pa (-55dB), polar pattern figure-8, impedance 200 Ohm, max. SPL 165dB, frequency response 30Hz - 18kHz, S/N ratio 70dB, equivalent noise level 18dB (A-weight), weight 794g, dimensions 76 x 165mm, incl. shock mount
order code 180190 € 98.- £ 70.-
the t.bone RB100, ribbon mic, Polar pattern figure-8, frequency response 30Hz up to 15kHz, sensitivity -54dB, max. SPL 148dB
order code 107913 € 77.- £ 55.-

the t.bone RB500
Ribbon microphone
warm and most natural sound, Figure-8, 2 aluminum ribbon (2 microns thick), frequency response 30Hz - 18kHz, sensitivity -55dB (0dB = 1V/Pa), max. SPL 165dB, including soft padded carrying bag and yoke mic stand, XLR chord
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StudioProjects B1
Large diaphragm cardioid microphone
1" Capsule, trans-formerless circuit, needs +48V phantom power, frequency response 20Hz - 20kHz, Sensitivity -34dB (0dB = 1V/Pa), output impedance <200 Ohms, maximum SPL 132dB, noise 12dB-A (IEC651), incl. foam wind screen, zippered bag and shockmount
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the t.bone SC1100
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Neumann U87 Set
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polar patterns omnidirectional, cardioid and figure-8, 10 dB attenuation switch for singing, speaker, overhead, bundle including shockmount EA87
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True condenser microphone
18mm diaphragm, cardioid pattern, excellent for drums (overhead, snare, hi-hat), acoustic guitar or piano, frequency range 20Hz to 20kHz, needs phantom power 48V, with Low-Cut (-3dB @ 75Hz or 150Hz) and Pad (-10dB or -20dB), max. SPL 130dB, incl. shockmount, windscreen and aluminium case
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SC140 Stereo-Set matched pair, incl. shockmounts, windscreens, stereo bar and aluminium case
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Rode M3
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Cardoid pattern, optimally for recording strings, choir and acoustic guitars, frequency range 40Hz to 20kHz, 130dB SPL, 15dBA noise background, impedance 200 Ohms, 48V phantom power or battery 9V, status LED, high pass filter, windshield and stand mount included, dimensions 225mm lenght, 33mm diameter, weight 360g
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Shipping charges
Standard shipping to the UK is £7 per order, however we now offer free shipping on all orders above £135. Minimum purchase is £15. All prices include 17.5% British VAT, 3-year warranty and our 30-day money-back guarantee. Please visit us on www.thomann.de for further information.

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cardioid pickup pattern, on/off switch, incl. clamp and bag, ideal e.g. for voice and percussion, incl. Millenium MS-2005 microphone stand and 6m microphone cable
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Shure SM58 Set
Bundle containing:
Shure SM58 LC, dynamic vocal mic, cardioid, 50Hz - 15kHz, incl. holder and bag
the sssnake SM6BK mic cable, XLR - XLR male, 6m, colour: black
Millenium MS-2005 mic boom stand, colour: black
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Rode NT5
Condenser microphone stereo set
2 matched 1/2" condenser mics with cardioid characteristic in a plastic case, holder and windscreen included
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Neumann KM 184 Stereo-Set
2 Condenser Microphones KM 184 (with sequential serial numbers), cardioid polar pattern, incl. 2x WNS 100 windshield and 2x SG 21/17 stand adapter, incl. wooden box.
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Colour: black order code 158648 £ 751.-

Shure Beta 58A
Dynamic vocal microphone
polar pattern hyper cardioid for improved feedback resistance, neodymium magnet, frequency range 50Hz - 16kHz, excellent also for female voices, incl. holder and bag
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Beta 57A
Dynamic instrument mic, polar pattern hyper cardoid, frequency range 50Hz - 16kHz, excellent for toms, brass etc., incl. holder and bag
order code 105762 € 133.- £ 95.-

Neumann KMS 105
Condenser vocal mic
Neumann sound for stage, extremely transparent, excellent for vocals and speech, transformless design, hypercardioid, very feedback resistant, incl. mic holder, weight 350g, length 18cm
Black order code 142052 € 477.-
Silver order code 144972 £ 341.-

Behringer MS40
Multimedia speakers
Active 2-way studio monitors for computer studios, multimedia and keyboard monitoring, 2x 20W, 24bit/192kHz D/A, optical and coaxials digital input, 2 analog inputs (3.5mm stereo jack and RCA), with volume controls, 2-band EQ, headphone output (6.3mm stereo jack), magnetically shielded. Pair Price!
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Active 2-way monitor
70 Watt bi-amping bass-reflex system, 5" woofer, 3/4" tweeter, frequency range 55Hz - 20kHz, XLR and TRS input, input level adjustable, Mid EQ, Room Control, High Trim, Low Cut switchable, full magnetically shielded, unit price.
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Yamaha HS80M 120W bi-amping, 8"/1", 42Hz-20kHz, unit price order code 186463 € 238.- £ 170.-

ESI nEar05
Active 5" studio nearfield monitors
Magnetically shielded, 33Hz - 22kHz, bi-amped, bass 42W, treble 33W, balanced XLR input, 6.3mm jack input, size: 250 x 166 x 200mm, price per pair!
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Alesis M1 MKII
Active studio monitors
6.5" (75W), 1" (25W), magnetically shielded, rather linear frequency response, weight 20kg, size 38 x 22 x 25cm. Pair price!
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Active studio monitor
nearfield monitor with very good bass, mid and high frequency response, very reliable and long-lasting, DCW technology for improved stereo playback, XLR and jack inputs, biamplified, compact shape, weight 5.7kg, unit price
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Finish: white order code 137974 € 355.-
Finish: grey order code 137976 £ 254.-

Adam A7
Active near-field monitor
6.5" Rohacell / kevlar sandwich woofer, A.R.T. tweeter, 46Hz - 35kHz, 100W (sin), 150W (rms), XLR and RCA input, dimensions 18 x 33 x 28cm, weight 8 kg, unit price
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Active studio monitors
70W bass, 30W highs, balanced XLR input and unbalanced jack input, 35Hz - 20kHz, pair price!
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Recording/mixing workstation
Main surface for keyboard, mixing desks, loudspeakers etc., slide-out keyboard shelf, top shelf for computer monitor, bottom shelf for computer and printer, cord management system, integrated 19" rack with 4U rack space, 4 wheels (2 with lock), overall dimensions WDH: 91 x 76 x 103cm.
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StudioRTA Creation Station
Recording workstation
main work surface (151 x 75cm) for mixing console, keyboard etc., top shelf for monitor and loudspeakers, bottom shelf for computer and printer, integrated 19" rack with 2x 4U rack space, 4 wheels (2 with lock), overall dimensions WDH: 151 x 75 x 95cm
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StudioRTA Producer Station
Professional recording workstation
Integrated 19" racks above and below the main work surface. Plenty of rack space: 33U (2x 14U below and 5 additional units above main work surface). Pull-out keybord shelf, top shelf for speakers and computer monitor, bottom shelf for computer and printer, integrated CD rack for up to 16 CD's, huge main workbench (183cm wide, 76cm deep), stable steel frame construction, 8 wheels (2 locking). Overall dimensions WDH: 183 x 76 x 104cm.
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the t.bone HD 880
Homestudio headphones
Neodymium magnet, 64 ohms, 1W, frequency range 20Hz - 20kHz, 104dB, 3m cable, weight 285g, incl. 3.5mm adapter to 6,3mm, excellent for computer recording, electric pianos and keyboard.
€ 24.90 £ 17.80
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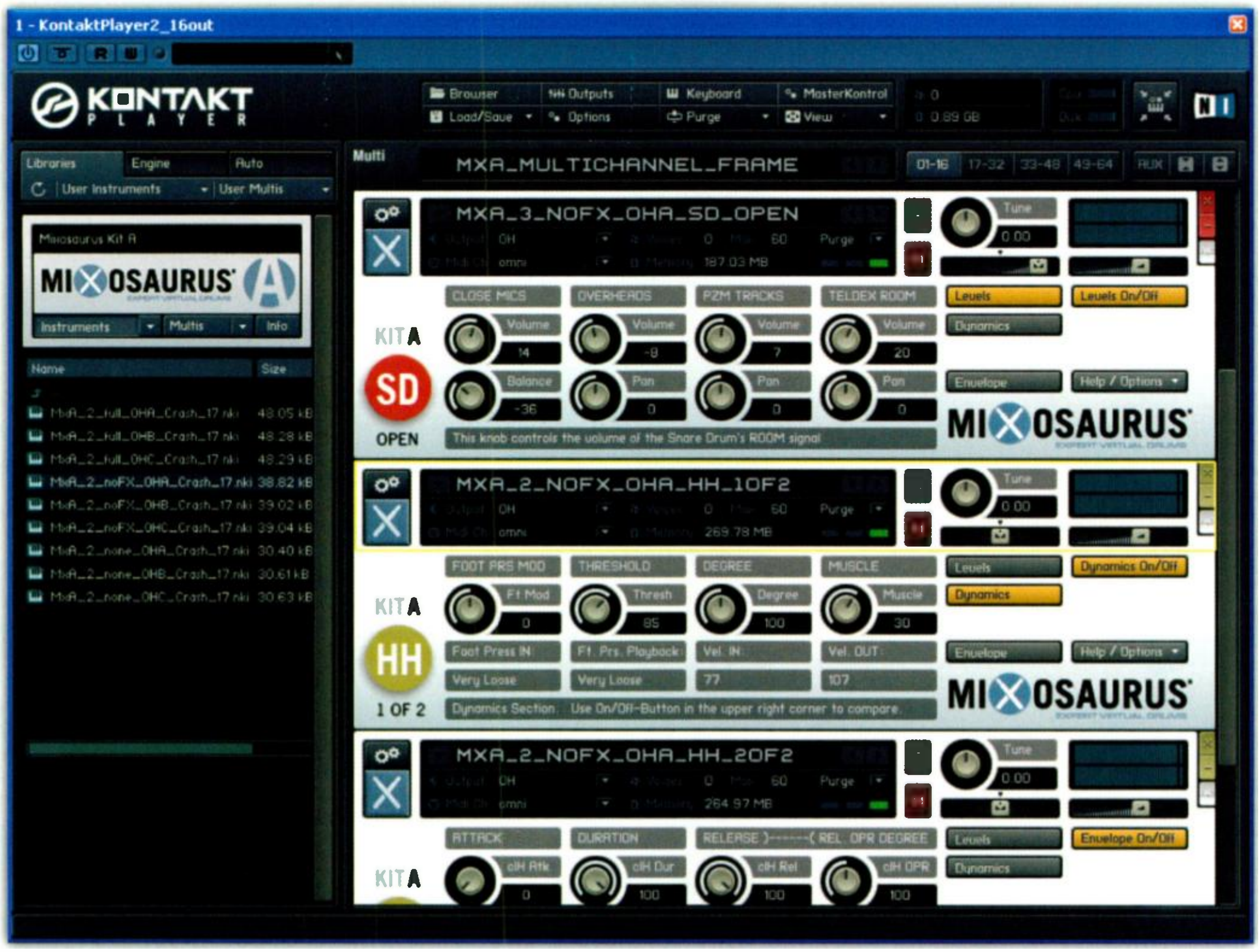
headphone holder for 19" racks (2 RU)
€ 4.90 £ 3.50
order code 147721
AKG K271 Studio
Studio headphone
dynamic, closed system, 55 Ohms, 91dB SPL/mW, 16-28KHz, ear-enclosing, automatic shut-off, incl. 3m cable, weight: 240g.
€ 118.- £ 84.-
order code 162407
extension cable
stereo jack 6.3mm, 5m
€ 10.90 £ 7.80
order code 153216
Sennheiser HD25 II
DJ headphones
Dynamic, 70 Ohms, 120dB, 16Hz - 22kHz (-3dB), closed system, 140g, incl. 200 cm cable, incl. velour earpads, bag and Native Instruments Traktor LE Software
€ 169.- £ 121.-
order code 117045
Sennheiser HD 212 pro
Powerful bass, 32 Ohm impedance, frequency response 12Hz - 19kHz, 112dB, 3m cable, ideal for DJs.
€ 36.- £ 26.-
order code 161045
Beyerdynamic
DT-880 Pro, dynamic, semi open, circum-aural, 5Hz - 35kHz, nominal impedance: 250 Ohm, pressure level 96dB, 3m coiled cable, weight without cable 295 g, gold-plated mini stereo jack 3.5 mm & 6.35 mm adapter (1/4")
€ 211.- £ 151.-
order code 197854
DT 770 M, closed system, 35dBA isolation from ambient noise, 105dB pressure level, 5Hz - 30kHz, impedance 80 Ohms, single side cable
€ 166.- £ 119.-
order code 166121
AKG K-701
Heaphones
revolutionary flat-wire voice coil technology, dynamic, open back, 62 Ohms, 105dB, frequency range 10Hz - 39.8kHz, 200mW, 6,3mm stereo jack, 3m cable, weight: 235g
€ 258.- £ 184.-
order code 185476
Millenium HA4
4-channel stereo headphone amplifier
output level control for each channel, input 1x 1/4" TRS connector stereo, output 4x 1/4" TRS connector stereo, DC 12-Volt adapter included, dimensions WHD: 12,9 x 3 x 8,3cm.
€ 19.90 £ 14.25
order code 107466
DOD SR460H
Headphone amplifier
Front and rear stereo inputs, 6-channel headphone distribution, master level control, "auto ramp" eardrum protection, 1W output for each headphone, 19"/1U
€ 68.- £ 49.-
order code 132381
Behringer HA4700 Powerplay Pro-XL
4-way headphone amp
2-band EQ per channel, 19"/1U, stereo main in with level control, stereo aux, mix/aux mix control, left/right mute, mono switch, up to 3 headphones possible, matrix mode, jack output
€ 109.- £ 78.-
order code 163272
HA8000 Powerplay Pro-8
8-channel headphone amp
€ 144.- £ 103.-
order code 165662
DOD SR430 QXLR
Graphic equalizer
Dual 15-band, constant Q, low cut filter at 50Hz with a 12dB/ octave slope, bal. XLR ins and outs, bal. 1/4" ins and impedance bal. 1/4" outs, 19"/1U
€ 68.- £ 49.-
order code 125455
DOD SR431 QX
31-band graphic EQ, XLR and TRS I/O, constant Q, frequency range 20Hz - 20kHz, 19"/1U
€ 58.- £ 41.-
order code 184646
dbx 1231
Graphic 2x 31-band EQ
Switchable boost/cut range (±6 / ±15dB), ground lift, balanced XLR and jack I/O, gain control, internal power supply, 19", 3RU
€ 325.- £ 232.-
order code 145238
the t.racks TFX-256 Pro
Stereo multieffect processor
256 presets, 20 bit AD/DA, 24 bit internal signal processing, 48 kHz sampling rate, true stereo reverb, delay, echo, flanger, chorus, panning and combination effects, hard bypass, two-colour level LED, input & output level controls, dry/mix control
€ 77.- £ 55.-
order code 191162
tc electronic M350
Dual engine rack processor
15 true and stunning stereo reverbs, 256 multi-effect/reverb presets + 99 user preset locations, a total of 355 presets, full DAW integration through VST/AU compatible editor, auto-sensing 24 bit S/PDIF digital I/O (coax), 44.1-48 kHz, 1/4" phone jacks in/out, MIDI in/out.
€ 188.- £ 134.-
order code 192915
For more information and a complete list of all our products and exciting offers here at Musikhaus Thomann, visit our new and updated website at www.thomann.de.
If you have any further questions please feel free to contact us on
0049 / 9546 / 9223 - 55
www.thomann.de
tc Helicon VoiceTone Create
Voice effects unit
99 music-style-based effects combinations offer fast setup, flexible A/B effects switching, 5 "favorite" user presets, quick and easy tweaking of patches, XLR mic level I/O in stereo or mono, clean studio quality mic preamp with phantom power, 2x XLR output (for mono or stereo operation), tap tempo
€ 229.- £ 164.-
order code 117131
Digitech Vocalist 2 Live
2-voice vocal harmonizer
2-part vocal harmony, selectable voicing: unison, 3rd, 5th-above or below your lead vocal, vocal enhancement effects matrix with separate compressor, reverb and EQ/enhance controls, XLR mic/line input with level adjustment, low-noise preamp and 48V phantom power, stereo 1/8" line out and mono XLR line out, power supply included
€ 315.- £ 225.-
order code 113250
Lexicon MX200
Stereo effect processor
2 different internal processors, 24-bit converter, VST plug-in architecture with USB interface, MIDI interface, tap-delay bottom, legendary Lexicon algorythm like halls, chambers, plates, delays, chorus, flanger, tremolo, rotary, etc., dbx dynamic algorythm, 2 simultaneous useable effects in diffrent stereo and mono configurations, analog and digital In and Outputs, int. power supply, 19"/1U, optional: foot switch for bypass and tap-delay (132867)
€ 215.- £ 154.-
order code 180403
Lexicon MX400
Quad processor multieffect
17 legendary Lexicon reverb algorithms, Lexicon delay & modulation effects, dbx dynamics, VST plug-in control via USB interface, 7 routing configurations, 2 x S/PDIF I/O, 4 analog inputs and outputs, 99 preset and 99 user programs for stereo and dual stereo, 25 preset and 24 user programs for surround, graphic display, impact, dust and humidity resistant design for serious live use, MIDI I/O, 19", 1U, weight: 3 kg.
€ 377.- £ 271.-
order code 195718
Digitech Vocalist 4 Live
Three to four voice vocal harmonizer
Uses MusIQ™ technology to directly analyze the output of any electric or pre-amplified acoustic guitar, generates the correct vocal harmony. Selectable voicing: unison, 3rd, 5th-above or below your lead vocal, vocal enhancement FX matrix with sep. compressor, reverb and EQ/enhance controls, on-board guitar tuner, XLR mic/line in, low-noise preamp and 48V phantom power, 1/4" guitar in, stereo 1/8" line out and mono XLR line out.
€ 419.- £ 300.-
order code 113244
tc Helicon Voice Works Plus
Vocal harmony and modeling processor
Voice modeling or harmony, advanced harmony humanization and control, mic, ad. harmony voice gender, megaphone effects, distortion, chorus, microtuning, flange with pitch feedback, pitch correction, reverb, delay with tap tempo, compressor, de-esser, 3-band EQ, gate, footswitch or volume pedal control, Burr Brown mic preamp with phantom power, Stereo or mono output
€ 759.- £ 543.-
order code 113771
ART Tube MP
Tube mic preamp
hand selected 12AX7A tube, limiter, +48V phantom power, phase reverse, XLR and jack I/O, perfect for hard disc recording or as a tube DI
€ 38.- £ 27.-
order code 191529
SM Pro Audio PR 8 E
8 channel microphone preamp
individual gain control per channel, peak light indicator per channel, phantom power on/off per channel, phantom power light indicator per channel, 8 balanced XLR/TRS inputs, 8 TRS balanced/unbalanced outputs @+4dB, distortion < .05% 20Hz- 20kHz, external power supply.
€ 148.- £ 106.-
order code 188256
SM Pro Audio TB202
Dual channel strip
2 hand tested 12AX7 tubes, Class A transformerless high voltage signal path, switchable phantom power, switchable phase reversal, optical compressor, EQ circuitry, input gain control, -20dB pad, 110V - 220V external power supply, dimensions: 1U rack
€ 148.- £ 106.-
order code 174864
SPL GoldMike 9844
2-channel tube preamp
Class-A, 48V phantom power, phase reverse, pad, "flair" switch for adding air
€ 385.- £ 275.-
order code 123370
SPL Channel One Mic Bundle
Including:
Channel One tube mic/line preamp, de-esser, compressor/limiter with ARC noise gate, EQ and headphone monitor, 48V phantom power, pad, phase reverse, separate distortion control
incl. the t.bone SC450 studio large membrane microphone with polar pattern cardioid
€ 945.- £ 676.-
order code 162062
RME OctaMic II
8-channel microphone preamp
192kHz/24bit AD conversion with 8 mic (XLR) and line (1/4" jack) channels, balanced TRS ins can be operated as real line inputs, max. input level of +21 dBu with gain range from 6dB up to 60dB, signal to noise ratio (SNR) ADC 107.5dB, 8 balanced line outs (1/4" jack), 2 ADAT outputs (S/MUX up to 96kHz), 4 AES/EBU outputs (D-sub, up to 192kHz), 1 AES/SPDIF sync input, phase, low cut and phantom power per channel, clip hold memory, 19"/1RU
€ 1169.- £ 836.-
order code 119911
Universal Audio LA-610
High-end vintage preamp
Combination of the classic all-tube designed preamp 610 and the optokompressor T4, (the heart of the Teletronix LA-2A), 48V phantom power, switchable impedance (500/2kOhm), bass and treble EQ, DI input, dimensions: 19" 2U, weight: 12lbs/5,5kg
€ 1444.- £ 1032.-
order code 175878
Rupert Neve Designs Portico™ 5012
2 channel class A mic preamp
Rupert Neve input and output transformer, switchable Phase, Mute, 48V phantom power, fully sweepable high pass filter (20 Hz to 250 Hz), SILK circuit adjusts the frequency spectrum, 9,5" 1U
€ 1590.- £ 1137.-
order code 191000
Avalon VT-737SP Bundle I
Bundle containing:
Avalon VT-737SP Mic Preamp, tube direct signal path (special performance), 2U 19 inch mono mic pre/opto-compressor/four band equalizer, four tubes + Class A discrete electronics, Mic-Line and DI inputs balanced in/out, large VU meter,
Studio Projects C3 large condenser dual diaphragm, cardoid/omni-directional/figure 8, Pad: 10dB, sensitivity 12mV/Pa=-38dB, max SPL 132/142dB by 1% THD@1000Hz, S/N 76dB, transformerless circuit, incl. mic cable
€ 2190.- £ 1566.-
order code 197808
RME MicStacy
8ch high-end mic/line preamp
192 kHz, 8 bal. XLR mic/line inputs with 85dB gain range, analog input level from -56.5dBu up to +30dBu, 8 bal. TRS line/instrument inputs with Hi-Z option, 8 bal. XLR line outputs, maximum level +27dBu, 4 x AES/EBU out per D-Sub, 8 channels @ 192 kHz, 2 x ADAT out, 8 channels @ 96 kHz, optional MADI I/O (164 MADI Card), LED level meter with 13 LEDs per channel, current state can be stored to 8 user presets, fully remote controllable
€ 3190.- £ 2281.-
order code 109027
dbx 266 XL
Stereo Compressor/Limiter/Gate
Patented OverEasy technology (attack & release times), XLR I/O, -10/+4db switch, stereo link, DBX sound
€ 138.- £ 99.-
order code 131808
dbx 166 XL
2-channel compressor/limiter/gate, stereo or dual mono, OverEasy or hard knee mode, PeakStop limiting, balanced XLR/jack I/O
€ 222.- £ 159.-
order code 131035
FMR Audio RNC 1773
Really Nice Compressor
Stereo compressor with two modes: normal for "effective" compression and Super Nice for totally transparent compression, controls: Threshold, Ratio, Attack, Release and Gain, switches: Bypass and Super Nice
€ 198.- £ 142.-
order code 179985
RNLA 7239, Really Nice Levelling Amplifier
10Hz - 100kHz 0.5dB @ 0dBu, Clip Point +22.5dBu @ 3% THD. The original proto of Really Nice Compressor, that was thought to be too "aggressive"
€ 235.- £ 168.-
order code 190166
BSS DPR 404
4-channel compressor
Threshold and ratio control, attack and release, fast slew switchable, high frequency de-esser, balanced inputs and outputs, 19" 1 unit
€ 999.- £ 714.-
order code 106583
Behringer SRC2496 Ultramatch Pro
Digital converter
24-Bit 96 kHZ AD/DA converter and sample rate/format converter, AES/EBU and S/PDIF (coaxial and optical), 16, 20 or 24 bit resolution
€ 125.- £ 89.-
order code 184267
Behringer Ultragain Pro8 Digit. ADA8000
8-channel A/D- & /D/A-converter
8 mic preamps with phantom power, 24bit AD DA 44.1 & 48kHz, Wordclock or ADAT-In sync, ADAT I/O, ADAT in can be routed to line outs, mic and line inputs are routed to ADAT out, excellent expansion for DDX 3216 or any interface or mixer with ADAT I/O
€ 225.- £ 161.-
order code 164573

Mixosaurus DAW Drums Kit A

Virtual Drum Instrument For Mac & PC



Sam Inglis

If you dream of the ultimate sampled drum kit, and you have a cutting-edge computer to run it, Mixosaurus's Kit A might be your Holy Grail.

Sometimes, it's only after you agree to review something that you realise what you're letting yourself in for. I knew that Mixosaurus were attempting something a bit special with their DAW Drums Kit A, but I had no idea quite how ambitious it would be. This is, without a shadow of a doubt, the most detailed and comprehensive sampled drum instrument ever created. If you don't believe me, take a look at the specs: DAW Drums Kit A contains over 1200 patches, made up of some 122GB of data. There are over 80,000 stereo samples here, all in uncompressed 24-bit format. It ships on its own hard drive, with a heavily adapted Kontakt Player front end. And — get this — it's one drum kit.

That's right: Kit A features just one kick drum, one snare and one hi-hat, along with four tom-toms, two ride cymbals, three crashes, a splash and a China cymbal. These few instruments are, however, sampled in unprecedented detail and with an extraordinary number of articulations and variations. Miking, for instance, encompasses three different sets of overheads including vintage ribbons and an M/S pair, internal and external kick drum mics, close mics above and below the snare, stereo PZMs and, uniquely, stereo signals from the reverb chamber at Teldex Studios in Berlin, where the recordings were made.

There are no fewer than 29 different articulations for the hi-hat alone, including tip, shank and 'crash' closed hats, each sampled at seven different levels of footpedal pressure. The snare is sampled using seven articulations — edge, halfway and centre plus three different rimshots and a rim-click — at different levels of muffling. Four different beaters are available for the kick drum, and as for crash and ride cymbals, well, you probably get the idea. Oh, and Mixosaurus include seven alternate versions of each and every sample in the entire set. Other sample libraries provide alternate samples, but usually only for the most-used sounds such as the louder snare samples, so this goes way beyond what's on offer in other instruments.

Going For A Drive

The hard disk containing Kit A is shipped in a very smart external caddy, engraved with the Mixosaurus logo. Mixosaurus recommend connecting either via Firewire 800 or eSATA, but a six-pin to six-pin Firewire 400 cable is also included. My Windows laptop has no eSATA or FW800 port, and only a four-pin FW400 socket; having scrounged an appropriate cable, I had no trouble mounting the drive or installing the software, and found that it is usable over Firewire 400, although it's not hard to run out of bandwidth when a full kit is playing back. As the manual states, "in a normal drum groove, Mixosaurus will continuously play 100-200 stereo voices", and a faster connection will definitely improve the Kit A experience.

The Kontakt Player application is installed from the hard drive, and you need to connect to the Internet and run NI's Service Centre application to authorise it. Annoyingly, the Mixosaurus installer insisted on reinstalling the Service Centre itself, thus overwriting the newer version that was already in place on my machine, so the first thing it did on finding the NI web site was update itself again! After that, installation went without a hitch.

The Game Of The Name

Mixosaurus have made full use of Kontakt's scripting capabilities to tailor the interface to the needs of this instrument, and they've done

Informative built-in Help is a nice touch.

a good job, but it still takes a little while to get your head around it. This is not one of those virtual instruments you can simply fire up and play. First, you need to choose one of two 'frame' Multis, depending on whether you're using Kit A in a multi-channel environment such as a DAW host, or in a stereo context, as when Kontakt 2 is running stand-alone and addressing a stereo soundcard.

Next, you need to understand Mixosaurus's patch-naming system. Selecting Instruments in the Kontakt 2 browser window displays a list of folders with friendly names like 'Snare_Drum_muffled' or 'Cymbal_Ride_light'. Open one of these and you'll be greeted with seven sub-folders allowing you to choose variants with the relevant number of alternative samples; as you'd expect, more samples provides a more natural sound at the expense of more sample RAM being used to pre-load them. Each of these sub-folders then contains (usually) nine different Instruments. The reason that there are nine is that you get a choice of three overhead miking setups (small-diaphragm condensers, vintage ribbons, and M/S) combined with a choice of three levels of processing. All of these are indicated by slightly cryptic conventions in the naming of the Instrument. For example, 'MxA_3_full_OHA_SD_mufld' is the muffled snare drum sampled using overhead setup A (vintage ribbons), with three alternating samples, and the full choice of processing. It gets more complicated with kit pieces that are sampled in particular depth, such as the hi-hats, which are divided across anything up to four Kontakt 2 Instruments, depending on how many alternating samples you require.

Double-click an Instrument or drag it into the main Kontakt Player window and, after a lengthy pause while the samples are pre-loaded, you can explore the editing and processing options made available by Mixosaurus's scripted interface. The interface is nicely done, presenting all the controls clearly on a number of pages, with a friendly built-in Help function.

The full processing palette provides a Levels option, allowing you to balance the relative contribution of the different mics for that Instrument, a simple but effective Envelope editor, MIDI dynamics (see 'Muscle Power' box), plus delay, filter and distortion effects. With the intermediate 'noFX' versions of each Instrument, you get the Levels, Envelope and Dynamics sections but not the last three, while there are also versions that offer none of these facilities.

In practice, I found I always used the intermediate versions of every Instrument. The no-frills versions I found too restrictive: the fixed balance of mics for each Instrument means you can't, for example, tweak the amount of snare going to the reverb chamber. By contrast, I'd always rather apply effects in a DAW than within a virtual instrument like this. In fact, to my mind there seems little point in using a sample library of this calibre if you're going to obliterate its subtleties with heavy processing. The Envelope settings in the intermediate versions are useful, however, ▶

SOUND ON SOUND

Mixosaurus Kit A 690 Euros

pros

- Absolutely no compromises have been made on sound quality.
- Makes you realise what a versatile and expressive instrument a single drum kit can be.
- Everything here is sampled with unprecedented thoroughness.
- The Teldex reverb chamber is a nice option.

cons

- Getting the best from Kit A requires a fair amount of drumming knowledge, and preferably the ability to play it from a controller.
- Demands considerable computing resources, and ideally, a lot more than 2GB RAM.
- Kontakt Player interface is not as friendly as dedicated drum instruments.

summary

Mixosaurus DAW Drums Kit A is an awesome feat of sampling. It sounds fantastic, but requires a lot of RAM and a fast machine.

The Kit A Envelope section. The 'OPR Degree' parameter allows you to make the release time in the overheads, PZMs and reverb chamber mics longer or shorter than in the close mics.

especially for taming the resonance of the toms. A particularly nice touch is that you can set up an offset to make the release time in the overheads, PZM and room mics longer than that in the close mics, so you can, for example, gate the snare's close mic without unnaturally affecting the way it sounds in the ambient mics.

The multi-channel frame Multi addresses eight stereo outputs into your DAW via a customised mixer window in Kontakt Player. This allows you to apply Kontakt's effects to individual mics, rather than on a per-Instrument basis, and balance the overall levels of the various mics, but it's a shame that it's not possible to mute or solo the individual channels here. By default, the instruments with two close mics — kick, snare, and hi-hat — emerge on stereo channels, with each mic hard-panned. You may want to change this in your DAW mixer, but in many cases I actually liked the sense of width this produced. In my system, the Kontakt Player plug-in defaulted to a surround output configuration rather than one based around multiple stereo outputs; to change this, you need to load one of the frame Multis, open the mixer, click Make Default, and then re-load the plug-in, but this only needs to be done once.

Overall, I think that Mixosaurus have done a good job of customising the Kontakt Player interface to suit the particular needs of their virtual instrument, but it's nevertheless fair to say that it will never feel quite as slick or user-friendly as a virtual instrument that is custom-built from the ground up. When you look at how the likes of BFD or Addictive Drums handle kit construction, mixing and editing, what's on offer here is less intuitive and takes longer to get to know. There is, thankfully, a printed manual: it's clear and well written, but sometimes seems to jump ahead of itself, and assumes prior knowledge of Kontakt.

That said, once you have that knowledge, the Kontakt Player interface is perfectly usable, and I imagine that most users will settle on one or two preferred setups fairly soon, after which there won't be much need for tinkering. And after all, the most important thing is the sound.

Power Play

So how *does* Kit A sound? Without wanting to be facetious, the best answer I can give is that it can sound as good as you want to make it sound. If you habitually program drums by knocking up two-bar loops in a grid editor or step sequencer, the results you'd get from Kit A probably wouldn't justify the investment of money or system resources. Not that it would sound bad, but the difference compared with smaller libraries wouldn't be startling.

Where Kit A comes into its own is when you play it live from a MIDI controller. When you work a lot with sampled drums, it's easy to forget that a real drum kit is a versatile instrument. It's easy to get into a mindset where "make the drums sound different" automatically equates to "load a new set of samples". Beyond switching the kick-drum beater or the muffling on the snare, that's not really an option with Kit A. Instead, you quickly realise that the way to make this drum kit sound different is to change the way you play it. Here is a snare drum that really feels like a living, breathing musical instrument; you can pound the crap out of it, but you can also do light, jazzy stickwork, and both will sound great. There are no separate samples of flams, drags or rolls, but convincing ones can certainly be played or programmed. The toms, likewise, have a really lively and musical ring to them, and the inclusion of really playable rim-clicks and rimshots is very welcome. The difference between Kit A and other instruments is particularly apparent at low velocity levels.

I was sorry that I didn't get a chance to test Kit A with a proper drum-kit controller, because I imagine it would work superbly. Features such as the multiple levels of hi-hat foot pressure cry out to be used with, well, something that responds to foot pressure! Kit A is very playable from a MIDI keyboard, where you use the mod wheel to control hi-hat foot pressure, but you will need a large one to take full advantage (keyboard, that is, not foot). The sheer range of articulations on offer means that Kit A Instruments cover nine octaves, although Mixosaurus have sensibly relegated less-used sounds like China, splash and mallet cymbals to the highest part of that range.

No details are given in the documentation as to what makes and models made up the kit, and it turns out that Mixosaurus's Uwe Lietzow has an interesting take on this issue. He believes, with much justification, that the choice of heads and drum tuning is at least as important as the name on the drum shell, and that the choice of sticks makes a huge difference with ride cymbals; with Kit A, his aim was also to create a versatile kit that would suit many different applications. As a result, he chose to use relatively standard-issue, albeit high-quality, drums and cymbals rather than esoteric or vintage models. The kick, for instance, is a 22-inch Pearl Masters series, while the toms were Yamaha Maple Custom.

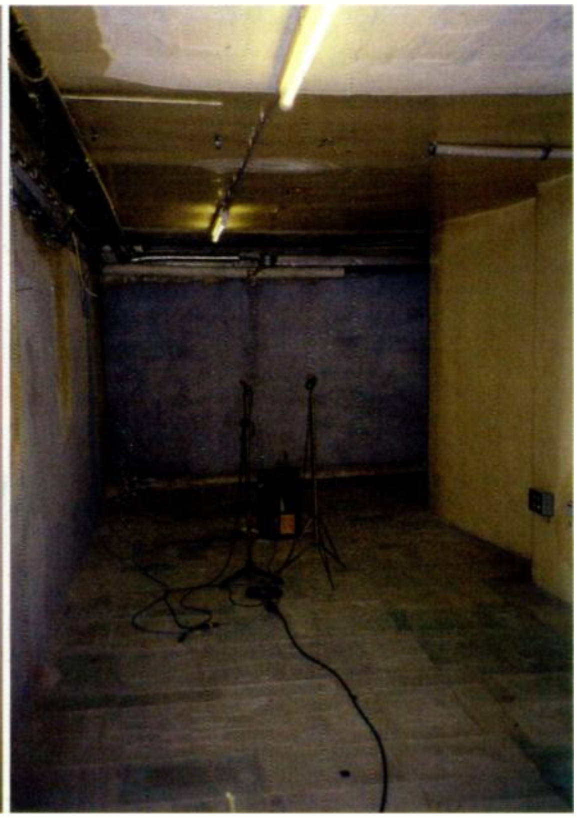
Another important respect in which Kit A is different is in Mixosaurus's approach to room miking. Most rival high-end drum instruments seem to present drums with what might be called a 'room signature': sampled in a live room with a distinct sonic character, with ambient or distant mics as well as conventional overheads and close mics. In Kit A, by contrast, what you get is dry overheads with a relatively neutral character, augmented with two 'special effects' channels. One is a stereo pair of pressure-zone microphones, heavily compressed; the other is a feed from the reverb chamber at Teldex studios.

In both respects, I think the results bear out Uwe's philosophy. The drums and cymbals sound great, and through careful use of different articulations, MIDI velocities and mic

Muscle Power

Among the many scripting enhancements Mixosaurus have made to the standard Kontakt Player interface are real-time MIDI processors that allow you to modify the apparent playing style of the drums without having to edit your sequences. The most important of these is a MIDI Dynamics processor for each Instrument. This takes the incoming MIDI velocities and compresses or expands them, with an additional Muscle parameter that acts rather like a make-up gain control. So if, for instance, the entire hi-hat is too prominent in the mix, you can instantly give it a negative Muscle value and have it played more gently — which, of course, sounds different from, and often more convincing than, simply turning down the level of the hi-hat mic.

Unlike some rival drum libraries, Kit A was recorded in a relatively small, tight-sounding room.



The reverb chamber at Teldex Studios provides a unique ambience.

balances, will give sounds that work in a wide variety of styles. The sampling is virtually flawless, and you certainly never find yourself thinking "I wish they'd sampled a handmade one-of-a-kind iridium-shelled piccolo snare instead of this!" Conversely, if you want to add the character of a real room to your drum mix, the world is full of convolution reverbs; but the PZM and chamber signals in Kit A are unique. The PZMs give you the sort of short, subtle ambience that acts more like a loudness enhancer than a noticeable reverb, while the chamber has a uniquely dense, complex reverb that is nonetheless short enough to work well with fairly up-tempo drum tracks. Leave them both off, and you have an untainted dry sound with unlimited potential for manipulation.

Mixosaurus Grooves

As well as Kit A itself, the hard drive includes several folders of MIDI files, which Mixosaurus call Grooves. The Standards Grooves folder contains 20 or so short sequences of typical beats for styles such as samba, bossanova and merengue, while Example Songs contains complete drum parts for a number of well-known hits, including Michael Jackson's 'Beat It', Robbie Williams' 'Let Me Entertain You' and rather too many Eagles songs.

Probably more useful in practice are two folders called Ride Cymbal Patterns and Hi-Hat Patterns. If you're not comfortable playing in ride and hi-hat patterns from a MIDI controller, it would take a lot of programming to exploit all the relevant articulations in a realistic fashion, so these will give you a big helping hand.

A Kit Too Far?

In short, I think Mixosaurus's claims to have created the ultimate sampled drum kit carry a lot of weight. A skilled programmer with a decent understanding of drumming can create a truly first-rate drum recording, and not just in a straight-ahead rock style (check out some of the demos on the Mixosaurus site for proof). If you have to have the best, you'll want to investigate Kit A, but be warned that its quality and versatility come at a heavy cost in terms of system resources.

I've already noted the fact that a Firewire 800 or eSATA port is desirable, and an even more pressing concern is RAM use. Although the instruments stream from disk, so many samples have to be pre-loaded that it's easy to run out of memory even when you have 2GB (and there would be no point even trying to run Kit A with less). In my system, with 2GB RAM, it was impossible to load the entire Kit A drum-kit at one time. In fact, if you want to make full use of the alternating samples, you can exceed the limits of a 2GB system by loading the hi-hat alone! According to Uwe, Kontakt Player has no problem addressing larger amounts of RAM on Apple computers and most Windows configurations, and I think 4GB at least should be considered desirable.

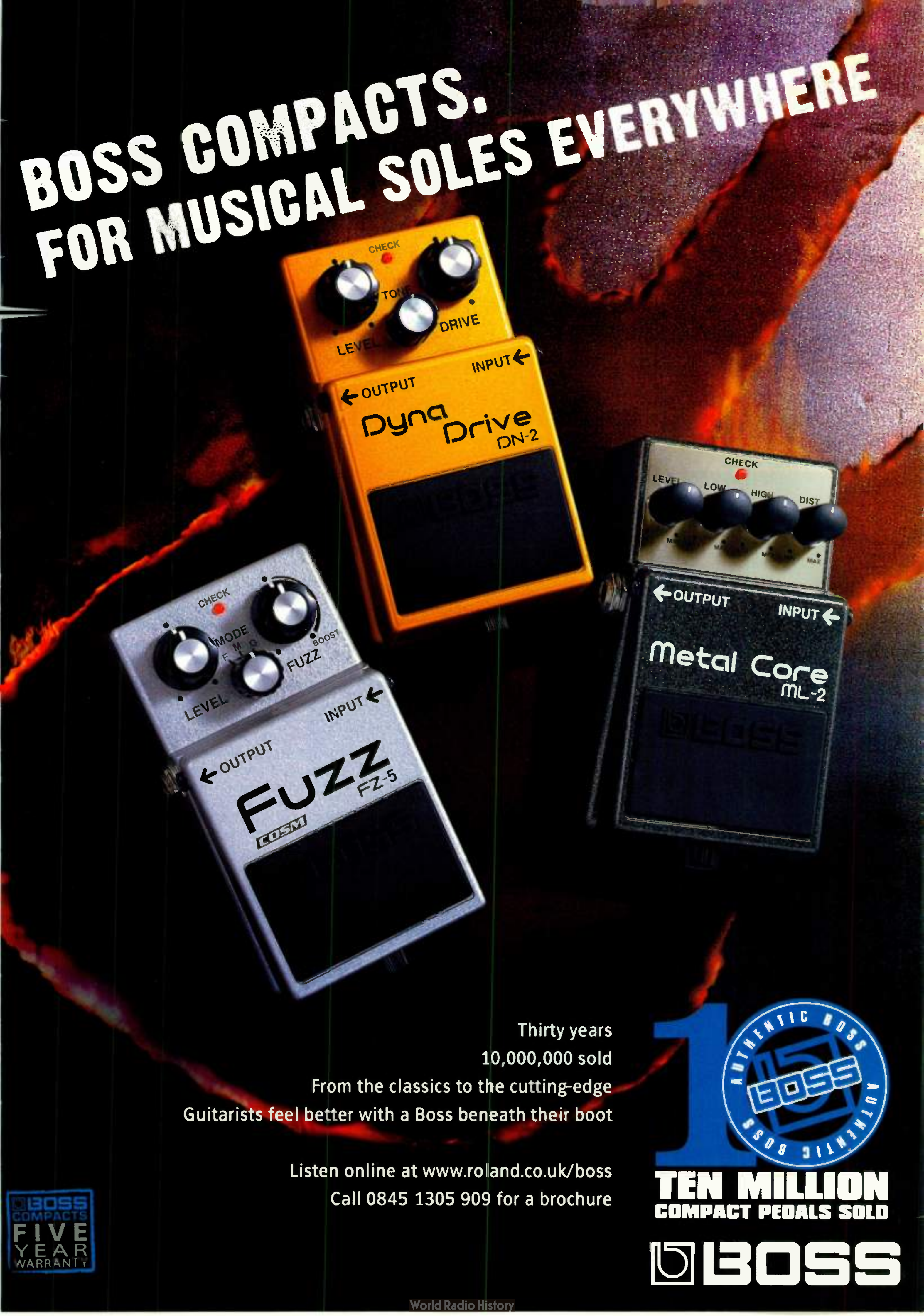
The best workaround in other systems is probably to use smaller Instruments with fewer alternating samples when you're playing or programming your drum tracks, and then switch to larger ones when you want to bounce down. You can use smaller pre-load settings for off-line bouncing, which will enable loading of more or larger Instruments.

In general, though, Kit A is sufficiently demanding that only a really cutting-edge machine will be able to run it live alongside other virtual instruments. More practical would be to create your drum tracks first and then bounce them down, or even use a separate computer just to run Kit A. People dedicate entire machines to orchestral libraries, so why not a drum library?

If that makes it sound as though Kit A is running at the limits of what's possible with current computer technology, that probably isn't far from the truth, and I get the impression that this is a product that will really come into its own when 64-bit music software gets going. When that happens, it will be a mouth-watering prospect, and I can imagine a future where session drummers turn up at studios with a set of V-Drums and a computer running Kit A... SOS

information

- **£** 690 Euros including shipping and VAT.
- **W** www.mixosaurus.com
- **E** inquiry@mixosaurus.com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WAVES API COLLECTION

▶ hard, medium and soft-knee compression curves, you can also insert a choice of two high-pass filters into the side-chain, thanks to a circuit API call Thrust. Unlike the side-chain filters in many compressors, these don't just roll off frequencies below a certain value; they apply a 'tilt' to the entire frequency spectrum. You can also choose whether the detector element is fed directly from the input signal ('feed forward') or from the gain-reduced signal ('feedback'). Finally, there's the unique option to have varying amounts of stereo linking between channels; not only can you set a percentage degree to which levels in one channel affect the other, but you can insert filters so that, for example, bass notes in one channel will only compress that channel, but peaks higher up the spectrum will compress both channels.

In the short time I've used it, the Waves API 2500 has become my first choice as a mix compressor. Some mixes just seem to lend themselves to being compressed, and you feel that almost any dynamics processor will do an adequate job. API 2500 does a far more than adequate job in these cases, but where it really shines is in more difficult situations. There are mixes where you don't really want to resort to multi-band compression, but where stereo compressors just seem to lurch between pumping queasily and sucking all the life out of the track. It's with these tracks that API 2500 really earns its stripes: it just seems to have the capability to put things in their place so that nothing leaps out obtrusively, but the punch and impact of drum beats is retained.

The Tone controls, in particular, make a profound difference to how 2500 responds. With a slow attack, 'normal' Thrust and 'Old' (feedback) tone Type, a drum-led mix will kick like a mule, but even settings that would result in bland smoothness on other compressors can produce a forward, in-your-face excitement here. It's almost as though you can pick out different elements of a mix and use them to drive the compressor. You can use external side-chain inputs in the RTAS and TDM versions, and my only regret as far as bus compression is concerned is that there is no programme-dependent release setting, but there isn't one on the hardware 2500, so it's hardly surprising that Waves haven't added one.

As you might expect, API 2500 also makes an excellent dynamics processor for individual sources. Again, it excels on things that trouble other compressors, such as very dynamic or tonally inconsistent vocal takes.

API Days

At £1463 for the TDM versions and half that for the native-only option, these are not cheap plug-ins, and if you're only interested in having an API-style EQ, URS's A-Series parametric EQ is more affordable at $500 (TDM) or $250 (native). In my opinion, however, the 2500 compressor is an absolute stand-out product. It's up there among the very best compressor plug-ins at any price, and if you're willing to pay for quality, it should be high on your list of things to try out. SOS

information

£ £1463 (TDM) or £734 (native). Prices include VAT.
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Mix Rescue

This month we tame a vocal that varies in dynamic range, and look at ways to create 'bigness' in the mix without overcooking levels.

Mike Senior

It's not often that *SOS* gets sent a track for Mix Rescue which lasts almost 10 minutes and has no bass part to speak of, but I suppose there's a first time for everything! The track in question, Imprint's 'Let Me Go', draped lush, gothic-tinged strings, bells and piano over a stark framework of distorted industrial loops and apocalyptic sound effects. At the centre of all this was an impassioned vocal performance reminiscent of singers such as Tori Amos and Bjork, courtesy of the song's writer, Sin.

First Stop: Vocal Processing

Listening to the band's original mix, my overriding concern was the vocals, so I started work there. While crucial to the song's emotional impact, these were rather indistinct and buried in the original mix. A more upfront sound, despite the obligatory long reverb, was required to make the lyrics more audible and encourage the listener to connect with the performance.

Apparently recorded in a single take with an AKG C900 handheld condenser mic, the wildly dynamic performance presented a number of technical problems. To start with, extreme level fluctuations made it impossible to find a good general level for the vocal fader within such a busy mix, and some serious compression was clearly needed to deal with this. However, I didn't want to end up with an aggressively squeezed rock-style sound, which wouldn't really have been appropriate here.

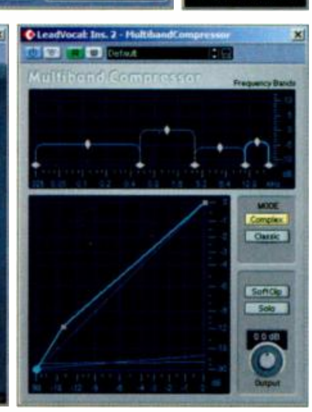
The compressor I always think of when I need heavy-but-transparent dynamic control is the classic Teletronix LA2A, so I went on the hunt for a plug-in which might emulate this design. Noticing an unmistakable similarity between the GUI of the freeware Antress Modern Painkiller plug-in (www.modernplugins.in-tw.com) and the front panel of the LA2A, I gave it a whirl and it turned out to be just the ticket, heaping on the gain reduction with comparatively few side-effects — I was even able to use the more assertive limiting option with no problems.

Against a basic rough balance of the other tracks, it quickly became apparent that other aspects of the vocal needed addressing too. It sounded as if Sin had been moving around quite a lot while performing, which meant that the amount of bass boost on the recording due to the proximity effect was pretty inconsistent. My first instinct was to just roll off some of the low end — I do this pretty much as a matter of course with busy tracks, because this biases the vocal tonality towards the upper frequencies and tends to improve intelligibility. It helped a little, but by the time I'd dealt with the boomier notes, the thinner ones had become 'size zero'.

I've encountered this problem before, and my solution to it is to use a multi-band compressor to strap down the lower frequencies firmly, evening out the vocal tone. I used Cubase's Multiband Compressor, and inserted it before the Modern Painkiller so that the low-end unevenness wouldn't cause the full-band compressor to react unmusically. A fairly high-ratio setting kept things well under control, and while I was at it I also gently compressed the octave above 11kHz to give the sound a slightly airier and more intimate quality.

This wasn't all, though, because Sin was making virtuoso use of the change between her 'head' and

Mike went to work on the main vocal first of all, using the processing shown here: the Multiband Compressor plug-in was used to deal with fluctuating proximity effect and sporadic vocal harshness; the full-band Antress Modern Painkiller compressor took care of more general dynamic control; and GVST's GSnap sorted out a few moments of shaky tuning. The Platinumears IQ4 plug-in was added right at the end of the mix process to cut a single abrasive frequency region when Sin was using her chest voice.



'chest' voices, and the microphone suited her head voice much more than her chest voice — especially in the 6-10kHz region. Lots of energy in this region was giving the head voice a nice, breathy quality, but it was also emphasising a harsh edge to her chest voice. Equalisation was, naturally, of no real use here, so I went back to the multi-band compressor and applied some high-ratio compression in the 6-10kHz range to try to take the edge off the chest-voice harshness.

Rescued This Month...

Based in London and the South East, Imprint comprises members Sin and Blink, who create their own unique blend of dark, ambient electronica. They are also involved in a number of collaborations with other artists, such as Dead Industrie and Autarky, and are currently preparing to take their sound out live.
www.myspace.com/imprintuk

Sin, who forms half of the band Imprint, lays down some vocals.

Reverbs & Delays

The dry vocal sound was now beginning to hold its ground against the other tracks, so I turned my attention to its effects, as this was another vital element of the mix that I felt I should tackle early on. A lot of the time, I wait until I've got all the dry tracks working together before getting too heavily involved in send effects, but here I wanted to have the freedom to create the desired 'intimate vocal in a vast underground cavern' sound I was after, before hemming myself in with the other tracks.

A long reverb was clearly essential, so I pulled up a four-second 'warm cathedral' impulse response in Christian Knufinke's new SIR2 plug-in. Although the reverb wash was important, I also had to make sure that it didn't pull the vocal back into the mix or get in the way of the clarity of the lyrics, so I took a few precautions against this. A long pre-delay setting of 160ms kept the onset of the reverb well away from that of the dry sound, and some adjustments to the amplitude envelope of the reverb softened the initial early reflections and focused the effect more on the smoother reverb tail — this also helped stop vocal sibilants from ricocheting around distractingly.

I now had some sense of a large space without swamping the vocal, and I supplemented with a couple of delays to enhance the sense of size. You might ask 'why not just use more reverb?' The reason is that too much reverb quickly fills up all the gaps in a mix, making it sound cluttered, whereas delays (and particularly tempo-sync'ed delays) can give much the same subjective impression of size as reverb but without taking up as much mix real-estate. ▶

I used two different delays in the end, one sync'ed to half notes and the other to quarter notes. The former was fairly uncomplicated, with a little feedback and some EQ bracketing (using a high-pass filter at 360Hz and a low-pass filter at 6.6kHz). I gave the quarter-note effect a bit of a different character by plumbing it through a mildly overdriven rotary-speaker simulator, and then compressing the delay return to duck the delay. This got it more out of the way while Sin was singing, but allowed the echoes to bubble up between vocal phrases.

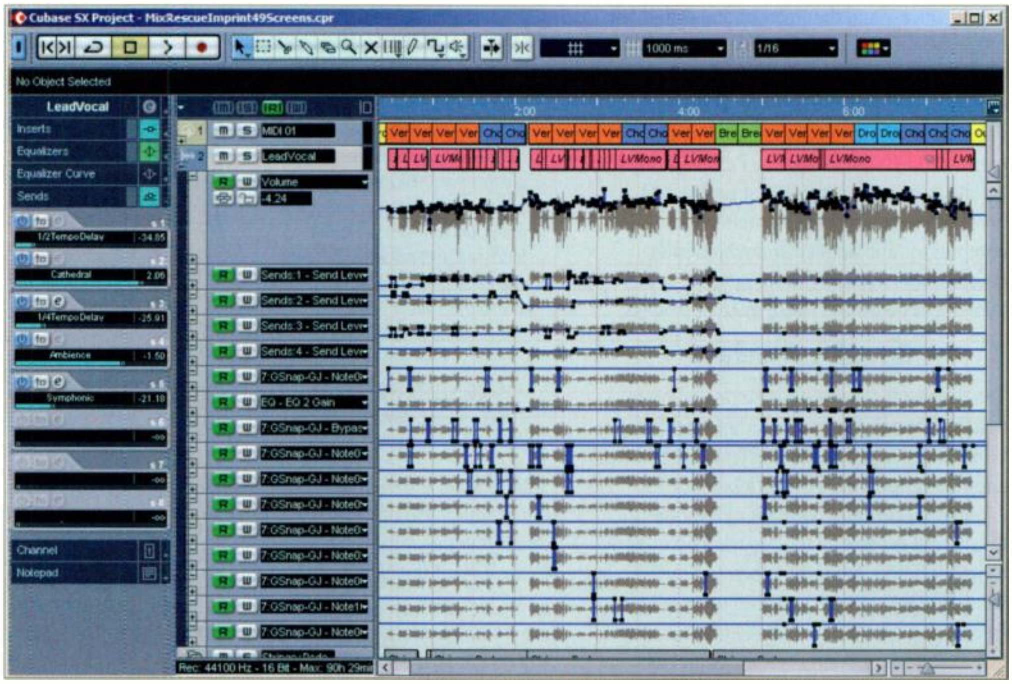
In a similar way to how I'd deliberately softened the onset of the reverb, I also tweaked the delay lines to 'blur' them a bit, and set them back in the mix. A little modulation of the delay time helped make the repeats a less distinct, but the main thing I did was to pass each of the delays through Silverspike's Room Machine 844 room-ambience simulator. This plug-in is great for giving the impression of a real space, but it doesn't weigh down your mix with a big reverb tail, so it was ideal for this task.

The drive and compression applied to the quarter-note delay was bringing up the sibilant sounds in the vocal part, drawing undue attention to the effect and making it difficult to find a decent level for it in the mix. Inserting a de-esser at the start of the effects chain easily sorted this out, though — you can get away with far harsher de-essing in an effects return than you can on the main vocal signal itself, which makes it a doddle to set this up.

By this point, I had a nice warm-sounding space surrounding the vocal, but with the singer well clear of the effects and perfectly intelligible. However, this did inevitably make the vocalist sound a little detached from the track as a whole. One way around this would have been to pull the existing effects a bit further forward, but I chose to use another separate effect instead, leaving myself more flexibility for re-using the vocal effects on other tracks later on.

What I went for was a very short ambience impulse response from SIR2, which I then high-pass filtered at around 650Hz to just add some early reflections to the upper end of the sound. Using a very short, bright reverb patch like this is a good way to glue a vocal to a backing track while still keeping it apparently 'dry' and upfront. I opted for using no pre-delay, to encourage the dry vocal to merge with the ambience and cohere better with the rest of the track, although I did use SIR2's envelope settings to soften the onset of the reflections again.

As a final touch, once the reverbs and delays were in place, I also set up one more send effect — a modulation treatment from Cubase's Symphonic plug-in. This was just to widen the stereo image of the lead vocal a little, which is a favourite tactic of mine where there's only one vocal in the track and I want it to seem bigger and more hi-fi.



Once all the main parts of the arrangement were in place, Mike spent some time automating the fader, EQ and effect sends on the lead vocal to maximise the intelligibility of the lyrics, and to keep a fairly consistent vocal tone as the mix texture changed. The lower block of nine automation lanes were entered while setting up the GVST GSnap pitch-correction, adjusting the plug-in settings to keep the processing as natural-sounding as possible.

Beefing Up The Drums

The other main challenge with this mix was dealing with the drums, which comprised a main loop and an extra, distorted layer. Sin had told me that these were as important to the track as everything else put together, and that they really needed to pack a punch, so I tried a variety of tricks to get them sounding as big as possible. Pushing up the fader was only the start — subjective power is as much about illusion as it is about raw signal levels.

There are various ways to fool the ear into thinking that drums sound louder than they are. One approach is to try to increase the sustain of the drum hits, and a way to do this is to compress. I used Digital Fishphones' Blockfish with fast attack and release times to dip the drum transients and bring up the tails of the hits. While I was at it, I took advantage of Blockfish's saturation control to add subtle distortion artifacts to the sound, which is another common way of increasing subjective loudness. For similar reasons, I also added some emulated tube saturation from Silverspike's Rubytube. There isn't any real logic behind the saturation settings I chose — I just fiddled with the controls until I found something that sounded 'bigger'. One thing to watch out for when using distortion on drums, though, is that overcooking things can begin to cause bass drums to break up and lose some of their punch, so I was careful to avoid that.

An unwanted side-effect of my processing was that the level of the one little treble percussion sound in the loop ended up being too loud in relation to the main drum sounds, so I popped Digital Fishphones' Spitfish de-esser in to poke it back down to a suitable level again. I also boosted a couple of decibels with a peaking filter at 800Hz to harden the 'knock' of the kick drums a little more, and notched down some drum harmonics that were dominating a little at the low end — cuts of 3-4dB at 47Hz, 59Hz, 96Hz, and 174Hz were all that were needed. I often high-pass-filter the low end of loops to leave room for the bass, but obviously there was no need for that here!

Although things were already better, the drums were still in mono (despite being supplied as a stereo audio file), and I wanted to give them a bit of width. On another recent project, I'd had some luck using a stereo pitch-shifter for this purpose, so I rustled up a send to MDA's Detune plug-in and set up a subtle five-cent shift. And it sounded rubbish! In a fit of pique, I smashed the hell out of it with Rubytube, which (to my

surprise) made it begin to sound quite good, giving a bit of life to the upper frequencies of the loop, so I high-pass filtered the return channel at 780Hz to home in on this. Curious now to see if the effect might benefit from a bit more sustain, I took a deep breath and broke a solemn oath that I'd made during my first week of using Cubase SX: thou shalt under no circumstances use the bundled Reverb B plug-in. Fortunately, inserting this between the Detune and Rubytube plug-ins worked so well that the resultant episode of self-loathing proved fairly manageable...

Adding some of each of the reverb and delay send effects to the drums glued them to the vocal, although I kept the level of the Cathedral impulse much lower here, relying more on the quarter-note delay and ambience sends. The ambience level was set quite high, as this also helped increased the impression of power, a bit like room mics can do in a drum-kit recording.

Ducking & Pumping

My final trick for inflating the drums was to use the drum channel to rhythmically duck the string and synth parts in the arrangement, much as you might routinely do in many dance styles. Cubase SX 2 doesn't make it easy to do this [*this has been overhauled in Cubase 4.1*], because there's no side-chain access to its plug-ins, but I got around it using Twisted Lemon's Sidekick v3, which implements its own side-chains, independent of the host sequencer. Because you can only use a limited number of instances of this plug-in at once, I set up the ducking as a send effect, regulating the amount of ducking per track using the individual track send controls. The disadvantage of this approach was that if I decided I wanted more or less ducking on a given track, changing the send level also changed the mix balance of that track. Not ideal, but workable nonetheless.

The ducker was set up with an instantaneous attack and a release time of around 200ms — which is longer than I normally associate with pumping effects, but it was what sounded best with Sidekick for this track. Having set up the ducking on the string parts, I also used it for the hyper-distorted drum layer, so that it would contribute more between the drum hits.

Finally, I very slightly pumped the track as a whole, using the Antress Modern Compressor plug-in, again setting the release time by ear to around 160ms. I brought its limiter slightly into play too, while also simultaneously sneaking up the drum fader, effectively ducking the whole backing track very briefly for each hit.

Fleshing Out The Drum Loop

The distorted drum layer which is added during more climactic sections of the track took a little more work. The tonality of the distortion was the biggest problem, as it included lots of energy at the frequency extremes. At the low end this clouded up the whole mix, while at the high end it masked the high frequencies of the vocals and strings, making them seem dull. High-pass filtering at 140Hz and cutting 18dB at 9kHz with a high shelf sorted out the worst of the problems, but I still needed to cut 3dB at 5.3kHz and boost 4dB at 830Hz before I could find a decent level for the sound in the mix. Again, the stereo image of this track was very narrow, so I sent to the Symphonic plug-in from here as well.

As the vocal and drum sounds began to slot into the rough balance of the other tracks, I began to feel that, despite the copious distortion, the drum parts still weren't delivering enough interest at the high end, so I decided to layer in an additional percussion loop for the second half of the track. A tambourine overdub clearly wasn't going to cut the mustard here, so I pulled something off a Glitch/IDM sample library I'd recently reviewed (Soniccouture's Abstrakt Breaks).

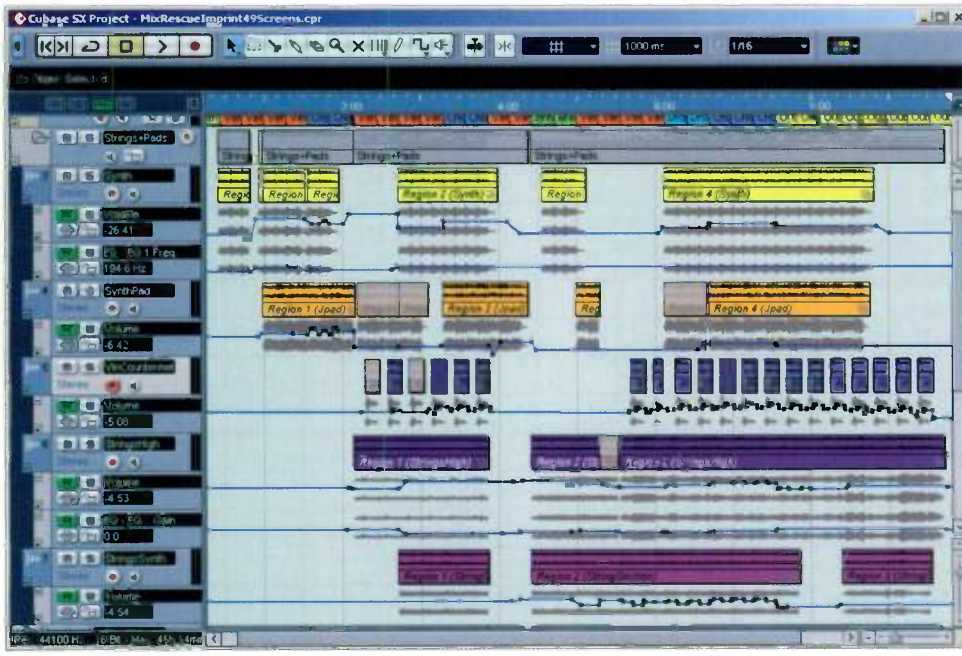
As luck would have it, the very first loop I tried slotted in perfectly and added a nice hint of backbeat. I used the hi-hat and percussion parts from the loop's construction kit, panned them half-left and half-right, and then compressed them both fairly hard to smooth off the transients and push the sounds into a background role. High-pass filtering at around 280Hz and shelving off a couple of decibels at 12kHz highlighted just the required frequencies. I inserted instances of the Sidekick plug-in to get a more pronounced ducking effect on these percussion tracks, and then sent to the quarter-note delay and distorted reverb effects for a bit more complexity.

Balancing Synth & String Parts

The string and synth parts took very little processing, other than a few decibels of low shelving cut between 200Hz and 450Hz on the different tracks. I did compress the lower

Compression and saturation treatments were used to get the main drum loop sounding as loud as possible, but this made a high-frequency percussion element in the loop over-prominent, so a Digital Fishphones Spitfish de-esser was set up to bring this back into balance. A little broad EQ boost around 800Hz was also dialled in, to emphasise the 'knock' of the kick drum's attack.

Although the synth and string parts required only a little bass EQ cut before they worked with the drum and vocal tracks, a fair bit of level and EQ automation was required to keep the parts in a suitable balance throughout the track. The arrangement of these parts already created quite a lot of changes in texture, but Mike also muted some further sections to help improve the overall dynamics of the mix.



violin counter-melody a little to keep it under control, though, and both synth parts also benefited from some thickening and brightening using Rubytube. I sent to a selection of the different vocal delays and reverbs, depending on which track suited which sound, aiming for a smooth blend of the different parts, and I set up Cubase's Phaser plug-in as well, to smooth some of the parts even more, making sure to set the effect's feedback to zero — I wanted just a gentle hint of modulation, rather than an obvious phaser sweep.

The biggest hurdle to get over with the strings was their balance throughout the

Remix Reactions

Sin: "I noticed a HUGE difference across the whole track straight away and was blown away by what Mike had done — it actually gave me goosebumps! When he sent us the first draft it was almost there already, but when we'd sent him back what we thought needed tweaking it arrived back sounding unbelievable.

"The vocals are now heard instead of being lost within the song, and the effects on them are perfect. The drums are punchy without being over the top, the strings sing and soar, and the bells finally sound like bells... All the little extras Mike layered in really add to the final result. It was always going to be tricky getting the balance right between the delicate parts and the more aggressive parts, as there's quite a lot taking place within the song, but now 'Let Me Go' sounds like it should.

"All in all this has been a huge learning experience, and one that I have really enjoyed. Comparing the original with the new mix, you can hear many, many differences, and while I think both mixes have their own merits, I have to say I prefer Mike's. I wanted this song to soar, and now it does!"

Blink: "The first thing you notice is the vocals, and the effects on them, which add a ghostly quality. A lot of the rhythmic sounds have been made fuller and a lot more immediate. When the drums kick in, they are very forward, which is good, as it retains the aggression in the song, although I still feel that the distorted drums could be brought forward a bit more, as a counterpoint to the main drums.

"The effects and reverbs add more texture to the song, without necessarily taking away the focus, and the use of the 'rhythmic rumbles' is very in keeping with what we were aiming for. I also like the addition of the snare sound, which fits well with the rest of the song. In general the whole mix sounds a bit warmer than the original, probably due to the added textures, although personally I also liked some of the cold sound of the original as well."

track, as the changing string arrangement was the primary source of mix dynamics, and keeping a good balance and level for the strings throughout these changes required quite a lot of fader automation. In addition, the drum parts also masked the frequencies of the strings in different ways as the track progressed, so I had to do some surreptitious riding of the low EQ gain controls as well, in order to keep the sound subjectively consistent.

Sound Effects, Bells & Piano

Turning to the remaining tracks, I first dealt with the two sound-effect hits which are most audible during the opening section: one a burst of distortion panning from side to side, and the other a huge reverberant drum. I compressed these a little to add sustain, and gave the drum hit more weight with a 7dB peaking boost at 70Hz and a 5dB high-shelf cut at 3.6kHz. A coating of the vocal reverbs and delays brought the sound effects into the space occupied by the rest of the parts, the cathedral impulse working particularly well in this case.

The dry bell track sounded, frankly, more like a rather cheesy door-chime than a death knell, partly because it had far too distinct an attack transient — it felt like you were standing in the belfry right next to the bell itself, rather than contemplating your mortality from the crypt. Some limiting from Cubase's Dynamics plug-in soon remedied this, while low-pass filtering moved the tonality into the right ball park, and a deep notch at 2.4kHz killed a couple of over-harsh resonances. Lashings of the cathedral reverb and a bit of the ambience and half-note delay completed the picture, although I had to reduce the modulation of the delay effect a little, as it was making the bells sound rather out of tune.

The piano sound needed fairly stiff processing to make it fit within the track, as the timbre wasn't particularly attractive, and very woolly into the bargain — so much so that I broached the idea of overdubbing a new piano part with the band, although in the end I figured that a lo-fi sound was probably more suited to the style of the track. A big 8dB peaking cut at 250Hz, combined with a 6dB shelving boost, made a useful difference, but the sound still proved very difficult to mix — weird resonances were appearing and disappearing all over the place in a very uncontrolled way. Crushing the sound with a rather ludicrous four-band limiter setting in Cubase's Multiband Compressor plug-in allowed me to salvage something more stable (if not exactly pleasant-sounding), and I could then sink this into the mix with the vocal delays and reverbs.

Adding Atmospherics

The balance of the tracks in general was now working pretty well, so I took the opportunity to ride the lead vocal fader to keep the lyrics as close to the surface as possible. Some EQ rides were also necessary to compensate for frequency-masking as the arrangement changed, and I adjusted the effect send levels to suit different song sections and to pick out the ends of some phrases. An instance of GVST's GSnap pitch-correction plug-in (www.gvst.co.uk) came in handy too, tightening up the odd wayward intonation moment — inevitable when dealing with a one-take wonder such as this. I don't always like the way automatic pitch-correctors sound out of the box, so I automated the GSnap settings to keep the sound as natural as possible.

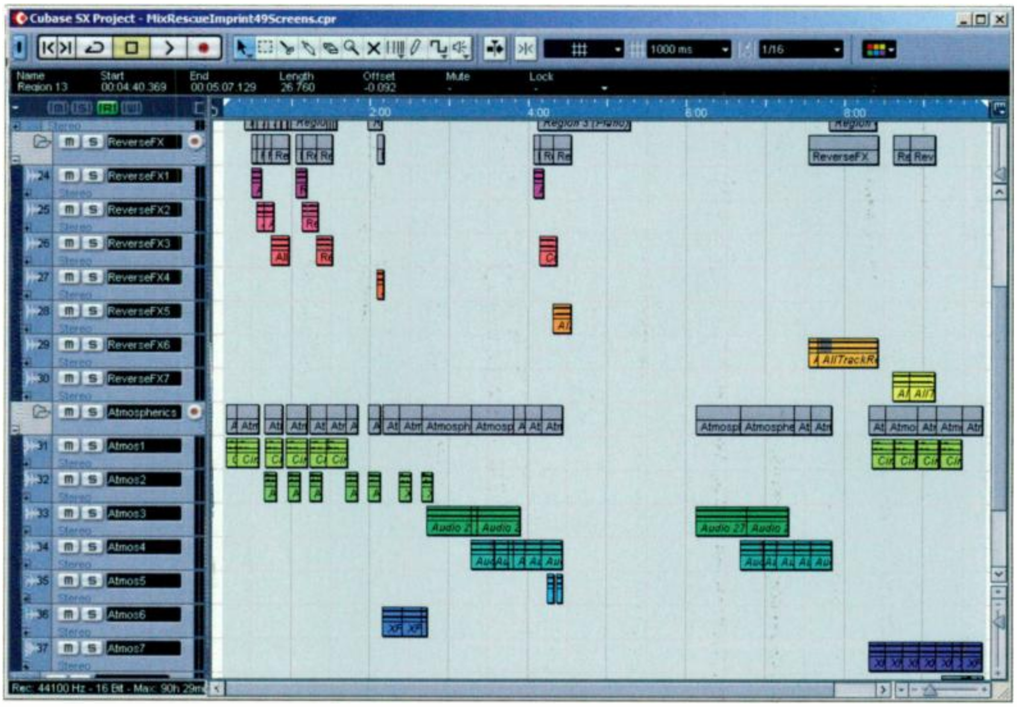
Although I was becoming increasingly happy with the mix from a purely sonic perspective, I couldn't help feeling that it wasn't yet involving enough, in terms of emotion — despite the space and depth implied by the effects, everything just seemed a bit empty and two-dimensional. So, although it involved stepping a little outside the normal Mix Rescue brief, I resolved to try pruning a few bits out of some of the tracks to give a bit more contrast and build-up, and also

Fast-attack compression and limiting were combined to round off the over-spiky attack transient in the bell part and give it more of a distant feel, and some judicious EQ cuts also helped soften the tonality too. Copious reverb and delay completed the job, leaving the sound much more ominous and in keeping with the track as a whole.

to experiment with adding in some extra atmospheric background samples from a film sound-effects sample collection.

Starting with the introduction (before the drums enter), one of the first things I did was underpin some of the existing sound-effect hits with a low, thunder-like rumble, as well as layering in a metallic swell which you can hear for the first time after the lyric '...to find my peace'. Then I recorded the output of the cathedral reverb using Silverspike's Tape It utility plug-in, reversed the audio, and reimported it into my Cubase Project, editing a few bits into the gaps between vocal phrases. With the help of the odd pitch-shift and a bit of high-pass-filtering, it was possible to achieve a nicely disconcerting and evolving backdrop. In tandem with this I began chopping out various audio sections to try to give a bit more light and shade in the arrangement.

I extended these activities to the rest of the track in a similar way: the drums had a couple of different textural loops introduced behind them during certain sections (one a kind of demonic whispering and the other comprising swirling brushed metallic scrapes); the outro was suffused with a subtle machine-room drone; and various reverse reverb snippets were scattered around the place willy-nilly. One other thing I did was loop a small section of the vocal track to create a disembodied vocal pad in a couple of places — you can hear it for the first time just under five minutes into the track.



To give the mix a bit more depth and texture, Mike added in a selection of background sound effects, as you can see in this screenshot: the upper folder of tracks comprises a variety of snippets of a reversed recording of the cathedral reverb, while the lower folder contains a variety of atmospheric film sound effects.

First Draft, Feedback & Final Tweaks

As usual, I sent off a first draft of my mix to the band for their feedback, and fortunately they seemed pretty happy with where I had taken the song, although they still wanted a couple of things tweaked. You can hear from the original mix that the distorted drum layer really makes its presence felt, and mine wasn't giving them that same jolt, so I pulled up the level a bit and added in a bit more of the distorted reverb effect to get this to fit the bill a little better.

The other thing they mentioned was that the strings needed to be more 'sweeping', and, on returning to the track after a couple of days' break, I could see what they meant — I'd overdone the low cuts on the string parts (a common foible of mine), leaving them a bit thin. This was fairly easy to remedy, though, with a couple of EQ adjustments.

For my own part, some referencing during the adjournment had isolated a different problem, which was that the lead vocal still had a rather piercing edge to it when Sin used her chest voice. I was unable to get any better results using the Multiband Compressor plug-in, so dealt with the issue more surgically using the Platinumears IQ4 dynamic EQ plug-in (www.platinumears.com/IQ4gui.zip).

Dynamic EQ is a powerful process, which combines the functions of compression and equalisation such that the gain of each equaliser band changes in response to the incoming signal in the same way that a compressor's gain-control element does. For this application, I only needed to use one band of the EQ, tuned to the most abrasive frequency (which turned out to be 8kHz). I set the threshold so that the band's gain reduction only triggered for the chest-voice notes, and chose a ratio which just reined in the problem. The Q, Attack, and Release parameters were then refined by ear for the most natural result.

One last thing bothering me was that the reverbs and delays (which took up almost as much space in the mix as the instruments!) were making the overall tonality a little dull, so I went though the returns one at a time, bypassing them to see which were the main offenders, and then applied a few low mid-range EQ cuts to those channels where I though it was necessary.

Mission Accomplished

This mix illustrated a number of methods that you might experiment with in your own mixing. For example, the processing I used to beef up the drums in 'Let Me Go' can be transferred to many different styles, although it'll always be a question of degree. A wide dynamic range is not uncommon in vocal tracks either, and some of my tactics on this track might well bail you out of a tricky situation when compression alone can't deliver the goods.

One of the trickiest things with very reverberant styles of music is keeping the mix sounding clear, and delays can be more useful than reverb here. However, there's a lot you can do with reverb pre-delay settings, EQ in effects returns, and effects level automation as well, so remember to give these a shot too. Finally, if your mix has comparatively few musical parts, it's often surprising how a bit of extra background atmosphere can make the final result sound more finished. And it's not rocket science finding suitable sounds for this — any budget film sound-effects library should have plenty of suitable material. SOS

Hear The Differences For Yourself!

If you want to compare Mike's remix with the band's original version, or fancy checking out some of the remix processing in isolation, you can find a series of audio examples at http://www.soundonsound.com/sos/jan08/articles/mixrescueaudio.htm

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Simon Posford & Benji Vaughan

Younger Brother

Paul Nagle

Younger Brother brings together two of the Twisted label's biggest names. We visited Simon Posford's studio in a bid to uncover the secrets of psytrance...

The names Simon Posford and Benji Vaughan may be unfamiliar to some *Sound On Sound* readers. After all, their background is the hazy world of psychedelic trance, a musical by-product of the LSD and ecstasy-fuelled parties and raves of the '90s. With its roots deeply entwined in Goa trance and, to a lesser extent, acid house, Psytrance remained an underground scene even at its zenith. Despite this, Posford's first album reached number 27 in the French album charts, and over the next decade his Twisted record label established a reputation as a source of high-quality dance and electronic music. Foremost amongst its artists was Shpongle, the collaboration between Posford and legendary innovator Raja Ram that straddles so many boundaries there's surely a UN charter about it.

Benji Vaughan is best known for his foot-stomping trance identity Prometheus, although his long list of credits includes performing remix work for EMI and Jive, scoring an advert for Sony and producing electronic funk band The Egg. Benji and Simon work together as Younger Brother, a partnership characterised by high production values, lush melodies and intricate arrangements. As their second album, *The Last Days Of Gravity*, was released in October 2007, we thought it high time we floated on down to see them.

Youth Before Beauty

Simon Posford's musical career began as a teenager, when he had the choice of either going to Oxford to study botany, or going to work at Virgin recording studios. The decision to go to Virgin as tape-op was a no-brainer, especially as it offered an opportunity to learn the ropes alongside such luminaries as Spike Stent. For a while, Posford rotated round the various Virgin studios — the Town House, Olympic, Town House 2 and the Manor — but the life of making tea and dealing with fevered egos couldn't go on indefinitely. Things came to a head with the band James, who had recently finished touring but had not yet abandoned 'tour mentality'. It was, Posford recalls, an absolute nightmare.

Seeing his unhappiness, Stent took him aside and told him that the producer Martin 'Youth' Glover was looking for someone to work for him at his Butterfly Studios. The offer was sweetened further as it offered the opportunity to work on some of Posford's own musical ideas. "The urge to make my own music was just too strong and so, wondering who this mad hippy Youth was, I went to work for him in 1991 or maybe '92. Prior to that I'd learnt a lot from watching these top engineers like Spike, Dave Bascombe, people like that, but now I had hands on — I was the engineer.

"Apart from sessions with the KLF, I hadn't really seen much electronic music at Virgin, it was all bands. But now I was starting to go to acid parties and it was from there I got into electronic music. I remember getting my hands on a sampler for the first time — an

Photos: Richard Ecclestone

Above: Simon Posford's studio is based around a Mackie 8-bus analogue mixer.
Left: Younger Brother: Benji Vaughan (left) and Simon Posford.

Akai S950. The very first sample I took was this 30-second chunk of Ozric Tentacles. I simply tuned it down two semitones, slotted it into a tune I was working on and it fit perfectly. If only all samples fitted in so well!"

In contrast, Benji Vaughan found he preferred the other-worldly tones of Aphex Twin and the Orb to the music he was hearing at acid parties. It was only after a trip to Goa that he threw himself fully into trance.

Younger Brother Live

The night before my visit, Simon and Benji had taken Younger Brother out to play in Soho, aided and abetted by some top session musicians. Simon explains: "The gig was fairly chaotic. The stage was tiny and we didn't have the bass player, so that came from computer along with some of the synths and backing tracks, all in separate channels to the mixing desk manned by Benji. I took a guitar and a controller keyboard and played the ImpOSCar rather than the real one. I also played a Roland SH101, which is such a great synth live. It's so nice to just look at the knobs and know what's going to come out."

The intention is that Younger Brother will do more gigs together as a live act and try to get another album together before everyone goes their separate ways; being in-demand session players, Posford is aware they might have a narrow window of availability. "Take Andy Gangadeen, the drummer — he's absolutely fantastic. The hardest thing with mixing live and electronics is the drumming. If the drummer is even slightly out it just sounds like the whole thing is falling down the stairs. Andy is metronomic and enjoys it. He's not one of these drummers who hates the idea of a click track or wishes he was in a 'proper band'. We could also lose Matt White, the guitar player, as he plays with a few bands and does session work — as does Ruu Campbell. We've got them all together, we're playing together, so we should record!"

Asking whether improvisation is a factor in live work reveals Simon's more production-based approach. "I quite like things to be tightly scripted. I was surprised to read how tightly scripted The Office was, for example. It feels loose and improvised, yet isn't. So we don't improvise so much, but increasingly with Ableton Live we're heading in that direction. We may start like the CD and towards the end we can go off. Once we leave the backing track behind the band can improvise, we can jam out the ends. At the moment the backing track is too complex and we keep it in Arrange mode rather than Session mode. There's a lot going on. Eventually we may use it in Session mode, so for a future album we might work with Ableton and the band that way. We don't want to get too free-form though, end up doing jazz!"

Those early experiments were with basic equipment — an Alesis SR16 drum machine and an Akai S01 sampler — but they paved the way for his first release, the trance classic 'Clarity From Deep Fog' with Sean Williams. More collaborations followed, including Process (with Williams), Citizen Kaned with Nick Doof and Cyber Babas with Raja Ram. He met Simon Posford while delivering essential 'studio supplies' and has been associated with the Twisted Label (formed in 1996 by Posford and ex-Dragonfly manager Simon Holton), ever since.

Hot Desking

An hour out of London, tucked away in quiet woodland, is Posford's unassuming home studio. Here, Speak & Spell machines are strewn casually amongst top-class outboard gear, and it's immediately apparent that comfort and a relaxed working environment rate higher than control-room acoustics or laboratory conditions. From the first Hallucinogen album to the latest Younger Brother release, all of Posford's output has been recorded on a Mackie 32:8 analogue desk. "I've had one since they first came out," he says. "I must admit I don't really like it but I can't complain too much — people do say our production is very good. I'd love a big desk — not digital — something like the TL Audio valve desk. I really like the SSL thing, A nice collection of vintage analogue keyboards includes Roland SH5, OSCar and Korg MS20 synths.

the half-controller, half-desk SSL AWS900. Maybe there aren't enough physical aux sends, though, and they only do a 24-channel version.

"I haven't mastered this mixing-in-the-box thing. I love mixing desks, I love the feel of them, the touch of them, feeling the faders. When something needs a bit more top, you reach over and the knob's there and it's done quicker than I can say it. On the computer you've got to find the right page, you've got to select the thing, you do it with the mouse or even a controller, and by the time you've actually done it, I could have done 10 other things on a mixing desk."

Ergonomics aside, though, Posford has no objection to the sound of software processors. "In the last two years, plug-ins have really started to come on. I mean the SSL stuff in Waves would take some beating in the analogue domain — you'd need a pretty serious desk. And a lot of stuff coming out on the UAD card, the Neve stuff, is absolutely fantastic. So in the end we use both plug-ins and hardware. Even though on this mix we EQ stuff with SSL plug-ins, it comes through the Mackie, where I might EQ it again. The desk is still a tool — and a valuable tool."

Benji Vaughan, by contrast, has been thinking of getting rid of his desk. "The thing that gets me most is when I'm working on a few things at the same time. I hate that feeling when I've committed to the desk. There's 32 outputs and all these outboard effects and you have to pull it all out to work on something else. For the last few months we've been editing this band, so I've been bringing it down here without committing to any outboard because I want to pass the files over to Si. I'm doing it all inside my laptop with the Waves stuff. The trouble is it never comes up right, to be honest. We both use Logic, we have the same plug-ins and still it doesn't come back correctly. Who knows why but it just doesn't copy plug-in settings; all the SSL EQs come up but they aren't set to anything."

Brotherly Love

Younger Brother was born after Survival International, a charity supporting indigenous tribes, requested a track for a forthcoming compilation album. They asked Si and Benji to work together, not even realising they were on the same label. Posford explains how the name came about. "They offered us this tribe called the Kogi from Columbia as our inspiration. With a completely different time perspective to ours, they even refer to events such as Columbus arriving as if it happened only recently! Their view of the world is that they are the 'older brother' and their ecosystem a microcosm of the entire planet. They refer to us as the younger brother — hence the name — and when they started noticing things going wrong, the water changing, wanted to tell us 'stop fucking up the world!'"

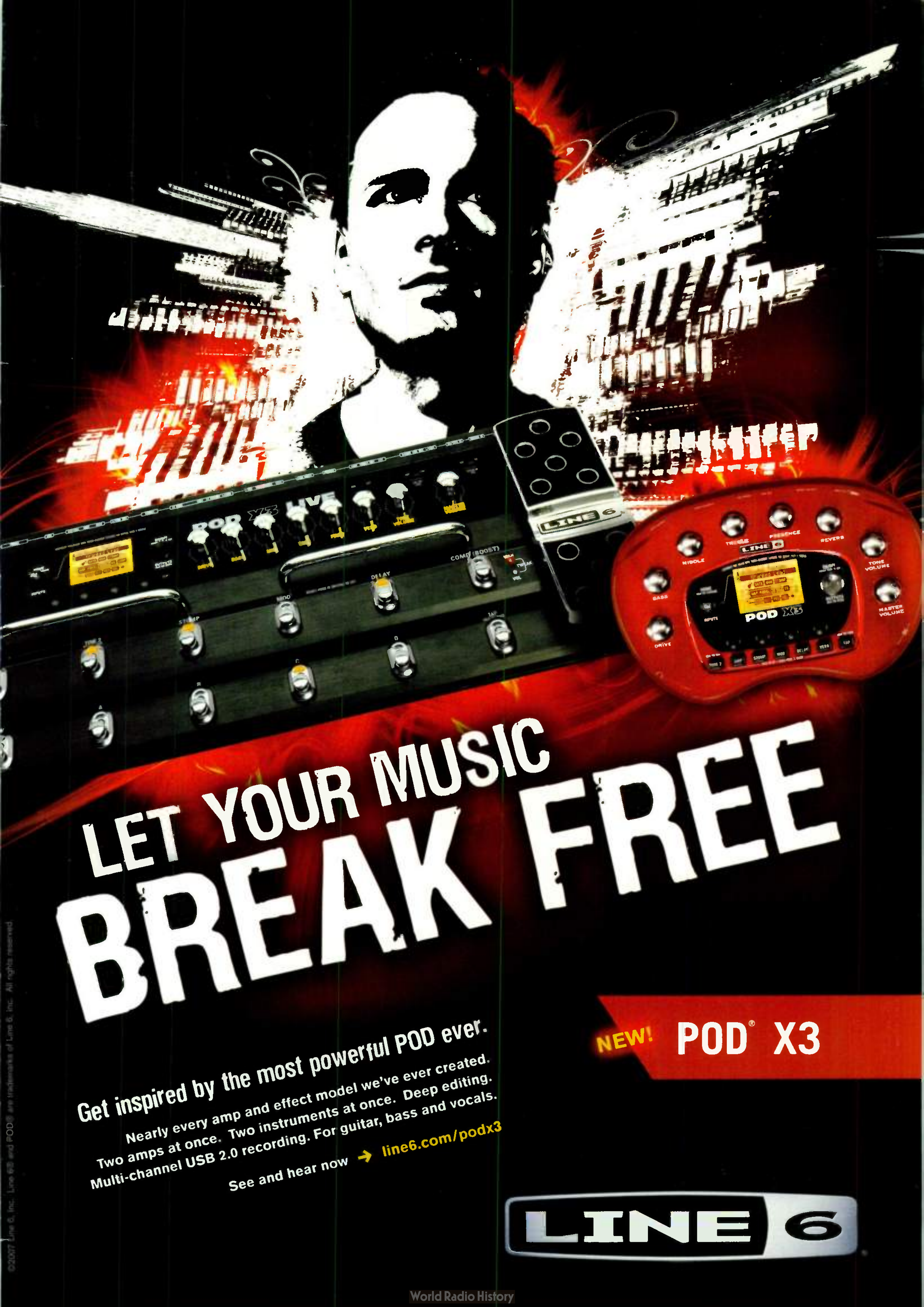
Logic Bomb

If mixing via DAW is subject to ongoing debate, there is at least agreement on the sequencer of choice. Both use Logic, although they contend that even this isn't perfect.

Simon Posford: "Logic is our main workhorse. The main feature we still want, and we hear people bang on about it all the time in *Sound On Sound*, is 'bounce in place'. So many times we've set up edits of audio on a track with loads of plug-ins and we want to bounce it to an audio file and use that in our arrangement. I counted the amount of clicks you have to do to achieve that in Logic and it's over 30 or something — utterly ridiculous! You bounce it, then import it into your audio window, then drag it out of the audio window back into the arrange, make a track for it, delete all your plug-ins, delete the old track that was there, take the parts off, you know... I'm falling asleep just talking about it.

"Every time a new version comes out it's got new graphics, which I really don't care about. I'd love it if, instead, they spent their time updating the time-stretch algorithms. In each version there's all this stuff that seems to have been there for 10 years but now has a new interface.

"You'd think that by now they'd incorporate some way of quantising audio that wasn't a complete disaster. Ideally you'd click on an audio part in Logic and it'd bring up a matrix like a MIDI part and you could move it around. It's key to music to be able to take audio and



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put it in time efficiently without cracks and clicks everywhere. If Melodyne can do it, if Ableton can do it..."

Last Days Of Gravity

The second Younger Brother release is an assured and diverse collection of tracks with, it seems to me, a decidedly band-like feel. I ask Simon if this was intentional.

"The first album was more centred around making a sound, then basing the song on that. When it was complete we found we wanted to do a proper album, without worrying about whether it was trance or whatever. We had a vibe and a direction and created sounds to fit in with that. We went back to our band roots — something we could do because the technology had changed so much. It's extraordinary to think that when we started Younger Brother, we were still using Akai samplers, which made putting on big chunks of guitar much more complicated. Now, years later, the whole process has changed and the dividing line between an electronic album and a band album has completely broken down. We now can be a band, just two people."

In his band days, it was always Posford who wanted to have a go on the others' instruments. "That's how I learnt everything. Back when we started, if you put in a slightly dodgy guitar part then it remained a dodgy guitar part. Yes, you could edit it, but it took hours. Whereas now you can get ideas and emotions down from an instrument and then fix it up a bit with audio quantise, without killing it.

"One of the rules for this album was, where possible, we wanted to do it all ourselves. We got in Gerry Hogan to play slide guitar and Ruu Campbell the singer, but otherwise we played the drums, the bass, the guitar, the keyboards."

From its wistful opening track, 'Happy Pills', with its lush pads, driving percussion and swirling synths, to the New Order-esque guitars and deeply processed vocals of 'Psychic Gibbon', *The Last Days Of Gravity* is varied beyond my ability to categorise. Its unusual time signatures and shifts of perspective suggest progressive rock at its most imaginative, and I reckon there's a spaced-out indie band tucked away in there too. Most surprising is the lack of any obvious dance tracks, perhaps because the drumming — although tight and effective — never loses its natural, human feel. And, as ever, the synthesizers are splendid throughout.

"On the album we used [*Native Instruments*] Reaktor, the Roland V-Synth, the Korg MS20, Roland SH101, a Mellotron, Macbeth M3X and the OSCAR — there's OSCar all over it!" says Vaughan. "And the M3X — Ken MacBeth makes crazy stuff. The sound of it is just gorgeous, so warm and silky and creamy, plus it's got that heritage; you want to buy it because you know he's some crazy guy up in Scotland knocking it together.

"Whenever fans get in touch they always ask if we use a Virus, as they say their dream is to have one. I've had two of them and neither worked. The potential is very good; as a synthesis engine it is quite powerful, but the TC Powercore one didn't work and the Virus Indigo crashed, which is the last thing you need in a keyboard."



Unique and circuit-bent gear is popular with Younger Brother.

By contrast, Posford's enthusiasm for the M3X nicely sums up Younger Brother's approach to music creation. "What we're after, and what musicians should strive for, is uniqueness," says Posford. "That's where hardware scores. You're getting a bit of uniqueness, a bit of character that software can't give you. Look for those weird old effects, guitar pedals, old synths, things that don't work properly. Or circuit-bent stuff. We've sent quite a bit of gear to CircuitBenders.co.uk for modification. Take the Alesis drum machine: it is quite uncontrollable, unpredictable and has these unlabelled sockets that you patch together so it glitches up in different ways. Some of the options decrease the bit rate or mix the samples so it gets really crunchy. Sometimes it totally crashes the machine and you have to power off and start again. We used it on the tracks 'Psychic Gibbon' and 'Elephant Machine' for that really dirty, lo-fi sound."

"The Voyager I quite like," says Posford. "I got it after using a Minimoog in a gig. It just sounded so fat and the Voyager is close. Each Minimoog sounds so different, though, and this wasn't quite like the Minimoog I was hoping for."

Hardware Versus Software

Noticing both a real Korg MS20 and the Legacy software version, I couldn't help asking Simon for his first-hand comparison. "The little thingy I never use. I was really hoping

PRODUCING PSYTRANCE

Simon Posford: "The Eventide stuff is an example of something purely digital that I've never heard any plug-in get close to. We may start with something really gritty, say made with Buffer Override, that comes out sounding all 8-bit, then the Eventide poshes it up. Makes it a massive, stereo, beautiful sound again. The Distressors are brilliant, my latest addition and one I'm really pleased with. I met the guy from Empirical Labs at the AES convention and he was like John McEnroe on PCP."

▶ for an easy replacement for the MS20; it didn't even have to sound as good, but I wanted something that was easier, as I use its filter a lot. But it was no fun, no joy; it all sounds the same. Put stuff through the real MS20 and you never know quite what you're going to get; the way the filter distorts is always changing and very different from the plug-in. We used it a lot on the CD, often just for the filter and modulation side.

"As much as software is brilliant and clever and can do fantastic things, I just find hardware so much more inspiring. It's like when you sit at a piano, it makes you want to play and write songs. I never think of sitting down at a controller, loading up a soft synth patch and playing. You don't feel like reaching for the mouse to select a new sound when you can reach over and grab a knob instead. Hardware is more fun but you get the uniqueness too.

"In the early days of software instruments you could say hardware always sounded better. I don't know if that really applies any more, but hardware certainly sounds more unique and characterful. With software generally it's good that you can get so much — a compressor on every channel or whatever — but with synths it's best when they stop trying to emulate. Something like Reaktor is absolutely fantastic, but when you go through magazines and see what's coming out, it's page after page of emulations, even of gear that's already been emulated by somebody else. Another Minimoog, another Odyssey."

Warped and often outrageously time-stretched vocals are a Simon Posford trademark, and I confess one of my ulterior motives for nabbing this interview was my desire to uncover these secrets. As is so often the case, it turns out that these effects are not attributable to any one, easily lifted process, but are the result of painstaking work with accumulated plug-ins.

"Sorry, but there's no secret formula. You might use something to get it in tune, then something else to fuck it up; then you might change the formant or something like that, then chop it up to get it in time. It's really all about editing and graft. We often employ lo-fi solutions such as putting it into the [*Korg*] MS20, which has a frequency-to-pitch converter, so you can add some analogue into the equation and mix it together."

Are You Shpongled?

Ultimately, one theme that Simon Posford keeps returning to is the fact that technical wizardry can't make up for a paucity of musical ideas. "I think that more people are now coming to music from computer or DJ backgrounds and less musical ones. It's a bit like photography, where in the old days if you wanted a photograph, you'd hire a specialist and he'd come round with this thing that looked like an accordion, put a towel over his head, set the equipment up and take an amazing photograph. Nowadays everyone has got a camera — but that doesn't mean I want to see everybody's pictures. There's still a market for good-quality photographs, and it's the same with music. The specialists will always stand out." SOS

Twisted Brothers

Founded in 1996 by Simon Posford and Simon Holton, Twisted Records is an underground or indie label, making it all the more remarkable that the first Shpongle album, *Are You Shpongled?*, sold in excess of 30,000 copies — mostly by word of mouth at gigs and parties, and via on-line forums. Twisted features some of the world's top trance acts and is a hotbed of cross-pollination. Artists include Shpongle, Hallucinogen, Celtic Cross, Younger Brother, Prometheus, Ott, GBU, Tristan, Koxbox and more.

Contemplating the future of the music business, I asked Simon whether he would be following Radiohead's example of selling music directly on a 'pay what you like' basis. "It's very nice to be able to record your album in top studios, pay off all these top engineers and producers, then give away your album for free. It's brilliant, actually, because it's cracked open the music industry in one week. At the same time, people are paying. Whether this would apply to mere mortals such as ourselves I don't know. *SOS* readers and we who are trying to make a career out of music would find it pretty risky. The important thing is still publicity and that comes from money spent. When you get a record deal, you get an advance and you then know you don't have to go out and get a job, be an accountant or whatever, for the next year. You can make a record after all."

Twisted Plug-ins

"We'd love to put out Twisted plug-ins," says Simon Posford. "We come up with new ideas each time we make a track. So if any programmers want to get in touch, give us a shout! We'd love to put them out for £15 or something from the web site."

"If companies want to stop piracy they just have to make plug-ins cheaper," says Benji Vaughan. "Like Waves, which are heavily pirated, if they just made them cheaper and you could select what you want, people would support them. I believe that. It's the same here at Twisted — the fans want to buy the stuff if they can get it. I often find piracy can be good. If I use something and like it, I want to have it forever, get all the updates and not have it cocking up on me, so I buy it. All the software I buy I've first tested out as a pirate."

Posford adds: "For the first time we want a hardware version of a plug-in: Buffer Override, a free download from DestroyFX, which is a glitchy machine that takes your audio and cuts bits out, screws around with the audio buffer, loops and repeats and is quite unpredictable. You get great sounds out of it and to have that in a guitar pedal would be fantastic — so if anyone can make that for us, please get in touch! That's what manufacturers should be striving for — to make different things. With all this computing power, stop emulating and do new things we've never heard before. We want to hear effects you could never have done before computers! Often the ones that come out and fulfil our brief are from small guys just pissing about, rather than from a large company."


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Earthworks TC20

Omnidirectional Condenser Microphone

Paul White

This high-definition omni mic has been designed specifically to work in high-SPL environments, but it is surprisingly good on quieter sources too.

Earthworks produce what they call High Definition Microphones, which are designed for accuracy rather than colour or character. They produce several ranges, many of which have a similar physical appearance (akin to a headless electric toothbrush!) to the TC20 under review here.

This microphone is one of their least expensive, and is designed for applications where the sound source is relatively loud, such as drums or brass. It is available singly or in matched pairs, and I receveived a pair for review, which came in a nice wooden case along with plastic stand-clips.

Design & Construction

Offering a fixed, omnidirectional pickup pattern, the TC20 is intended for both live and studio applications, but because it is designed specifically for high-level sound sources, it may be rather noisy to use when recording quieter instruments. In addition to the applications mentioned above, the TC20 is recommended for use under the strings of a standup bass and inside a kick drum, in place of the more conventional choice of kick-drum mic, and it is also well suited to miking guitar and bass speakers.

The designer behind these mics is David Blackmer, an engineer well known in design circles as the inventor of the highly-specified Blackmer VCA, which is used in many dbx processors. David developed a number of new technologies for use in Earthworks microphones, which, in combination with the very small diaphragm, deliver improved transient handling, with less mechanical 'ringing' or settling time than conventional mics. This can only be achieved by building a capsule with a very wide frequency response that extends well beyond the human range of hearing: a typical Earthworks microphone operates up to 25kHz and beyond. A class-A balanced output stage, capable of driving long cables, completes the signal path.

Mic design is a bit of a juggling act, because the smaller the diaphragm, the less audio energy you collect — and so the more gain you need to raise the signal up to a usable level. However, small diaphragms behave more like the theoretically perfect single pickup point than do large ones. They therefore have better-controlled polar patterns, and produce less acoustic shadowing of the sound being recorded. So there's a choice: you can have a small diaphragm with a beautifully-controlled polar pattern and a great transient response, but at the expense of some increase in background noise; or you can use a larger diaphragm to get lower background noise, but pay for this in accuracy. Accuracy isn't always a key factor (many large-diaphragm studio mics are chosen for their flattering characteristics rather than for their fidelity) but where it is vital, small diaphragms win every time. The usual compromise is to build a 'stick' mic with a capsule around half an inch in diameter, but

SOUND ON SOUND

Earthworks TC20 £359

pros
- Detailed sound that lives up to its high-resolution tag.
- High SPL handling, with excellent transient response.

cons
- This particular model is a little on the noisy side of average.

summary
As a mic for loud sound sources that include a lot of transient detail the TC20 is really very hard to beat.

Alternatives

While there are numerous small-diaphragm mics that make a good job of recording the same kind of sounds, few other manufacturers take the small-diaphragm approach as seriously as Earthworks, other than in the area of measurement microphones. When it comes to fidelity, obvious rivals are DPA and Sennheiser's MKH-series microphones.

Earthworks use a much smaller capsule and work hard on the electronics to minimise the noise.

I mentioned that the TC20 is one of the least expensive Earthworks mics (it is their cheapest omni), but it isn't exactly 'budget'. The lower price has been achieved by designing the mic for high-SPL operation, which means that self-noise isn't such an issue — the 27dB SPL noise figure specified here would be unimpressive if applied to a general-purpose mic. Furthermore, the frequency response isn't quite so extended as with some other models, though it remains essentially flat from 10Hz to 20kHz, ±2dB.

The maximum SPL handling of this mic is a massive 145dB, with a sensitivity of 8mV/Pa (that kick drum recommendation was no idle boast!), but, because of the somewhat high noise figure, this isn't going to be the best choice for recording very quiet or distant instruments or ensembles.

Studio Test

True to its claims, the TC20 is quite happy sitting over percussion, where it picks up a very detailed and articulate sound that feels more 'in focus' than what you get from most conventional mics. I tried it on a range of ethnic percussion, including djembe, dharbula and balafon, and it was the audio equivalent of wearing very clear reading glasses: it gave a good, crisp picture that really brought out the transients.

Despite its highish noise floor, I also decided to use it to record a lute, to see just how well it could perform on quieter sources. I ended up with a mic distance of around 10 inches from the lute body and was rewarded with a recorded sound that was just slightly brighter than how I felt the lute sounded acoustically, with fantastic definition to the string plucks. I repeated the test using another favourite omni mic, which gave a sound that was tonally closer to what I heard in the room, but without the sense of super-focus that the TC20 gives you.

On playback, the self-noise of the mic was only audible at playback levels that were louder than normal, during pauses where the instrument wasn't playing. Subjectively, I'd say it was less than I've heard with some Far Eastern mics that have claimed noise figures of 20dB or better.

Having used Earthworks mics on a few occasions now, I can see why some engineers become really obsessed with them. Of course, a mic like this really excels as a drum overhead or percussion mic, but it is far more flexible than you might imagine, and it doesn't produce a clinical or unexciting sound just because it is accurate. I wouldn't buy a mic like this specifically for recording quiet acoustic instruments, but if you have one around, don't exclude it: you might just be pleasantly surprised. SOS

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RECORDING ACOUSTIC PIANO

mechanism is often more prominent out back as well, so that position's likely to be a duffer most of the time.

On the other side of the piano, the lid affects the dispersion of high frequencies in a different way, bouncing them out towards the audience. Roughly speaking, it does this most effectively along the line of the instrument's hammers, so a mic placed on that line will capture pretty much the brightest sound. As the miking position moves towards the foot of the piano, the lid gets less effective at reflecting the highest frequencies, and the sound loses some of its airiness. A similar effect occurs for mic positions on the keyboard side of the instrument, behind the player.

At this point it's time to turn to my first set of audio examples. You can access these in both WAV and MP3 formats on the *SOS* web site at www.soundonsound.com/sos/jan08/articles/pianorecordingaudio.htm. (Further information about the recording sessions can be found in the 'Recording The Audio Examples' box.)

To illustrate the range of tonal variation I've been talking about, I recorded six identical omni mics around the piano, all angled towards the instrument about 1.5m from its centre and 1.7m off the ground, as you can see in the photo on the previous page. The recordings can be heard in the following audio files:

- **HorizDispLidOpenMic1:** The first mic was on the keyboard side of the piano, behind and slightly to the right of the player to try to avoid high-frequency shadowing of the upper strings by his body.
- **HorizDispLidOpenMic2-5:** Mic two was in front of the piano, directly on the line of the hammers; mics three and four were positioned progressively around towards the foot of the piano; and mic five captured the sound directly at the foot of the piano.
- **HorizDispLidOpenMic6:** The final mic was positioned behind the piano.

Although it's possible to hear general trends in the dispersion of the very high

Miking An Upright Piano

I've focused in mainly on recording grand pianos, and for reasons of space haven't gone into detail on recording uprights. However, many of the considerations are the same for both types: you still need to think about the distance and height of your mic placement. There are some differences, though. For example, unless you can take some of the panels off the instrument, the only way to get access to the strings is to open the lid at the top and mic from above.

If you're using a spaced stereo technique, extremely close miking is likely to risk a hole in the middle of the stereo image, even if you're using good omni mics, so it would also make sense to space the mics more closely than you might when close-miking a grand.

One other thing to bear in mind is that the very characteristic which helps spaced stereo techniques to give grand pianos a more diffuse and blended sound also tends, in my experience, to emphasise the 'honky-tonk' element of an upright's sound, so you might prefer to go for a coincident technique if you hoping for your upright to make the best of a grand's job.

A more natural and open sound can be achieved with upright piano by removing the casework panels to expose the instrument's strings and soundboard. Spaced stereo techniques tend to be prevalent here, and the three different pairs seen here were compared for the 'UprightPanelsOff50cm' set of audio examples.

frequencies, it's also plain that the sound around the piano changes in a lot of other less predictable ways, especially at the low end. Mic one is particularly boomy, for example. This is because each mic sees different phase cancellations between the reflected sound from the lid and floor, and each is also affected by the room's resonance modes in a different way. Some theoretical knowledge will help you get into the right ballpark as regards mic position, but there's simply no substitute for a bit of trial and error if you're going to get your mics into a truly star-spangled position.

What About The Lid?

Now I've been assuming for the moment that the piano lid is fully open, but what if it's not? Well, closing the lid completely is rarely a good idea, because the closed box attenuates high frequencies more than low frequencies, so you get quite a muffled sound that's rarely of much practical use. If you're lucky, some high frequencies from the strings might still escape from the narrow aperture behind the music stand, where they could be picked up by a mic on the keyboard side of the instrument but, frankly, it's pretty meagre pickings. You can hear the effect of closing the piano's lid in my 'HorizDispLidClosed' set of audio examples, which were recorded with mics in exactly the same positions as for the 'HorizDispLidOpen' set above.

For chamber music, the balance of the instruments is sometimes improved by using the shorter support stick to open the lid only part-way. As you can hear from the 'HorizDispLidHalfOpen' audio examples, this more subtly mellows the tonality as a whole, but the relations between the tonal characters of each mic position remain broadly similar, and the frontal positions still capture the most high frequencies.

Although removing the lid completely can be worthwhile when close-miking, this is unlikely to give you a suitable sound with ambient mic techniques. Without the lid, the high-frequencies direct themselves primarily upwards, rather than being focused outwards towards any of the mics, which dulls the sound out at the front of the instrument. Furthermore, the removal of the lid also changes the resonant qualities of the instrument and often results in a reduction in the sound's low-end weight. Overall, the timbre becomes more uniform around the piano, as you can hear in the 'HorizDispLidOff' audio examples (recorded with the same mic setup).

Vertical Dispersion

Clearly, the piano doesn't just radiate its sound in two dimensions (who'd want a flat sound anyway?), so the height at which you set your mics also needs consideration. As in the horizontal plane, the sonic changes resulting from mic movements are complex, but there are some principles which can help guide you. Again, the high frequencies from the strings like to travel in straight lines, so putting your mic high enough for it to 'see' the strings over the edge of the piano's case will help get you the brightest sound. However, if you're miking at the front of the piano and go too high, then the open lid will start to shadow the high frequencies as well — you'll find you get the clearest high-frequency reproduction somewhere underneath the line of the open lid. The other thing to bear in mind is that the reflections from the piano's soundboard to the floor will become more prominent in the recording if your mic is lower to the ground, giving what I'd characterise as a more strident timbre.

My next set of audio examples (filenames beginning 'VertDisp') gives some idea of how these changes in miking height affect the recorded sound. I set up six identical omni mics in front of the piano (in the same place as mic two from the 'HorizDisp' files), all of them 1.5m from the centre of the instrument, but at different heights above the floor: to be precise, at 280cm, 235cm, 195cm, 155cm, 115cm and 85cm high for mics one to six respectively. Mic one was just below the line of the piano lid, and both mics one and two deliver a bright, clear sound, whereas mics three and four begin to sound a bit mellower. Mics five and six were below the point at which they could 'see' the upper strings and demonstrate a greater contribution from the soundboard.

Again, the general principles I've mentioned are only one set of factors involved — there's a considerable low-end tonality difference between mics one and two, for instance — but the other changes in the sound are a lot less easy to predict. That said, I've found that vertical repositioning doesn't seem to have quite as drastic an effect as moving the mics around the piano, so I'd recommend first finding a rough position for your mics in the horizontal plane before faffing about too much with their heights.

Getting An Even Sound

One final useful little trick to keep in mind when setting up a close-miked sound is to get the pianist to play a simple full-range scale, as this reveals level unevenness between different registers much more quickly than normal playing. If you find a problem with this test when using a spaced pair, then you can try changing the distance between the mics or raising them higher over the strings. With crossed coincident pairs, the mutual angle can be used to balance the mid-range levels with those of the outer registers — increasing the mutual angle will increase the relative level of the outer registers.

Direct Versus Ambient Sound

The other major decision you need to make is how far away from the instrument you place the mics, the primary issue being that you get a more reverberant sound as you move the mics further away. To illustrate this, I set up my six identical omni mics along an imaginary line reaching from the centre of the piano through the first mic in the 'VertDisp' setup, as shown in the photo. The mics were distanced from the centre of

When creating an ambient recording of a piano, the distance of the microphone dictates to a great extent the ratio between the direct and reverberant sound levels captured. You can hear this in action in the 'Distance' audio examples, where six identical omni mics were set up at different distances from the centre of the piano, as shown in the picture.

the piano by 325cm, 285cm, 250cm, 220cm, 185cm and 140cm respectively, with mic one furthest away and mic six closest in. The recordings from these mics can be heard in the 'Distance' set of audio files.

If you're recording a live classical concert, remember that the ambience levels you get while setting up before the gig may not hold for the final performance if the auditorium is empty. When the hall is later packed with the great and good of the parish, their besuited persons will absorb more room ambience, and may leave your recording sounding too dry.

Once more, it's apparent that the change in miking distance affects not just the amount of room ambience, but also the piano's tone, so there's little use in sweating over fine position tweaks to get the tonality perfect if the level of ambience you're getting isn't yet suitable. It makes more sense to get the ratio of direct to ambient sound pretty much right before finessing the recorded piano tone with small changes in mic position.

How much difference do small changes of mic position really make in practice? To answer this question, six mics were set up very close to each other and their outputs recorded to create the 'TightPattern' audio files — judge for yourself!

How small does a change in mic position have to be to make a difference? Well the simple answer is that even minutely different mic placements sound a bit different, but the real issue is whether movements of a few inches make a big enough difference that they're worth sweating over when you're short of session time. This is obviously a very subjective thing, so I created the next set of audio examples to help you decide for yourself. I left mic four set up from the 'Distance' file recordings, and then set the other mics around it within a few inches of each other (as shown in the photo) to create the 'TightPattern' examples. (Of course, I might just have copied the same file six times to mess with everyone's minds...)

Spaced Stereo Techniques

So far, I've deliberately simplified matters by recording my audio examples with just a single mic in each test position. However, mono piano recordings are pretty thin on the ground these days, so let's look at what kinds of stereo techniques you might try.

One common tactic is to use a spaced ▶

stereo pair. As we've already heard from the 'HorizDisp' audio examples, the differing tonalities of two differently positioned mics will give a kind of stereo effect straight away, but the stereo imaging mostly relies on time-of-arrival differences between them. One of the significant advantages of this approach is that you can use omni mics, with the low-end and off-axis benefits these afford. However, there are also some disadvantages with spaced-pair techniques. The first is that the stereo imaging tends to be rather vague, although a lot of classical engineers and listeners find this sound more musically satisfying, so this could, conversely, be seen as a benefit. What is more clearly problematic in certain situations is that if you don't pan the mics hard left/right you will get phase cancellation between the two signals, which can change the tonality of the sound dramatically. Even if you're not planning on changing the pan settings at all yourself, it still pays to audition any spaced mic pair in mono to check that the sound doesn't completely collapse, as some broadcasters still transmit in mono.

This picture shows the three different spaced-pair stereo rigs lined up together for the 'SpacedPair' audio comparison files: wide omni and cardioid pairs on the outside and a narrower omni pair on the inside. A further central mic was added to demonstrate one way of dealing with the 'hole in the middle' problem when working with spaced-stereo methods.

Because it's tricky to adjust the stereo width of a spaced-pair stereo recording without its tonality suffering, it's important that you try to get the image width you need while recording, by adjusting the distance between the two mics (the further they are apart, the wider the image). A word of caution here, though, as putting the mics too far apart can cause the sound to bunch up towards the edges of the stereo image, producing an effect often called 'a hole in the middle'. I think you'll find that any spacing above about 1m is liable to start running into difficulties, and at the other extreme, Richard King has sometimes placed his mics as close as 45cm apart for piano recording. One way around the hole-in-the-middle problem is to set up a third mic between the main pair, and use this if necessary to fill out the centre of the stereo picture. This can be a good safety net, but a side-effect is that the left and right mics will both cause phase cancellation with the central mic, so it may take a bit longer to get the recorded tonality you're after in the first instance.

There is nothing to stop you using spaced-pair techniques with directional mics as well, although the mics will need to be placed further away from the piano to achieve the same degree of room ambience. Depending on how you angle the mics, you may also find that a hole appears in the middle of the stereo image earlier than it would with omnis, so you should keep a keen ear out for this. And talking of mic angles, even if you use omnis you might still want to experiment with angling them towards the higher strings if you're after the brightest sound, because the high-frequency response of your mics will almost certainly be best on-axis, especially if you're using large-diaphragm models.

The next set of audio examples shows how some of these mic-placement variables affect the sound. I set up an array of seven spaced mics roughly centred on the position of my favourite mic in the 'TightPattern' setup. All the mics were pointing straight ahead, but angled down towards the centre of the piano, and were recorded to the following files:

- **SpacedPair1:** A pair of Rode NT55 omni mics spaced at 40cm.
- **SpacedPair2:** A pair of Rode NT55 omni mics spaced at 1m.
- **SpacedPair3:** A pair of Rode NT55 cardioid mics spaced at 1m.
- **CentreMic:** A single Shure KSM141 omni mic placed centrally between the other mic pairs.

A quick note about stereo polarity here: for all the ambient techniques in this article I've stuck with the convention of having the higher strings of the piano to the left of my stereo image and the lower strings on the right, which is fairly common practice in the classical domain where you're usually trying to recreate the audience's perspective. However, in pop productions engineers usually prefer a player's perspective

Where Should I Set Up The Piano?

If you're using ambient recording techniques for classical-style recordings, it's vital that you find the best possible acoustic for your recording sessions, as it will be all over the final recording. Whatever venue you're in, the question of where to set up in not easy to answer. One thing to factor in is that having the piano right back against a wall or in a corner is likely to boost the low frequencies, because of the way reflected sound from the walls interacts with sound heading out into the room — rarely a desirable effect for classical recordings.

A hard wall within a few metres of the piano can help to brighten the sound overall, by reflecting some high frequencies directly back to the recording mics. If there are no walls close by, the sound of the piano has to go a long way to be reflected back, and as high frequencies don't travel through the air as efficiently as low frequencies, the reflected sound will be duller. However, any strong reflection from a nearby surface may cause some phase-cancellation artifacts at the microphones, which may make it trickier to find decent mic positions.

To give an idea of the scale of these effects in practice, I've recorded the same grand piano with the same omni microphone in four different locations within a concert hall to create the following audio example files:

- **LocationCentreOfHall: For this recording, the piano was in a position about two-fifths of the way down the rectangular hall, and was firing down the long dimension towards the microphone and the remaining three-fifths of the hall. This was also the piano position where the majority of the other audio examples for this article were recorded.**
- **LocationAgainstWall: The piano was moved to the end of the hall, firing down the long dimension towards the mic.**
- **LocationInCorner: The piano was moved into the corner of the hall, firing out towards the centre of the hall and the mic.**
- **LocationFiringAtWall: The piano was four-fifths of the way down the hall, firing at the wall, with the mic set up between the piano and the wall.**

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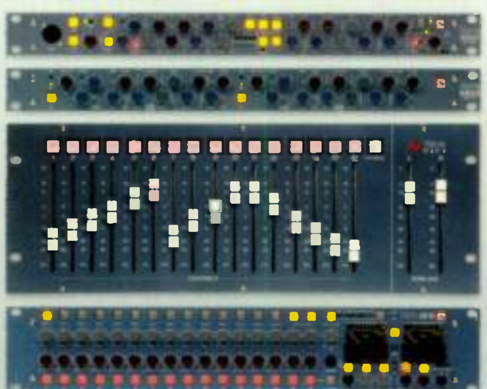

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Here you can see a number of commonly suggested close-miking setups where the mics remain outside the piano's casing. You can listen to how these sound by checking out the 'CloseOutside' audio examples.

(high and low strings the opposite way round), so I've made the audio examples that way for the close-miking discussion later on. If you'd rather hear things the other way round, you'll just have to go and listen to the files in the mirror...

Other Stereo Mic Setups

If you're concerned, as a lot of broadcasters are, about mono compatibility, then spaced-pair stereo techniques don't really cut the mustard, no matter how good they might sound in glorious stereo. In such cases, coincident techniques are the order of the day, all of which place the microphone diaphragms as close together as possible so that phase-cancellation artifacts are negligible when the signals are summed to mono. Because of the mono-compatibility of these techniques, you're also free to narrow the stereo image of the recording at a later date, simply using the pan controls on the two mic channels.

The main family of coincident setups are the crossed pairs, which derive the stereo image from level differences between the mics. The width of the stereo image is proportional to the angle between the mics, usually called the 'mutual angle' — the larger the angle between the mics, the wider your stereo picture will be, although mutual angles beyond about 130 degrees may begin to open up a hole in the middle of the stereo field. Crossed-pair techniques typically give a much clearer and more precise stereo image, but at the expense of what detractors sometimes describe as a rather 'clinical' sound. The fact that such approaches require directional mics, with their potential for off-axis and low-frequency deficiencies, is another reason why some engineers steer clear of them.

There is another coincident option, though, which does allow you to use omni mics: the Mid/Side (M/S) technique. This also gives you the ability to control the stereo width of your recording without moving the mic setup at all, which is great for situations where the best mic position isn't the most accessible. If you're interested in investigating M/S further, or just need a bit more information on stereo mic techniques in general, check out the links to Hugh Robjohns' articles in the 'Help! I'm Buried In Jargon!' box.

Something of a halfway house between the spaced and coincident methods is another family of near-coincident techniques. These use directional mics with fairly small spaces between them in order to capture both level- and time-difference information about the stereo field. On the one hand, you could say that this combines the more precise imagining of the coincident techniques with the more musically involving sound of the spaced techniques. Alternatively, you could also say that this combines restricted mic choice with the potential for phase-cancellation problems! To help you decide which side of the fence you're on, here are some more audio examples. I left one pair of spaced omnis set up from my 'SpacedPair' recordings, and set up coincident and near-coincident pairs between them. All the mics were angled downward towards the centre of the piano, and you can hear how they sound by listening to the following audio files:

- **StereoPair:** Two Rode NT55 cardioid mics in a coincident crossed pair at a mutual angle of around 110 degrees.
- **StereoPair2:** Two Rode NT55 cardioid mics in a near coincident pair at a mutual angle of around 110 degrees and with the capsules 17cm apart (the 'ORTF' standard developed and widely used in French broadcast circles).
- **StereoPair3:** The pair of Rode NT55 omni mics spaced at 1m, as before.

One final thing to mention before we move on to closer miking methods is that some engineers combine the techniques we've been talking about, in order to overcome potential problems with specific approaches. One example of this would be setting up a single low-pass-filtered omni mic alongside a coincident stereo pair for better low-end response. Another common tactic is mixing in a little of the signal from a widely spaced stereo pair with a main closer coincident pair — the omnis give a more involving ambience and good bass

Help! I'm Buried In Jargon

Talking about mic techniques can get pretty technical. If you're feeling a bit daunted by all the jargon, you'll be glad to know that help is at hand from the *SOS* web site. Here are a few links that I'd particularly recommend:

• Paul White's comprehensive introduction to different microphones and their usage can be found in *SOS* September 2006.

W www.soundonsound.com/sos/sep06/articles/microphones.htm

• Detailed explanations of all the stereo mic techniques I mentioned (and many more besides!) are given in Hugh Robjohns's encyclopaedic two-part series in *SOS* February and March 1997.

W www.soundonsound.com/sos/1997_articles/mar97/stereomictechs2.html

W www.soundonsound.com/sos/1997_articles/feb97/stereomiking.html

• If any bits of techno-speak still sneak through the net, then try the *SOS* web site's huge on-line glossary as well.

W www.soundonsound.com/information/Glossary.php response, while the coincident pair fills the hole in the middle of the widely spaced omni image and gives clearer stereo imaging. Once you get a feel for the principles we've been discussing, these combined techniques present no greater fundamental problem than the simple techniques, beyond the practical considerations of setting up, positioning, and phase-checking extra mics during the session.

Moving Closer In

If you're not recording classical music, then you'd be forgiven for stifling the odd yawn so far. Why bother with all this stuff about ambient recording techniques when they are rarely appropriate for more modern commercial production styles? The answer to this question is that a lot of the same principles also apply when you're close-miking, so it's useful to have an understanding of them even when you're planning to move your mics in much closer — which is what we're going to do now.

Most of the information I've found on close-miking grand pianos deals with positioning mics inside them, somewhere over the strings. However, before we get carried away with that, it's worth considering positions just outside the case, in the curve at the front of the instrument. This is an area Hugh Robjohns recommended in his piano-recording article back in *SOS* May 1999, and there are two different techniques like this mentioned in Huber & Runstein's *Modern Recording Techniques*.

Once you're this close to the instrument, the positions of the different strings inside the casing begin to have a greater impact on the sound as you move mics around. The upper strings occupy a comparatively small space behind the right-hand side of the music stand, while the mid-range and lower strings extend right down into the foot of the case, crossing over as they do so. Positioning mics closer to one set of strings or the other inevitably affects the balance of the sound, as does angling directional mics in this way.

Finding a good mic position still isn't quite as simple as that, given the reflections from the instrument's lid, so I've recorded some audio examples to give a taste of the sonic range on offer here. I set up four omni mics at different points along the curve of the piano, and also set up a coincident crossed pair in the middle. You can see how these microphones were placed relative to each other from the photos (opposite), and you can hear the resulting recordings in the following audio files:

- **CloseOutsideMic1-4:** The individual omni mics were all 30cm from the instrument and 30cm above the lip of the case (to minimise shadowing of the high strings). Mic one was closest to the foot of the piano and mic four was closest to the keyboard.
- **CloseOutsidePair1:** This stereo file combines the inner pair of omni mics to create a fairly tightly spaced stereo pair.
- **CloseOutsidePair2:** This stereo file combines the outer pair of omni mics for a wider image.
- **CloseOutsidePair3:** A crossed pair of cardioids, set up with the two capsules pointing to the high and low strings respectively.

Inside The Piano

Broadly speaking, techniques for close-miking inside the piano fall into two main categories: spaced-pair techniques and coincident techniques. Looking at the former to start with, there seem to be two main schools of thought when it comes to deciding where to put the two mics. The first is to tuck them somewhere fairly close ▶

A variety of spaced stereo techniques can be seen in action here, as used for the 'InsideSpaced' sets of audio examples. The pair behind the music stand (in conjunction with the extra crossed pair up by the edge of the lid) was used to recreate techniques described by legendary audio engineer Al Schmitt, while the remaining four mics are set up in line with the preferences of two other experienced professional engineers, Brian Tankersley and Cookie Marenco.

behind the music stand, covering the two halves of the instrument's register before the low- and mid-range strings start seriously overlapping. Al Schmitt, for example, talks in *Behind The Glass* of setting up his pair of Neumann M149 omni mics in this way "usually a couple of feet off the high end and a couple of feet off the low end, kind of at 45 degrees to each other". He also adds, in another interview, that he tries to aim the mics at the hammers to get sufficient attack in the sound.

The second basic approach is to place one mic (typically an omni) over the middle of the group of high strings behind the music stand, and a second mic closer to the foot of the piano, catching the low- and mid-range strings roughly where they cross over. Brian Tankersley referred to this technique back in *SOS* October 2002, and it also crops up in Hugh Robjohns' article and Huber & Runstein's book. I came across another interesting technique courtesy of an engineer called Cookie Marenco, who uses a near-coincident rig in the middle of the piano, taking advantage of the directional characteristics of two cardioid mics to pick out the low and high strings respectively.

To compare these different sounds, I placed three pairs of Rode NT55s 30cm above the piano strings (as shown in the photo) to create the following audio files:

- **InsideSpaced30cmPair1:** This recording is from two omni mics behind the music stand, positioned roughly as described by Al Schmitt.
- **InsideSpaced30cmPair2:** Here, I had one omni mic over the high strings, and another omni closer to the foot of the piano, where the low- and mid-range strings cross over.
- **InsideSpaced30cmPair3:** Two cardioid mics were positioned in a near-coincident pair over the middle of the piano, with the two capsules pointing downwards and angled towards the upper and lower strings respectively.

Irrespective of which spaced technique you might decide to use, it's worth checking how your mic setup's tonality comes across in mono. If phase cancellation is destroying your carefully adjusted sound, try shifting the mics a couple of inches in relation to each other — this will usually change the effects of the phase cancellation quite a lot in mono, while making very little difference to the sound in stereo.

You may already have spotted that there is an extra crossed pair of mics in the photo up by the crook of the lid. I put those there to check out another of Al Schmitt's recommendations, namely adding some extra stereo ambience from a crossed pair in this position to supplement the sound from the omni close mics — I used a crossed pair of Shure KSM141 cardioids for this, recording them alongside the NT55s. You can hear them on their own in the 'SchmittAmbience' audio file, and I've mixed them in with the close mics at a fairly subtle level for 'SchmittMix'. Schmitt isn't the only engineer using this kind of technique (Tony Visconti, for example, mentions using additional room mics in *Behind The Glass*) so it's worth having a go, especially as you can probably get away with using less high-quality mics for the ambient pair than for the main pair.

An additional mic underneath the piano can help bolster the low end of the piano sound when close-miking — for my 'UnderPiano' audio example I used a C414B-XLS in omni mode, positioned as shown here.

Different Heights For Close Mics

There is some disagreement amongst different authorities as to how high the mics should be within the piano, with some engineers (often in pop, blues, or rock styles) preferring a closer position, where each string is more distinct, and others (such as Al Schmitt) favouring a more distant placement where the harmonics of the different strings have more chance to blend. The more distant placement has the practical advantage that it keeps the levels of the notes in different registers sounding more even — a mic placed very close in will pick up the strings next to it much more strongly than it will the strings further away.

You can let your ears decide which sound you prefer by listening to the 'InsideSpaced15cm' audio files, which were created from the same mic positions as before, but this time with the mics only 15cm above the strings. While I was repositioning the mics, I also set up another mic (an AKG C414B-XLS omni) underneath the piano and recorded it alongside the 'InsideSpaced15cm' mics to create the 'UnderPiano' file. This mic placement was something which Paul White mentioned might be worth experimenting with in his piano-recording article back in *SOS* October 1994, the idea being that it picks up a more mellow sound from the soundboard.

A lot of *SOS* readers don't have access to omni mics, though, so what kinds of results might you be able to expect using the two spaced-omni techniques with directional mics such as cardioids instead? The biggest risk is that the directionality of the mics will 'spotlight' certain regions of the strings at the expense of others, making certain ranges of notes over-prominent, and also potentially leading to something of the hole-in-the-middle problem we've already encountered in relation to ambient miking. You'll also get a drier sound than with omnis, although you might prefer this on a subjective level. To help give an idea of the kind of change involved, listen to the 'InsideOmnisToCardioids30cm' files, where I used the same mic positions as for the first

For studio productions some engineers, notably Elton John's producer Gus Dudgeon, like to place their microphones further from the piano's strings by removing the instrument's lid. To demonstrate the effects of this approach, the 'InsideSpaced' configuration of microphones was repositioned at a greater height in this way, as you can see in the picture, and recorded to generate the 'InsideSpaced60cmLidOff' audio examples.

two 'InsideSpaced30cm' recordings, except that I switched the mics' omni capsules for cardioid ones.

When we interviewed Gus Dudgeon in July 2001, the legendary producer of Elton John remarked: "I never close the lid on a piano. It's the worst thing you can possibly do. Taking the lid off is even better, if you can get the lid physically off. The lid is only there to bounce the sound out into the hall when you're playing live with an orchestra." This is a view shared by a number of recording engineers, so I felt it would be worth investigating how this affects the sound by removing the lid and re-recording exactly the same mic pairs I used for the 'InsideSpaced30cm' recordings. These recordings can be heard in the 'InsideSpaced30cmLidOff' files.

Clearly, removing the lid gives you more scope to raise the mics, something which Gus Dudgeon went to great lengths to take advantage of, even though that meant building an elaborate baffle to reduce spill from other instruments during his recording sessions. To try to give a sense of what kind of difference a bit of extra height makes, I moved all my mic pairs up as far as I could (so that they were all about 60cm above the strings), and recorded the 'InsideSpaced60cmLidOff' files.

Coincident Pairs Inside The Piano

For similar reasons that some classical engineers favour coincident stereo techniques over spaced pairs, there are also devotees of coincident methods inside the piano. Probably the most commonly mentioned position is somewhere behind the music stand. Both Ed Cherney and Jay Newland advocate this approach and this option also appears in Huber & Runstein's book. Although there appears to be some consensus that a sensible height for these mics is about 20-30cm from the strings, exactly how far behind the stand the stereo pair should be seems more open to debate. Positions closer to the hammers give a brighter and more percussive sound, whereas the tone gets progressively warmer the further back you go. To hear this for yourself, check out the 'FrontToBack' audio files, where I set up six identical omni mics along the centre of the piano roughly 30cm from the strings. Mic six was right behind the music stand, and the other mics were progressively further towards the foot of the piano.

Huber & Runstein describe one other coincident position that is also worth considering. This is where the mics are placed just inside the piano (over the soundholes) at a height roughly midway between the case and the lid. The two mic capsules are then pointed diagonally down towards the low and high strings respectively to create the stereo picture. Because of the positioning over the sound holes, you get more of a contribution from the soundboard than with the other technique, which gives quite a different timbre (this is a tone for which Tony Visconti expresses a preference in *Behind The Glass*, although he adopts a spaced-pair approach).

The following files give some idea of how the sounds of these techniques compare in practice (you can see how all the mics were rigged from the photo, below left):

- **InsideCoincidentPair1:** A crossed pair of cardioids 30cm directly above the hammers, pointing downwards.
- **InsideCoincidentPair2:** A crossed pair of cardioids 30cm above the strings in the centre of the piano, pointing downwards.
- **InsideCoincidentPair3:** A crossed pair of cardioids set up just inside the front of the piano, as described in Huber & Runstein's book.

In the case of the first of these pairs of mics, the position gives quite a bright sound, so it's not uncommon for engineers to mix in the signal of a further mic to reinforced the low end. An approach that Ed Cherney has used is to mike up one of the low soundholes really close with a directional mic, such that the already bassier sound at that point of the piano is exaggerated by the proximity effect of the mic. Jay Newland also likes to put an additional mic towards the foot of the piano, resting on a piece of foam, for a similar purpose. So, while recording the above audio examples, I also had an AKG C414B-XLS cardioid mic and a Shure KSM141 omni running to test out Cherney's and Newland's techniques respectively. You can listen to these mics for yourself by downloading the 'ChernyBassMic' and 'NewlandBassMic' files.

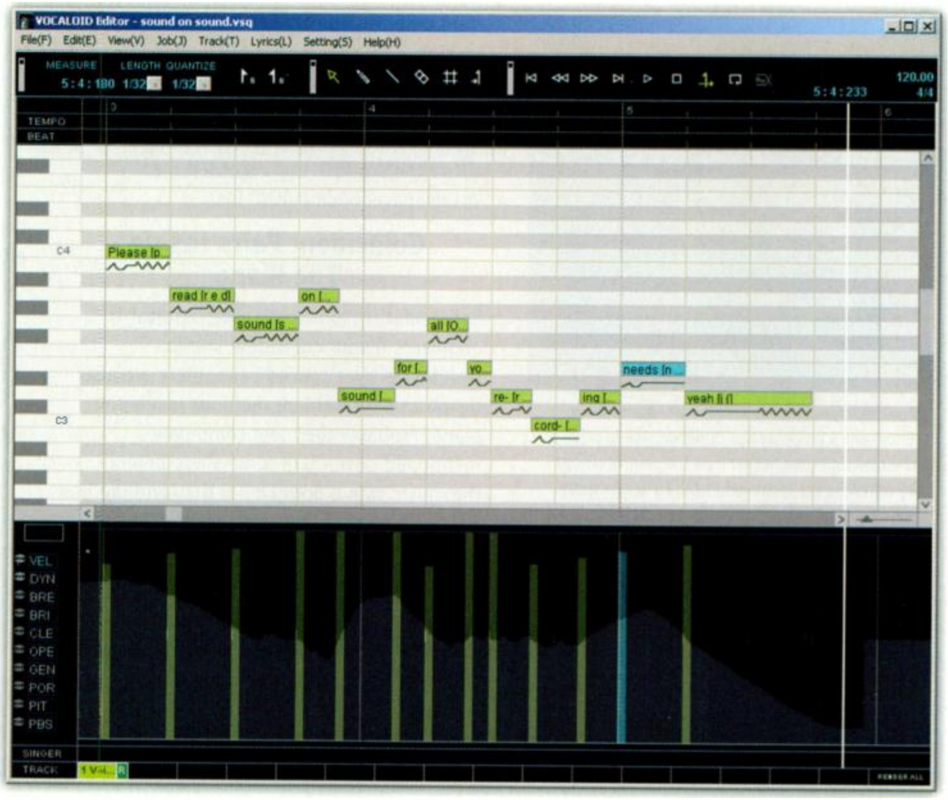
Narrowing Down The Choices

It would be fair to say that the number of options available to the recording engineer when recording piano can be bewildering, and most home studio owners simply don't get a chance to hear enough different mic techniques to decide what does and doesn't get the sound they hear in their head.

If you've been able to work your way through the audio files as you've been reading this, though, you should already have a much clearer idea of which miking approaches are likely to work best for you. That means you'll already be armed with a couple of good starting points for miking up your next piano session, and can spend the session time refining them into something that sounds stunning, rather than wasting hours eliminating masses of less suitable alternative rigs. SOS

Here you can see the three coincident stereo close-miking techniques and you can compare them using the 'InsideCoincident' audio files. You can also see two extra mics at the foot of the instrument (a C414B-XLS cardioid on a stand and a Shure KSM141 omni resting on a folded towel), the placement of which follows suggestions from high-profile engineers Ed Cherney and Jay Newland.

Yamaha's Vocaloid technology has now been upgraded to version 2 and Sweet Ann, from PowerFX, is the first virtual singer based on the new release. So just how much further forward have Yamaha moved their intriguing vocal synthesis technology?



The stand-alone editor has a less cluttered look in Vocaloid 2 and a much improved control track area.

PowerFX Vocaloid 2 Sweet Ann

Singing Synthesis Software For PC

John Walden

Yamaha's Vocaloid technology caused quite a stir when Zero-G released the first virtual singers, Lola and Leon. These were reviewed in the March 2004 issue of *SOS* (www.soundonsound.com/sos/mar04/articles/vocaloidlandl.htm), and Zero-G followed the initial releases with Miriam, based on the voice of Miriam Stockley and reviewed in the December 2004 issue (www.soundonsound.com/sos/dec04/articles/miriam.htm). For many songwriters and producers, the possibility of creating complete vocal performances by simply typing in lyrics to go with a MIDI-based melody was — and still is — an appealing prospect.

In its first incarnations, Vocaloid was undoubtedly a remarkable and innovative product and, with experience and patience, was capable of producing results that could be frighteningly realistic. The catch, however, was gaining the experience and having the patience. Although creating backing vocal phrases and harmonies was a feasible proposition, attempting to craft a realistic lead vocal (that is, something not intended to be in the 'special effect' category) represented a significant undertaking. Detailed editing of the phonetic sounds was necessary to get Vocaloid's pronunciation right, and a lot of work on the various expression controls was required to give the vocal some 'life' and dynamics — factors that come built in with most warm-bodied singers!

Of course, creating a virtual vocalist is an ambitious project and, to their credit, Yamaha have persisted with the challenge. Vocaloid 2 is now upon us and PowerFX are the first company to license the technology and release a product based upon it. They describe Sweet Ann as a 'space lounge robo-vocalist sensation', and also have Big Al — a male singer — in the pipeline. Meanwhile, Zero-G have announced that their classical vocalist

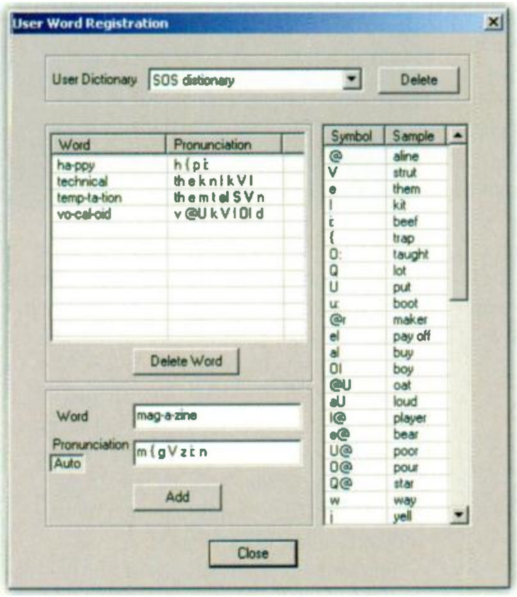
SOUND ON SOUND

PowerFX
Vocaloid Sweet Ann **$169**

pros
- Much easier to use than Vocaloid 1.
- Real-time VST Instrument plug-in provides a very intuitive way of working.

cons
- A number of workflow issues could be addressed to improve matters further.
- Great for backing vocals but convincing lead vocals still require a lot of editing work.
- Sweet Ann's voice will only suit some musical styles.

summary
Vocaloid 2 brings considerable improvements in ease of use, and Sweet Ann can certainly serve a useful role in some vocal contexts. Convincing lead vocals are still a challenge, but Yamaha deserve credit for pushing forward with their remarkable Vocaloid technology.



The User Word Registration table allows customisation of word pronunciation if required.

Prima is slated for release before the end of 2007. So how much closer have Yamaha moved us to having a singer in a box, and just what does a 'space lounge robo-vocalist sensation' sound like?

Repeat Performance

As before, Vocaloid is provided as a stand-alone editor application with Rewire support and as a VST Instrument plug-in. New with version 2 is a 'VSTi realtime' plug-in, of which more a little later. Although there are some significant changes in the new version of Vocaloid, the basic principles of its operation remain the same, so if you are new to the product, the previous *SOS* reviews mentioned above are well worth a quick read.

Sweet Ann, like all Vocaloid-based virtual singers, consists of two elements: the Yamaha synthesis engine and a singer database. The former provides a piano-roll-style editor into which the user can enter notes to create a melody, before assigning lyrics to each of these notes, along with various controls to add expression. The singer database consists of a sample library where the singer has been sampled singing all possible phonetic sounds and transitions between different phonemes. Once the lyrics are entered, Vocaloid extracts the necessary set of phonetic samples, links them together at the required pitch, adds the expression and — as if by magic — sings the vocal part required. The editing required is less of an issue when creating short phrases suitable for backing vocals and, usefully, Vocaloid allows multiple tracks to be created for harmony production.

Take 2

While there are all sorts of detailed changes in Vocaloid 2, the most significant new features include a new synthesis engine, some improvements to the user interface, and the addition of the real-time instrument plug-in version. One outcome of the changes to the synthesis engine is that it does not seem possible to use sample databases created for the version 1 engine with the v2 engine. However, I was able to open files created in Vocaloid 1, and the new editor did a decent job of translating them into Vocaloid 2 format.

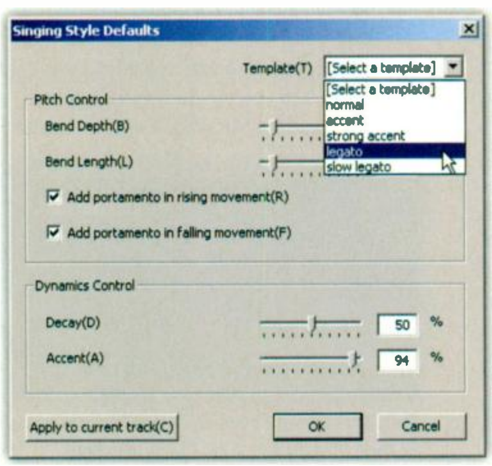
The piano-roll editor is essentially the same as before, but the toolbar has been streamlined (with some options moved to the main menus). Perhaps the more significant change, however, is in the control track. A list of control parameters is displayed down the left-hand edge, with the currently selected parameter highlighted in blue. The control track is now semi-transparent, with the previously edited parameter remaining visible when a new parameter is selected. This makes adjusting multiple parameters much easier.

The set of control parameters has also changed. The rather mysterious 'resonance' controls from version 1 have gone and the relationship between Velocity and Dynamics parameters seems better defined, with Velocity influencing the length of consonants at the beginning of notes (useful for adjusting pronunciation), while Dynamics alters loudness, for adding volume changes. The majority of the other parameters — Breathiness, Brightness, Clearness, Opening and Gender — change the character of the voice, although they do need to be used carefully or obvious audio artifacts can be introduced.

One of the comments I made in reviewing Lola, Leon and Miriam was that it would be nice if Vocaloid included some default 'styles' for expression that could be automatically applied to a vocal line to speed up the initial stages of vocal creation. Yamaha made some useful moves in that direction with an updater for Vocaloid 1 and this process has been developed a little further in the new release. The Settings / Singing Style Defaults option offers a selection of templates, and also allows manual setting of a number of pitch and dynamics controls. These settings can then be applied to all notes in the current track, providing a good starting point for further note-by-note editing.

The control track aside, editing of all the details associated with individual notes is now done from within the Note Property dialogue, which is available via a pop-up menu when you right-click on a note. This includes further drop-downs to customise the expression and vibrato settings, and edit either the lyric or its phonetic translation. It is also possible to protect a note once you are happy with its execution, which prevents any subsequent track-based editing from overwriting the note properties. For detailed editing, this dialogue is simple and effective, but it would be a really big help if it included an 'audition' button so the single note could be rendered by the synthesis engine, allowing you to hear the results of any edits without returning to the piano-roll editor. As it stands, you have to close the dialogue and play through the arrangement to hear the changes, which is not great for the workflow as you are fine-tuning a performance.

Yamaha have, undoubtedly, made some excellent improvements to the Vocaloid editing process, but it is a little surprising that the Undo feature still only supports the last action — multiple levels of undo would be considerably more useful. Basic MIDI input into the stand-alone editor to create your initial melody would also have been nice to see.



Real Time

For me, the real highlight of Vocaloid 2 is the real-time VSTi plug-in. This uses the same engine as the stand-alone editor but, as far as the user is concerned, operates in a very different fashion. Once added to a project (for example, via the VST Instruments panel in Cubase), like other VSTis the plug-in is then available as a possible output destination for a MIDI track. Clicking on the Lyrics Edit ▶

Editing for individual notes can all be done from the Note Property dialogue.

System Requirements

Windows XP, 512MB RAM (2GB recommended when using real-time VSTi plug-in), 2GHz Pentium 4, Athlon XP200+ or faster CPU, 2GB hard disk space, DVD-ROM drive.

button opens a further dialogue into which lyrics can be typed or pasted from another application. When the phrase is complete, Vocaloid breaks it into its various phonetic sounds. The user then presses the 'aA' button and the synthesis engine does its work behind the scenes. The phrase can then be triggered either live from a MIDI keyboard or from a pre-recorded MIDI track, with each MIDI note triggering a single syllable.

The user still has control over a number of expressive options. The Settings button opens a dialogue for customising the vibrato, pitch-bend and portamento properties. The faders, which can be controlled in real time either via a mouse or a hardware controller, allow changes in the voice quality to be made, while note velocity (attack of first consonant), pitch-bend, modulation (to control vibrato) and expression (volume) can also provide real-time control.

The Delay and Decay faders provide some useful control over pronunciation. Delay influences the length of the consonant at the beginning of a note, while Decay adjusts the length of the consonant at the end, with the numeric values in milliseconds. I found it useful to adjust these, using different settings for rapid and slower phrases, but they can be bypassed by pressing the Fixed button, which just applies a preset length.

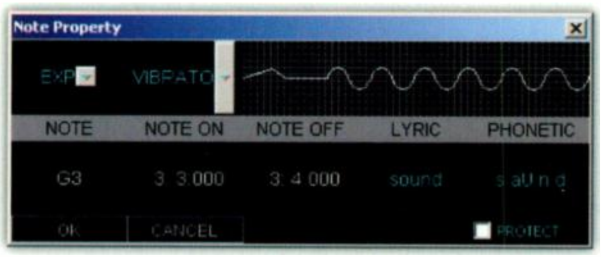
The other interesting options concern the Mono and Poly buttons. In Mono mode, the plug-in creates a single voice. Unlike the stand-alone editor, however, overlapping notes are allowed, which permits a syllable to be stretched over multiple pitches — great for adding movement to a melody. In Poly mode, the plug-in provides up to a four-part harmony, and the obvious application here is for backing vocals. If you press a single note and hold it and then add a second, third and fourth notes, the voices gradually appear singing the same syllable. Only when all notes are released and a new set of notes are triggered does the engine move on to the next syllable. Again, this add further flexibility, although it does take a bit of practice to get used to 'playing' a set of voices in this fashion.

For me, the VSTi real-time interface is a much more intuitive approach to creating a synthetic vocal than the stand-alone editor. If Yamaha continue to develop Vocaloid, I would imagine this is the direction in which they will take it.

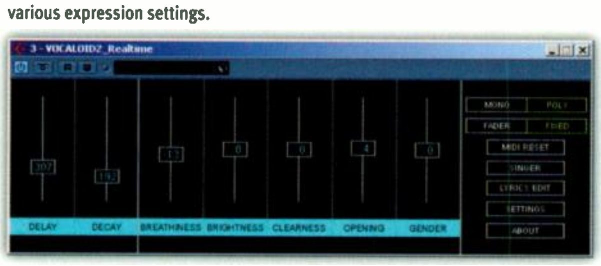
Audition Time

So much for the operational improvements in Vocaloid 2 — but do they result in a more convincing virtual vocal, and how does Sweet Ann sound? In brief, the answers to these two questions are 'sometimes' and 'sweet'. I tested both the stand-alone editor and the real-time plug-in within Cubase, and the changes to the user interface made it much easier to get an initial vocal line beyond that 'this is obviously synthetic' stage. That said, going that extra mile to edit the expression and pronunciation in such a way that the vocal sounds convincingly natural is still a challenge. In fairness to both Yamaha and PowerFX we should not lose sight of what is being attempted here: synthesizing the most expressive of musical instruments is not an easy task. Vocaloid 2 is most certainly a step up from the earlier release, both in terms of ease of use and synthesis quality, but the technology still has some way to go before human singers need worry about being routinely replaced.

That said, how appropriate Vocaloid and Sweet Ann are as tools depends upon the job you have in mind. For gentle 'ooh'- or 'aah'-style vocal lines or harmonies, it is perfectly possible to create something very convincing, and you can do it much quicker with this release of Vocaloid. The same is true for short phrases using real words for vocal hooks or backing vocals; these require a little more work, but it can certainly be done. And as with Vocaloid 1, creating vocals that are intended to sound synthetic (as might be used in some dance styles, for example) is a breeze. That leaves the question of whether a convincing lead vocal can be created and, at this stage, I think that's still right at the edge of Vocaloid's talents, although I'm sure a dedicated few will attempt it.



The Settings dialogue for the real-time VST Instrument plug-in provides access to various expression settings.



The new Vocaloid 2 real-time VST Instrument plug-in running within Cubase 4 — it might look a little bland but it is a lot of fun to use!

The Lyric Edit screen for the real-time VSTi plug-in makes it easy to enter the phrase you wish to have sung.

Test Spec

- Vocaloid version 2.0.2.4.
- Athlon dual-core 4400+ with 4GB RAM, TC Electronic Konnekt 24D and Echo Mia 24 soundcards, running Windows XP Pro Service Pack 2.
- Tested with Steinberg Cubase 4.0.3.

In terms of vocal styles, Sweet Ann is — as her name suggests — quite sweet-sounding and, synthesis quality aside, the voice is perhaps more suited to certain pop styles or the purer side of musical theatre than sexy R&B or raunchy rock. This is perhaps understandable, as I would imagine it would be more difficult to reproduce a variable smoker's rasp via the synthesis process!

Pop Idol?

So where have Vocaloid 2 and Sweet Ann taken us to? The new synthesis engine does seem to have bought some improvements in the quality of the vocal that can be created, but these improvements should probably be seen as incremental steps rather than a revolutionary leap forward. In practical terms, the biggest strides — at least on the basis of Sweet Ann — appear to have been made in the area of the user interface. This now makes it much less hard work to get towards the best that the synthesis engine has to offer, and the VSTi real-time plug-in suggests the beginnings of an approach that is much more musician-friendly.

I can see many musicians (myself included) putting Sweet Ann to use for a range of supporting vocal tasks, and although Vocaloid has not yet reached the point where it could be regarded as a first-class singer in a box, Yamaha and partners such as PowerFX deserve considerable credit for pushing the envelope a little further. I, for one, hope they can continue to resource further developments, because this is remarkable and fascinating technology. SOS

information

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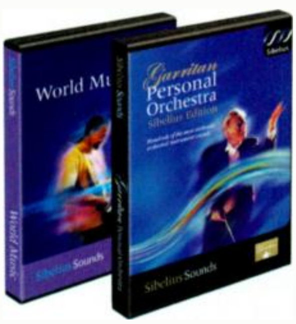

Over the years, Sibelius has come to be regarded as the go-to tool for computer-based scoring. It all started back in the mid-'80s with British twins Ben and Jonathan Finn developing Sibelius 7 for Acorn Computers. By the late '90s, Windows and Mac versions of Sibelius were available, and derivatives for education were on the market and spreading into schools worldwide, helped by the development of foreign language versions. The Sibelius Group was set up in 1999.

The turn of the century saw even more successful Sibelius offshoots and by 2003 the Sibelius Group were considered market leaders, beating their competitors on revenue by 20 percent. A year later, in 2004, Sibelius products were being used in an amazing 50 percent of UK schools.

In 2006, the Avid group acquired Sibelius Software for a staggering $23 million, integrating them as a business unit of Digidesign. Months later, in June 2007, Sibelius 5 was launched.

The new version of the software, for the first time ever, can host VST or Audio Units plug-ins, enabling any compatible virtual instruments to be used as sound sources inside the program. Other improvements over previous versions include a new technology called Soundworld, which intelligently decides how instructions in Sibelius' score should relate to virtual instruments, and the facility to export audio off-line, faster than real time.

This month, thanks to Sibelius, we're giving away a scoring bundle with a total value of £1312. Included is, of course, a copy of Sibelius 5 (which normally costs £595), plus Sibelius World Music, a collection of high-quality samples of indigenous instruments from around the world (worth £119), GPO Sibelius Edition, the acclaimed virtual orchestral software from Garritan (usually £129), and M-Audio's fully weighted Pro Keys 88 stage piano, which normally retails for £469 and has been kindly donated by M-Audio UK. All you'll need is a computer on which to run the software, and you're away!

If you would like to be in with a chance of winning these fabulous prizes, simply fill out the entry form at the bottom of this page and post it to the address on the coupon, or enter electronically via the *Sound On Sound* web site. Please make sure you answer all the questions and complete the tie-breaker. We also require your full address, including postcode, and your daytime telephone number, so we can let you know if you've won. The closing date for entries is **January 31st, 2008.** SOS

the small print

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

What is the surname of the brothers who founded Sibelius?
- a. Twin ☐
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Sibelius tie-breaker

If you were to name a piece of software after a composer, what would it be called and what would it do? Answers in 30 words or fewer, please.

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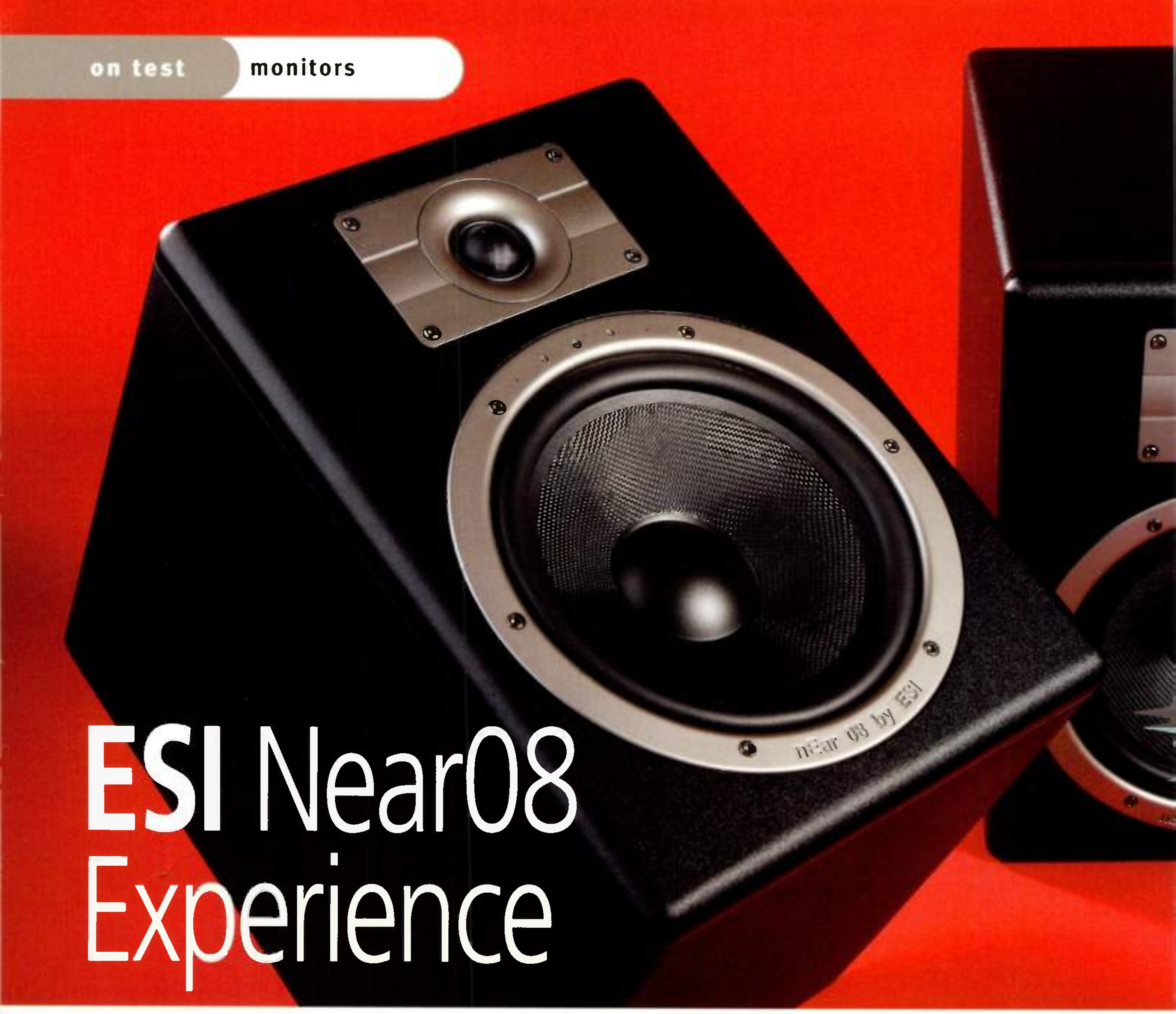


ESI Near08 Experience

Nearfield Active Monitors

Can you expect decent project-studio performance from monitors costing under £300?

Paul White

The ESI Near08 Experience is a two-way, rear-ported, active monitor priced to meet the needs of the home-based project-studio owner. It's distributed in the UK by sample giants Time + Space, and represents their first foray into studio hardware.

Overview

The monitors are built around a Kevlar-coned eight-inch bass/mid driver, partnered with a ferrofluid-cooled, neodymium-powered, one-inch, soft-dome tweeter. They're built into economically constructed hi-fi-style cabinets, made from black vinyl-laminated MDF, and measure 358 x 255 x 322 mm. The speakers weigh 10kg each, making them suitable for stand or shelf mounting, and the drivers are symmetrically positioned, so the right and left speakers are identical. The cabinet edges are slightly rounded, which not only looks better than sharp edges but also helps reduce cabinet-edge diffraction.

All the connections and controls, including both TRS jack and XLR balanced inputs, are fitted to the metal rear panel, which also acts as a heat sink for the amplifiers, and the cabinet tuning port is also at the back. Rear porting has the advantage of diminishing port noise caused by turbulence, but it does mean that the speakers need to be set up at least a few inches from the wall behind. I couldn't see any internal cabinet damping material through the port, and tapping the bass-driver cone produced a definite note somewhere in the kick-drum range, which is usually a sign that speakers will exhibit a degree of low-end overhang, making bass sounds seem more full than they actually are.

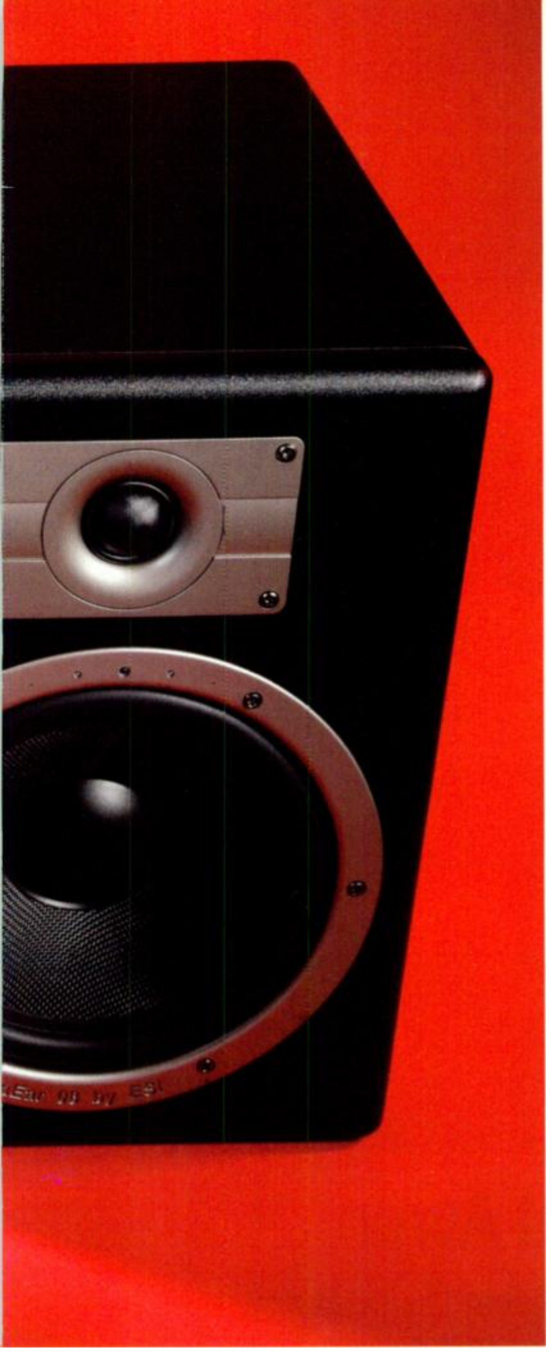
Sound On Sound

ESI Near08 Experience £266

pros
- Inexpensive.
- Punchy, detailed sound.

cons
- May be too aggressive-sounding for some tastes.

summary
These monitors deliver surprisingly good sound quality and at generous SPL levels, given the price. They definitely have a forward character, but many people seem to like that sound.



Photos: Mike Cameron

The two internal amplifiers are rated at 70W each, and the system claims an impressive frequency response of 40Hz to 24kHz, though no maximum SPL is quoted. Both drivers are magnetically shielded by means of field-cancelling magnets, and the amplifier circuitry includes RF filtering, subsonic filtering, output current limiting and over-temperature protection, as well as the usual mains-panel fuse. While the designers claim that the use of Kevlar is a major improvement over the paper and polypropylene drivers used in other products, it is interesting that the designers of many other highly acclaimed speakers still swear that doped paper sounds the best, as it has good self-damping properties — there's really no general agreement within the industry as to the ultimate cone material (in fact, Fostex make their high-end driver cones out of bananas!). Certainly Kevlar, a key component of bullet-proof clothing, is both strong and light in weight, and manufacturers such as KRK have been making extensive use of it for many years.

The manual states that the crossovers have been designed to give the flattest possible frequency response (though no crossover details are provided). Some designers take the position that it is more important to achieve a flat phase response, and that small irregularities in the frequency response are less objectionable than the phase errors created by using filters to flatten peaks or dips in the cabinet and/or driver response. Ultimately, all you can do is use your ears to see what really works.

The Near08 Experience monitors also include a degree of frequency-response adjustment, via a control on the rear panel. A rotary switch allows the high-frequency (HF) level to be adjusted over the range -2 to +1dB in 1dB steps, though I left it set flat for my tests. A low-frequency 'boost' rotary switch (with settings of 50, 60, 85 or 100Hz to match the room and mounting position) is used to adjust the low end, and this may also be useful when teaming the system with a subwoofer, such as ESI's own SW10K. There's no explanation of what this switch actually does, but from listening it sounds to me like a switchable high-pass filter, which can be used to restrict the bass extension in smaller rooms (the term 'boost' may be misleading). Mains power comes in via the usual IEC socket and the adjacent power switch, and there's an AC selector for 110V or 230V AC operation. The blue LED set into the woofer mounting ring indicates when the speakers are powered up.

Testing

After wheeling out all the usual test tracks and mixes, I soon got a pretty good idea of how these speakers behave. Firstly, although no maximum SPL is quoted, they are capable of playing back at very high volumes, while still remaining clean and punchy. Tonally, I found that at the flat setting I chose for my initial test they were noticeably bright and splashy, verging on the aggressive — in a blindfold test I would have sworn I was listening to titanium tweeters rather than soft domes. No doubt those into NS10s and the like will find this perfectly acceptable, but I quickly began to find it fatiguing and reset the HF switch to its -2dB position. This helped — I felt the sound was still on the forward sound of neutral but it was no longer so aggressive. The low end also turned out to be better behaved than I'd expected from the cone tapping test I described earlier, and though it was a little unfocused it didn't dominate proceedings, and seemed well balanced against the mid-range, with a convincing degree of depth. The stereo imaging was also fine (though not exceptional) and the overall sense of detail and clarity was good.

Alternatives

If you're considering these monitors and want to audition alternatives, you could look at models such as the Alesis M1A MkII, the Samson Rubicon, Fostex's PM6A and the Tannoy Reveal Active.

Conclusion

It must be borne in mind that these are far from expensive monitors, and once you've made the necessary HF adjustments and found the best position for them, they actually turn in a very decent performance, with lots of bass extension for use in a typical project studio. They are a little forward-sounding for my personal taste, but I know lots of engineers who think that this is how a good monitor should sound — and it would certainly be no problem doing good mixes on them after first having got used to them by auditioning a range of good-quality commercial recordings. There are only three or four other monitors that perform well in this price range, and the Near08 Experience models deserve to be taken seriously alongside them. SOS

The metal rear panel doubles up as a heatsink, as well as providing high and low frequency controls and a master volume.

information

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Photo Nicola Fourie

Secrets Of The Mix Engineers: Neal Avron

Paul Tingen

The unsung production hero behind Fall Out Boy, described as "emo's first superstars", tells us how he shaped the lead single from 2007's smash hit album, *Infinity On High*.

Neal Avron is predominantly known for his work with hit parade-storming younger rock bands, among them Linkin Park, Fall Out Boy, Weezer, the Wallflowers, Plain White T's, Lifehouse, Yellowcard and Everclear. With a background as a musician, he has a predilection for working with live musicians and particularly with bands. Avron's work with Fall Out Boy is among his most successful. He engineered, produced, and mixed the band's third album, *From Under The Cork Tree* (2005), which went multi-platinum and put them in the arena league. Avron steered the follow-up *Infinity On High* (2007) to similar success. Three of the album's 14 tracks were produced by others, among them famously hip-hop producer Babyface, but Avron produced the rest, including the album's first hit single, 'This Ain't A Scene, It's An Arms Race', the subject of this article.

Infinite Preparation

Work on the album began with six weeks of pre-production. "I am a big believer in pre-production," stresses Avron. "We rehearsed for a couple of weeks in Chicago, and the rest of the time we were at Swing House rehearsal studios in Los Angeles. The pre-production time included rehearsals and some writing, and working out all the sounds and arrangements. I also do very crude recordings of the pre-production sessions, as a reference in the studio."

Avron recorded all the tracks he produced on *Infinity On High* at The Pass studios in Los Angeles, which is based around a vintage Neve 8078 desk. The drums were recorded first, with singer/guitarist Patrick Stump playing and singing along, to add "the vibe and energy that comes from playing with someone else".

Avron began by getting the sounds right in the recording area. "The most essential ingredient to a drum recording is to have great-sounding drums. I use a drum tech, and we had a bunch of great-sounding kits, including Andy Hurley's own set, and we went through every song and decided which kick to use, and which toms, and which snare, and which hi-hat, and so on.

"All sound starts with the instrument and the player. Figuring that out first makes things a lot easer, because there isn't necessarily as much trickery necessary later on in the control room to make things happen. So when I put the mics up and go to the control room, I first and foremost try to get a great representation of what happened in the live room, listening to the close mics and the room mics together. I'll EQ as needed, but if I find that I'm turning knobs too much, I know that I have to start looking in other places: maybe the mics aren't right, or they're not placed correctly, or maybe the instrument is not sounding good."

The drums were recorded "to a 24-track Studer A800, 30ips, no Dolby. The main reason for doing this is the sonic thing. Tape saturation does something to the transients of drums and percussion that to me is very warm and natural-sounding. I'm not doing it with bass or guitars or vocals, because they don't have the same peak information as drums. It's not to say that recording them to tape doesn't sound good, it's just that I find that that the drums benefit the most from being recorded on analogue tape. After we have recorded several takes, and I feel the drummer has done enough to nail a performance, whether a whole performance, or to edit one together from different takes, the drums are dumped into Pro Tools.

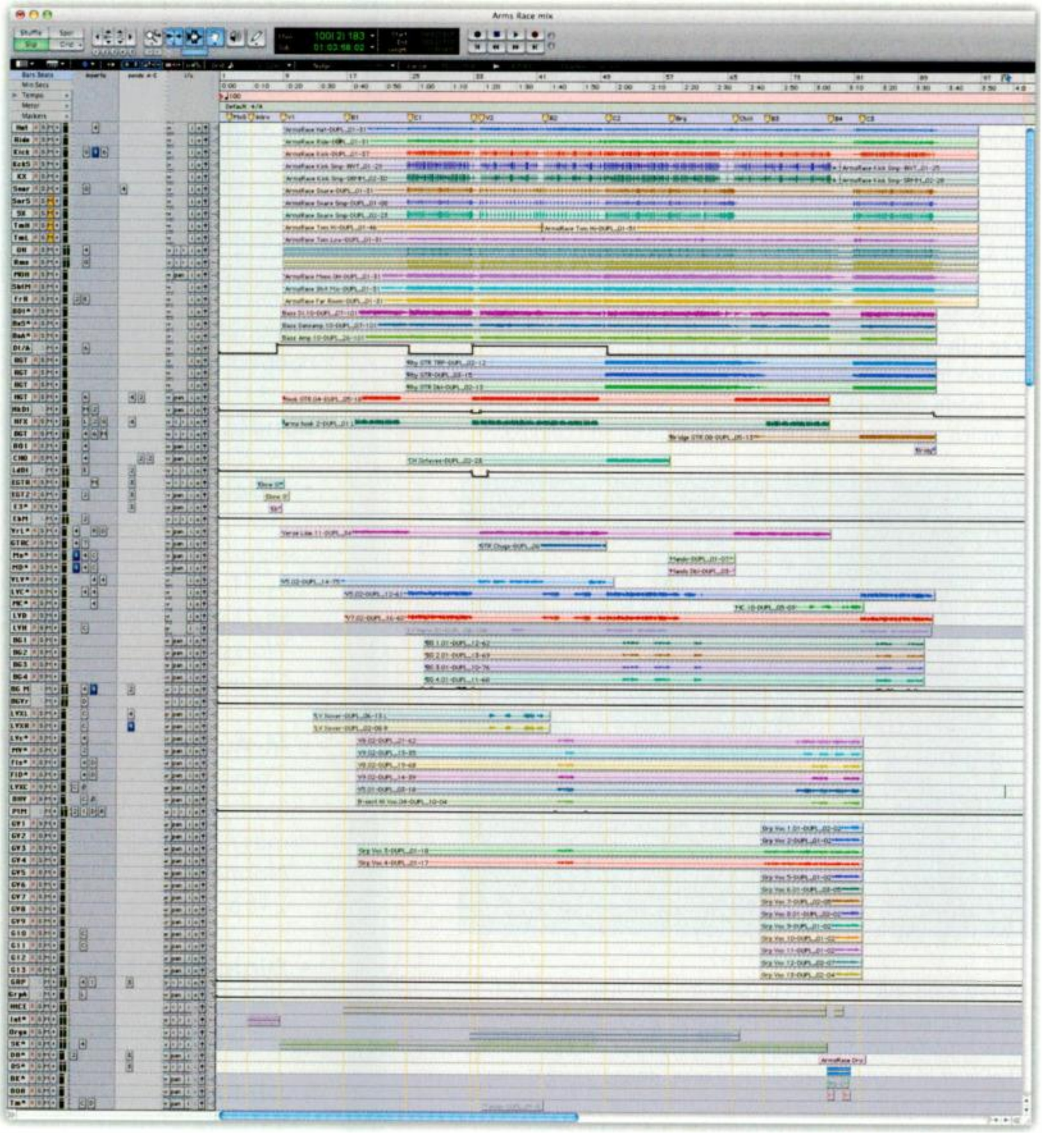
"I made notes while were recording Andy, deciding on the fly which sections of which takes were good, so I could give

my Pro Tools guy a road map, saying something like 'Take the verse from this take and the chorus from that take,' and so on. I prefer to edit in large chunks. I'm not necessarily a believer in micro-editing drum tracks, unless we're trying to lock up the drums to a loop or a programmed groove Wherever possible I prefer to leave the original feel in there.

"I like to use Pro Tools as a tape recorder, and this is partly the result of me enjoying the music of the past, which was not edited to death and in many cases not even played to click track. When I listen to a Beatles or Rolling Stones track I don't say 'Listen to how out of time or tune that is.' Instead I just notice that it feels really good. So if you have good musicians playing, you should let them do their job. ▶

Out With The Outboard

Unusually, Neal Avron always records through the desk, in contrast to many modern engineers, who prefer outboard mic preamps. "I have an issue with the whole mic preamp situation," explains Avron. "I'm simply not fond of having a kick drum preamp on the left of my control room, a kick EQ behind me, and a compressor somewhere else again. I prefer to have everything in front of me, all EQs and faders and compression, on a desk. It means that I don't have to think about so many outboard boxes everywhere, and instead I have more space to listen to the sounds and focus on the music. I've listened to a lot of mic preamps, and there are differences, but I've found that by the time you get to mixing the music, there's going to be more EQing and compression and panning and whatever other treatments that shape the sound, so as long as the console has its own high-quality mic preamps, the convenience of having everything in front of me far outweighs any sonic difference."



This composite screen grab shows the Edit Window for 'This Ain't A Scene, It's An Arms Race'.

I'd rather have a performer play several takes than create a performance out of bits they've done. And I never listen to the click when I hear the drummer play. I'm not interested in how close he stays to the click, that's his job. My job is just to listen for when it feels great. Perfection is called for in certain types of music, but for me, it's not always so appealing."

The Rhythm Method

Many engineers and producers like to get the rhythm section down first, and will often record drums and bass at the same time, or overdub the bass next. But Avron goes for rhythm guitars as the second step. "The drums are edited in Pro Tools, and of course, its editing facilities are undeniable. The Undo button is such an amazing and magical thing.

"Once the drums are edited, typically with rock songs I will record the rhythm guitars next. Over the years I've had issues with recording bass first, especially when someone is hitting the strings really hard. For me it's difficult to tell whether the bass is in tune, because the fundamental is so low. When laying the rhythm guitars down first, it's much easier to tell whether the bass is out of tune or not. It also means that the bass has a place to fit. In the 'Arms Race' track we did the heavy rhythm guitars in the choruses first, to get a sense of how distorted and loud they were going to be.

"Once we had enough rhythm guitars down, we put the bass guitar down, and after that we overdubbed other guitars and instruments. I really like to record the singer as early as possible. The moment we have enough music for him to actually sing to, I like to get him going. It's a good opportunity to get a few takes under their belt, and you either get a great take, or he can figure out what he likes and doesn't like, and have another shot at it later on.

"On 'Arms Race', after the guitars, the bass and the lead vocals, we recorded the rest of the vocal production, first what are called the 'BG' tracks in the Edit Window, and then the 'V' tracks, which were sung by Patrick in different accents and with different voices, falsettos, basses. He's an amazing singer. Finally there were the 13 tracks of group vocals at the end. Along the way we also recorded the keyboards, including a Hammond B3, and the intro is a synth with a guitar played with an E-bow and some pedals I was messing around with, among them a Fuzz Factory pedal, and Electro-Harmonix Micro Synth and Memory Man pedals.

"I am a firm believer in recording things the way I think they should sound in the final mix. I don't rely on the idea of recording everything flat and fixing things in the mix. To me, the sounds that are being generated are integral to the vibe of a song. If a song needs punchy drums, yet they have been recorded fairly loose-sounding, trying to figure out how all the other sounds are going to relate to them is going to be impossible. So for me, while recording, each sound has to really represent what we're hearing in our heads, and that means adding EQ, compression, pedals and other effects."

Drum Mics

"Regarding the mics that I use, I always try and change them, but I have certain regulars that I go to," recalls Avron. "Starting with the drums, on 'Arms Race' the kick mic was probably a *[Neumann]* 47 FET on the outside, and there are about three or four different mics that I'll use on the inside. The snare is typically an SM57 on top, and if I mic the bottom, also a 57. For the hi-hat I may have used a *[Neumann]* KM84, for the ride cymbal also a KM84 or perhaps a *[AKG]* C451. I change the overheads all the time; in this case I recorded a mono overhead as well, probably using something like a *[Neumann]* U67 or an AKG C12. The 'far room' mic was probably *[Neumann]* U87 or AKG 414 or a 67, sometimes I use a C12. I really mix that up. I also had what I called a 'shit mic', a low-quality mono mic,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just to see if it added an interesting vibe to the track.

"In the second Edit Window at the bottom you can see a sound called 'SK', which is a four-to-the-floor kick-drum sample that we used in the verses. It's a sound that comes from Patrick's original, pre-production demo, and it really drove the verse. It had a vibe, and we looked for

some replacement kick-drum sounds, but none had the energy that this one had. So he gave it to me via a USB drive, and we loaded it into Pro Tools and programmed it. There's also 'DS', which is a shaker, and some other drum samples, one being like an 808. There's a mono track called 'D4' in the screenshot, which is the drum kit recorded with just one microphone, it may have been an Electro-Voice 666, and squashed pretty hard with a Neve compressor. This mono drum track was used in the breakdown section, where you hear only drums.

"On the first Edit Window you'll also see some drum samples, in this case a snare and a kick sample. Once we have the drum sound and the tuning that we like, I'll record kick and snare and tom samples from the kit, for a couple of reasons. Mainly, after each take the tuning of the drums will go out of whack, and the sample gives me a tuning reference. If the tuning changes during recording, the drum tech can use it as a reference to re-tune the drums. That way, while editing takes, it doesn't sound like the snare is changing pitch.

Avron also added a small bass lift to the bass guitar, with a mid boost at 1.5kHz automated to kick in when needed to compete with the guitars.

The other thing is that if the drummer is playing a very fast pattern and can't quite keep the same velocity going in his attack, and the part won't quite cut through, I can change over to the sample for a few bars, to get a better level, with exactly the same sound. In general I try to take care to get the samples from the actual drum kit, and not use samples from elsewhere.

"I don't recall doing anything crazy when recording the drums for 'Arms Race'. I compressed the mono 'far room' probably 4:1, sometimes I'll do 2:1, and the shit mic was pumping pretty good too. I probably compressed the overhead just a tad, usually about 4:1. This was using the Neve desk; if it was outboard it would have been stuff that was built in the room. I love the Urei 1176 and the Neve compressors, like the 2264, 33264 and 33609. They all sound great. At Ocean Way they have some great Fairchilds and old RCA limiters, and if I'm working there, I'll use those. I'm not set on one piece of gear: whatever sounds great. Unless I'm looking for something in particular, I don't like to compress the drums too much, because I already use tape compression, and I don't want to flatten the transients."

Guitars & Bass

"With regards to the guitars, like with the drums, we spent a lot of time getting them to sound great at source. We tested many different amplifiers and cabinets, as well as a bunch of different guitars, trying out whether we want humbuckers or single-coil, etc. I tend to use an SM57 on the speaker cabinet and a Coles or Royer ribbon microphone. I might have had a Neumann U87 or a U67 mixed in as well. Usually I'll put up several different mics so I can choose the ones that give me the colour we're after. I would have used a compressor, probably the Neve 2254 compressor they have at The Pass, or perhaps a UREI, I'm not sure.

"For the bass, the mics I use range from the Electrovoice RE20, AKG D112, or Neumann 47 FET, U87, U67, sometimes a Sennheiser 421. For this track, it might have been a 47 FET or a U67, but I might have had an SM57 mixed in as well. I probably used the same compressors as for the guitar. I recorded Patrick's vocals with a U47 tube in most cases, a U67 in other places, and compressed with a Neve 33264, a Distressor or an 1176. The vocal compression is not so much about level adjusting as about the sound it gives, giving the vocal the right attitude so it sits better in the track. It may have been 4:1 or 6:1, perhaps even 8:1. Typically I'll hit the vocal pretty hard."

'This Ain't A Scene, It's An Arms Race'
Writers: Fall Out Boy
Producer: Neal Avron

Given the care Avron takes in recording his sounds, it comes as no surprise that he describes his mix process as mostly being about balancing, though he emphasises that "even in the cases where I haven't recorded a track, a big portion of my mix time is still dedicated to balance. Since I had recorded and produced 'This Ain't A Scene, It's An Arms Race', I already had my vision for the track. Basically, the challenge in mixing it was twofold. The verses were all about the four-to-the-floor beats and the vocal production, while the guitars were treated like a loop: they are hypnotic and lodge in your brain and then you kind of forget about them. They become part of the fabric.

"What really sticks in your mind is this incessant beat coming from Patrick's kick sample, and his lyrics and his delivery of them. The choruses go into this big, fast rock movement, and the trick was to make the verses big and thumping, while the choruses are also huge, in their own way. The choruses are too fast to be big and thumping, so they had to be more aggressive in the mid-range with the guitars and at the upper end of the drums. I knew that getting that balance right was the key to the song working.

"I know that this will sound oversimplified, but the key thing here really was balancing. It took a little while to work out how the choruses would sound, and then I backtracked to the verses to figure out where things would sit volume-wise. There wasn't any guesswork or any crazy processing going on; all the sounds already worked in the way that they were intended to. You'll notice on the screenshots that the EQ settings are very minor. Yes, there are lots of plug-ins on the Edit window, but in many cases I'll already have set the EQ for the whole part on the desk, and the plug-ins then just work on a small section of the part."

The desk that Avron refers to is the 56-input, 6056 SSL E/G series that's in Paramount Studio A in Hollywood. "I went there for the mix for a combination of reasons," explains Avron. "First of all, the recall ability of the SSL, and secondly, the sonics. Having compression on every channel, as well as a well-functioning four-band EQ makes things very easy. As I said before, I like things in front of me: it makes grabbing knobs easier and faster. I do still feel very comfortable mixing on a console.

"I do some submixing in Pro Tools, however: regardless of how many tracks are in the Pro Tools file, I usually mix down to 40 channels. The SSL has only 56 inputs anyway, and not all tracks have the same importance. I don't need to have those 13 group vocals separate, I prefer to mix them down to stereo for the end mix. Having fewer channels also makes the end mix more manageable. In the days when I was mixing Everclear records, I used 70-80 channel desks and brought back every track individually. It was doable, but very cumbersome. Pro Tools gives me more flexibility, because the automation is great, I can change balances from section to section, or automate EQ settings, and as always there's this Undo button.

"With 'This Ain't A Scene, It's An Arms Race' I worked my way up from the bottom to the top, starting with drums, then the bass and the guitars, until I had the rhythm section happening. I checked the vocals regularly, to see how they were holding up in comparison, and then finished the track working on them."

Digidesign's Lo-fi plug-in helped to make the keyboard sound gritty.

Drums: SSL EQ & compression, McDSP Filterbank E6

"The 'G' that shows on the kick drum in the Edit Window is a gate; the '4' is probably an EQ, but I didn't use it; and the '6' is the McDSP Filterbank E6, which boosts the low end a little around 154Hz and takes out mid-range around 400, just to give it a bit more definition. I probably used very little outboard gear on the kick drum, perhaps just a little compression and EQ on the board. I hate to be boring, but I had already worked hard at getting exactly the sound I wanted during the recording process. I recorded all the other parts around the kick drum I wanted, so I knew that they also fit together. Mixing in this way becomes more like mastering, ie. fine-tuning rather than surgery. I muted the overheads and the room in the verses, so Andy's drums really blended in well with the sample kick, and it sounds like someone programmed the drums, rather than Andy playing along with a sample kick."

Bass: McDSP Filterbank E6

"The bass had some touch-up EQ, again the Filterbank E6, just adding 1.5dB at 65Hz in the choruses. Once the guitars start pumping, the bass gets a little lost, so I just automated a little extra mid-range to let the bass compete. It's a very subtle thing. You can see on the Edit Window that I recorded the bass DI, via a mic on the cabinet, and via a Sansamp. For the mix I combined these three and put them on one Aux track, and that came up on one fader on the console."

Guitars: SSL EQ, McDSP Filterbank E6

"The only thing I would have done to the main rhythm guitars is some touch-up EQ on the console. There's a hook guitar ('HGT') on which I applied a McDSP E6 to give some more definition, adding around 564 and 4k, to make it cut through better in certain areas. The bass and the drums in the verses are very bottom-heavy, and when listening to the mix I felt that the hook guitars could be more defined."

Keyboards: Digidesign Lo-fi, Focusrite EQ

"There's also hook keyboard ('HFX') that plays the same part as the hook guitar. The keyboard sounded so clean that I put it through a Lo-Fi plug-in to distort it and noise it up. I took it down to 6 bits and set the distortion to 11. The '4' on 'HFX' is a send, the 'L' is the Lo-Fi and the '2' is a Focusrite EQ that had -0.5dB at 14k. The next plug-in was a delay that split delays left and right, with a 4ms and a 23ms delay to help spread the keyboard."

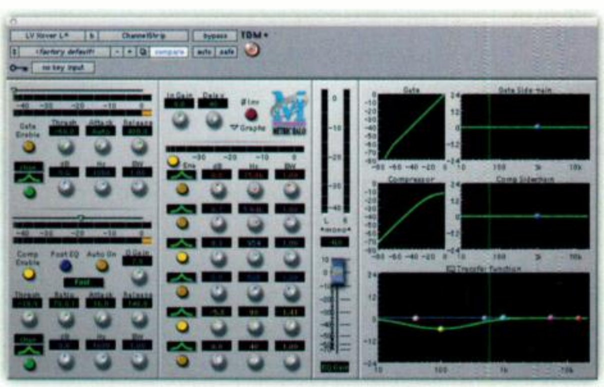
The lead vocal required different EQ settings in the verse (top right) and chorus, and was also run through a C4 multi-band compressor to tame mid-range harshness.

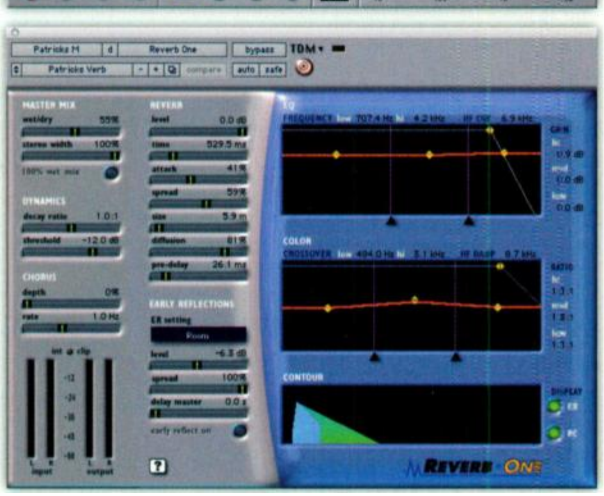
Lead vocals: McDSP Filterbank P4, Waves C4

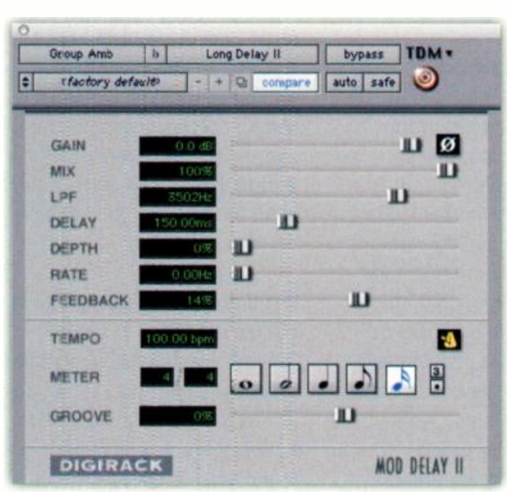
"I used two McDSP Filterbank P4 plug-ins on the lead vocal comp, one on the verse and one on the chorus section. As before, they're both very subtle, and different, due to how Patrick is singing and what register he's singing in. In the verse he was singing in a more low register, so I wanted a little bit more top end on his voice to match it to the way it sounds in the choruses, so I boost at 8.4k. On the chorus I did very little. I also have a Waves C4 parametric processor on the lead vocals. This was an experiment, as I hadn't used that plug-in a lot. There were a couple of places where things were getting a little abrasive in the upper mid-range, so I tried doing multi-band compression. It's only compressing, or lowering, those specific frequencies when he's going into that register. As for outboard on the lead vocal, I don't recall, I might have used a short room reverb, just to give it more size. Sometimes I'll use short delays, or harmonisation, to give more size. But overall this track was kept very dry, so it stays in your face."

Backing vocals: Metric Halo Channel Strip, Digidesign Reverb One and Mod Delay II

"There are three elements here: Patrick's left and right answering vocals in the choruses ('LVH'), the funny voices he did ('LVXL'), and the group vocals. I had the Metric Halo Channel Strip on the lead vocal harmony, basically just a little bit of EQ to differentiate them and set them apart from the lead vocal. I took out a little bit of low end to make them sound a little thinner, and added a bit of mid-range. I used the Reverb One on the six tracks on which he sang in all sorts of different voices. Its job is to set these vocals apart from the other ones. You can do that in all sorts of different ways — compression, EQ, panning — but in this case I chose reverb. As you can see, it's a short reverb, about half a second, with very short pre-delay. I put it on an insert of the group, and so it's set to 55 percent wet. Finally, there's the Digirack Mod Delay II on the stereo submix of the 13 group vocal tracks. Again, it shows you that there's not a lot of reverb on this track. There just wasn't much room for reverbs. For the group vocals I created space by using short delays. I think they're set to 16th notes. I either set delays right on the beat, or in the case of more moody, ballad-like tracks, a little slow. I don't like delays that rush. In more vibey songs, a slightly behind the beat delay can make things sound a little bit more laid-back." SOS







Backing vocals were EQ'd with Metric Halo's Channel Strip. A short reverb from Digidesign's Reverb One was added to some, and a delay helped to create space around them.

Zero-G Carnival Drums: The Spirit of Brazil

Acid WAV, Stylus RMX, REX 2, NNXT, Kontakt 2, EXS24, Halion

Zero-G's Carnival Drums provides both loops and individual hits and, as the sub-title suggests, aims to give a flavour of Brazilian percussion that's typical of the streets of Rio. In each format, about 1400 loops (1.4GB of sample material) are provided, and though this sounds like a lot, there's an element of duplication. Many of the performances have been recorded using multiple mic positions (close, overhead and room) and each of these is provided, along with a 'full mix' version that combines these different mic positions. The loops are complemented by over 600 individual drum hits taken from the same sessions. Usefully, the individual hits are organised into multi-layered instruments for NNXT, Kontakt 2, EXS24 and Halion (I auditioned the last) and can be used to create your own patterns or add variations to those provided by the loops.

The loops are organised into two groups: full ensemble construction kits and individual parts. The ensembles feature 10 percussionists (Samba Baterias) and are dominated by 24

different performances, each presented as a full mix and the individual mic performances. The full mixes sounded excellent — brimming with power and offering excellent ambience — but the different mic positions provide useful flexibility if you need more control over the degree of 'room' in the sound. Samba is the name of the game, but there's a range of flavours, including rock, reggae and maxixe (Brazilian tango), as well as various tempos. A sub-folder of 'Ready To Go Mixed Loops' provides additional rhythmic variations, though they're not provided in multiple mic positions.

The Individual Parts group is organised into eight sub-folders, based on drum type: Agogo, Caixa (a type of snare), Pandeiro, Repinque (a metal drum), Surdo de Corte, Surdo de Primiera e Segunda (the two lowest pitched bass drums), Tamborim and Timbal. In each case, a number of loops is provided and (as well as the multi-mic options) these are duplicated in '2 drummer' and '10 drummer' formats, allowing the user to select between a more intimate, smaller sound and a more dramatic, larger one.

While there might be fewer performance variations than the headline '1400 loops' might at first suggest, this is compensated for by the excellent quality. The full ensemble performances sound very authentic, and have obviously been well-recorded and well-played. For media composers needing a dash of genuine Brazilian carnival atmosphere, this collection would be well worth having to hand. Carnival Drums provides an excellent slice of Rio percussion and will have you dancing around your studio — now, where did I put that whistle? *John Walden*

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Best Service Drums Overkill

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A name like 'Drums Overkill' sets high expectations, but when it comes to living up to that name this library doesn't disappoint: at over 5GB, and with more than 1200 different drum kits on offer, covering analogue and digital drum machines from the '70s, '80s and '90s, as well as acoustic drum kits, there's something here for everyone. It is unarguably — and I mean this in the nicest possible way — 'overkill'.

Using the bundled Kontakt 2 player (the library comes in several formats including EXS24 and Halion, and it can, of course, be loaded into the full version of Kontakt), the 27,000 samples are organised in a pretty intuitive way.

The myriad kits are logically grouped into sub-folders categorised by genre, with self explanatory names such as '80s Pop Dance Hits', 'Drum & Bass Kits', 'RNB-Rap Kits', 'RNB Pop Kits', 'Real Drumkits' and so on — most urban, dance and rock bases are covered.

The various kicks, hi-hats and cymbals are also organised into instrument-specific banks. There's more than drums, though: there are various percussion banks too, with instruments ranging from Agogo to Darbuka to Udu, via the more conventional fodder such as shakers and tambourines.

But the best thing about this library is the range of digital and analogue drum machines. The chances are that if there's a machine out there you'll find it sampled here — there are 20 Roland machines alone, not even counting the Boss branded models! Huge velocity layers are not available, but the sounds are all well recorded and the patches easy to find.

The acoustic kits feel a little more limited. They're not as expressive as the kits you get in, say, BFD, though the sounds themselves are perfectly useable and should be well up to the job for programmed pop and urban styles.

To make the most of this library you really need to use Kontakt 2 or the bundled Player, which comes with useful effects and processing options, including compression, distortion, saturation, bit-crushing, delay and reverb — so there's plenty of scope for sound-mangling. The graphical 'skins' for each Kontakt instrument weren't to my taste, but they're clearly laid out and should appeal to someone!

My only real criticism is that it is sometimes difficult to audition some of the full kits, as they aren't mapped to the GM standard — which means that if you've been programming with one instrument and want to try the pattern on a Drums Overkill kit, you'll find that you need to do a bit of remapping.

However, this is not a major gripe, and if you've time to explore Drums Overkill in depth, you'll find a drum and percussion palette that could keep you making original beats for years. You might balk at the asking price, but it isn't bad value given the breadth of what's on offer. It should be a welcome addition to your sample library. *Matt Houghton*

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Core Values Multi-core Processors For Musicians

Some music applications will completely fail to take advantage of the multiple cores of a modern CPU — but which ones, and why? We find out, and advise on how you can make best use of however many cores your PC has.

Martin Walker

Over the last couple of years, the PC musician has been offered first dual-core processors, then quad-core models, and octo-core machines (currently featuring two quad-core processors) are now available for those with deep enough pockets. Competitive pricing has already ensured a healthy take-up of DAWs based around a quad-core CPU, yet many users haven't cottoned onto the fact that not all software benefits from all these cores. Some existing software may only be able to use two of them, reducing potential performance by a huge 50 percent, while older software may only be able to utilise a single core, reducing potential performance to just 25 percent of the total available. This month PC Musician investigates which audio software works with dual-core, quad-core PCs and beyond, what benefits you're likely to get in practice over a single-core machine, and which software may for ever languish in the doldrums.

A Brief History

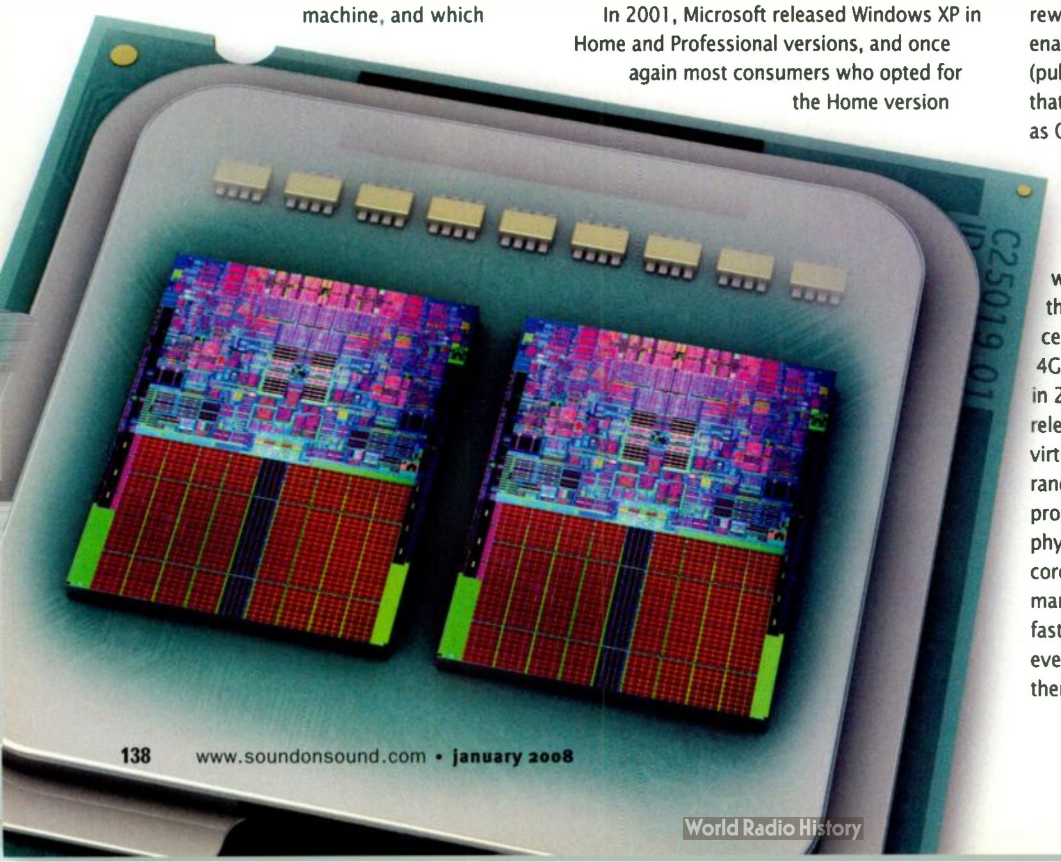
In the days when most musicians ran Windows 95, 98 or ME, the question of running multiple processors didn't arise, because none of these operating systems supported more than a single CPU. It was Windows NT and then Windows 2000 that introduced us to the benefits of being able to share the processing load between multiple CPUs: Windows 2000 Professional supported one or two processor chips, while the more expensive Server version supported up to four, and the Advanced Server up to eight. However, at this early stage each processor was a physically separate device, so to be able to (for instance) use twin processors, you needed a specially designed motherboard with two CPU sockets. Many audio developers and interface manufacturers didn't actively support Windows 2000, so most musicians stuck with Windows 98.

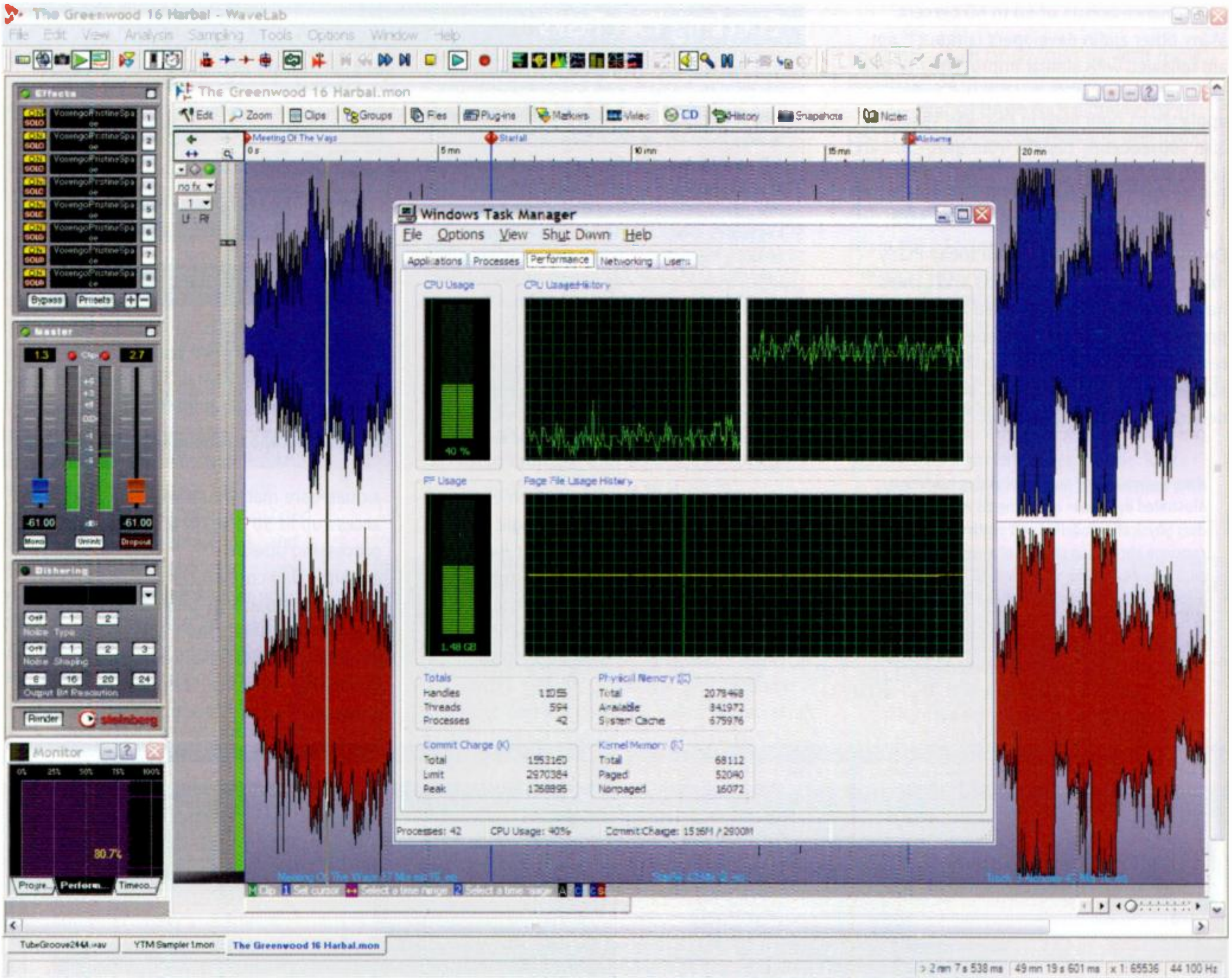
In 2001, Microsoft released Windows XP in Home and Professional versions, and once again most consumers who opted for the Home version were limited to a single physical processor, although the Professional version supported two. By this stage many musicians were straining at the leash, wanting to run more and more plug-ins and software instruments, and this Professional version let them do exactly that, using dual-processor motherboards and twin Xeon or Pentium 4 processors.

Multi-processing options really opened up the following year, when Intel introduced first Xeon and then Pentium 4C processor ranges with Hyperthreading technology, which let these CPUs appear to both Windows XP Home and Professional (or Linux 2.4x) as two 'virtual' processors instead of one physical one. They each shared the various internal 'sub-units', including the all-important FPU (Floating Point Unit), but could run two separate processing 'threads' simultaneously.

Intel claimed up to a 30 percent improvement with specially written applications over a standard processor, but as many musicians soon found, having a Hyperthreaded processor didn't necessarily benefit them at all unless they were running several applications simultaneously, since applications like MIDI + Audio sequencers had to be rewritten to take advantage of Hyperthreading. Steinberg's Nuendo 2 was one of the few music apps to support it, but although various others followed, a few (such as Tascam's Gigastudio) needed a major rewrite before they would even run with HT enabled. Nevertheless, my own tests (published in PC Notes June 2004) showed that with optimised audio applications such as Cubase SX2 you could expect a significant drop in CPU overheads where it really mattered, at low latencies of 3ms or under.

The biggest change came in late 2004, when both AMD and Intel seemed to agree that processor clock speeds had reached a ceiling. Intel abandoned plans to release a 4GHz model in their Prescott CPU range, and in 2005 both companies largely switched to releasing dual-core models. Unlike the twin virtual processors of Intel's Hyperthreading range, these featured two separate processing chips mounted inside one physical package. By placing two processor cores into a single piece of silicon, manufacturers could provide significantly faster performance than a single processor, even when under-clocking them and running them at lower voltages, so that they didn't





When you're running stereo audio editors (such as Wavelab 6, shown here) and stand-alone soft synths or samplers, and even in most multitrack sequencers when you're only running a single track, only one core of a multi-core CPU will be heavily used, although any others available may help with disk access, the user interface and other applications that are running simultaneously.

run hotter than the single-core variety.

By late 2006 we had been introduced to quad-core processors, which have now dropped in price and can even be run with Windows XP Home (which is licensed to run a single physical processor, however many cores it has inside). However, if running XP Professional (and the x64 64-bit version), Vista Home Premium, Business, Enterprise or Vista Ultimate you also gain the option of installing two quad-core processors on a suitable motherboard, to provide a total of eight processing cores. Unfortunately, as with so many new hardware advancements, much software has had a long way to catch up before it could take advantage of so many cores.

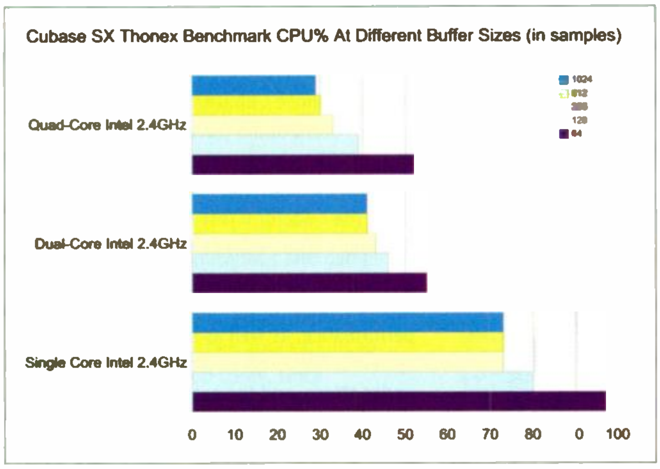
Multiple-threaded Applications

When you're using a PC with multiple processors of whatever type, to gain any significant performance benefit the software you run has to be specially written or adapted with multiple processors in mind. The way multi-processing works is that applications are divided into 'threads' (semi-independent processes that can be run in parallel). Even with a single processor there are huge advantages in this programming approach. Many applications use multiple threads to enable multi-tasking, so that one task can carry on while another is started; and when multiple processors are available, different threads can be allocated to each CPU.

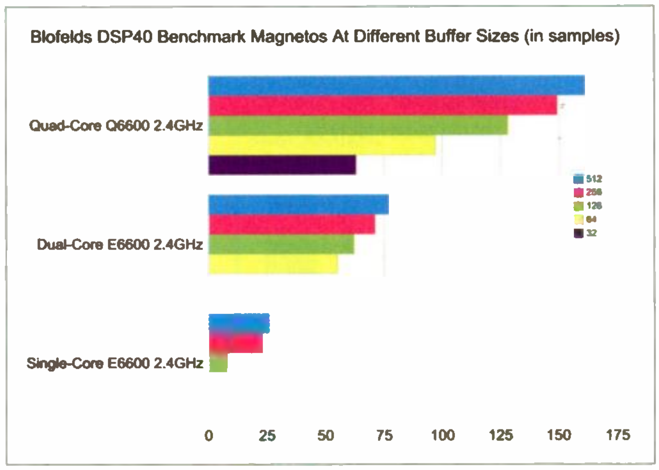
With some processor-intensive programs, such as 3D graphics and CAD software, it's comparatively easy to split off different functions to each processor. However, the situation becomes somewhat more complicated with an application such as a MIDI + Audio sequencer, since all the different tracks are generally being streamed in real time and must remain in sync.

Early schemes used by audio software for sharing tasks between multiple processors were fairly crude; they tended to devote each CPU to a specific duty, so that (for instance) audio mixing and effects were handled in one thread, MIDI processing in another, and user interface responses in yet another. When a MIDI + Audio sequencer is run with several identical processors under such a scheme, the entire audio-processing workload is normally handled by one processor, with any remaining tasks left to the others. Since audio processing is by far the most significant overhead for any music application, this approach resulted in a typical overall performance improvement of just 20 to 30 percent for a dual-core processor over a single-core processor running at the same clock speed.

To gain further improvement, you need to split the audio processing in some way between the various CPUs, so that it can be processed in parallel. This means added code and complexity, and rather explains why some audio software really benefits from four or more cores, while some doesn't. Steinberg introduced their 'Advanced Multiple Processing Support' on Cubase VST version 5, splitting the audio processing between the processors and giving much larger



The older Thonex benchmark masks the true performance of systems with four cores and beyond, here displaying a very modest performance increase of no more than 30 percent between identically-clocked dual-core and quad-core systems.



These DAWbench Blofelds DSP40 results illustrate a much healthier scaling from two cores to four, and although the number of Magneto plug-ins isn't the whole story (there are a smaller number of Dynamic and EQ plug-ins also being run) it nevertheless shows that the small extra cost of a quad-core CPU over a dual-core model can give you almost double the performance — as long as the load is shared well between the cores.

your audio application may also affect the ideal number of cores, and thus which is the 'best' PC for the job. For drummers and vocalists monitoring their own live performances on headphones, the Holy Grail is to run a system that runs with barely discernible latency. Many would be happy using a buffer size of 64 samples, which would mean a total real-world latency for audio monitoring with plug-ins of just under 5ms (at a sample rate of 44.1kHz), or around 3.5ms for playing soft synths. If you still find this unacceptably high and prefer not to rely on 'zero latency' monitoring solutions (which bypass any plug-in effects), 32-sample buffers would offer total audio monitoring latency of around 3.5ms (around 2.7ms for soft synths), again at a 44.1kHz sample rate.

Blofelds DSP40 tests by a range of DAW builders who have access to lots of PCs based around different processors have shown that at really low buffer sizes, such as 32 samples, a single quad-core processor will always outperform a single dual-core processor or (more interestingly) a system featuring two dual-core processors, and sometimes even a dual quad-core system. In some tests at these really low latencies, when stressed with lots of plug-ins and instruments, the single quad-core machine was the only one to complete them successfully, making it the current king for low-latency performance.

If you're happy to run use a higher buffer size, of 128 samples or above (audio monitoring latency of around 8ms), you'll probably be able to run significantly more plug-ins and soft synths using two quad-core processors than one. Those involved in lots of recording work who want 'real time' monitoring may thus prefer a single quad-core, while others who rely mainly on samples and soft synths may get even more mileage from a twin quad-core system.

Hang On, I'm Busy

While it's possible to specifically assign each Windows programming task to a separate processor, you can also let Windows handle its CPU resources dynamically across a single processor by giving each task a specific priority. The lowest priority is nearly always given to the user interface, which is why screen updates can get sluggish on a single-core machine when you run lots of real-time software plug-ins.

Conversely, any PC with multiple cores is always likely to remain more responsive even when most of the cores are stressed, because the user interface is still happily ticking away on another one. Even if you're running elderly applications that are not multi-threaded, you can still benefit from a dual-, quad- or octo-core machine if you're running several such applications simultaneously, as Windows will allocate each one to a different core.

Which Audio Apps Benefit?

This is the biggie: it's all very well having a hugely powerful quad-core or octo-core PC, but not a lot of use if your software only uses two or four cores from those available, or makes a poor job of sharing resources between them. The secret is for the application to balance requirements across the available cores, so that you don't get any audio glitches as a result of one or more cores running out of juice while there's some still available from the others.

For the reasons mentioned above, stereo audio editors may not take full advantage of a multi-core PC — something I soon confirmed with Steinberg's Wavelab 6, which only used one core for DSP processing during playback or audio rendering. Its author Philippe Goutier says that a second core will be used for disk access and the user interface, which does at least mean that the application will always remain responsive to new commands, but he hopes to improve core-sharing now that so many musicians have multi-core PCs.

The vast majority of stand-alone soft synths also seem to mostly use a single core, but as soon as you load the VSTi or DXi version into a host VSTi or DXi application, this host should distribute the various plug-ins and soft synths across the available cores to make best use of resources. Fortunately, most multitrack audio applications can distribute the combined load from all your tracks between as many cores as they find, although it's perhaps inevitable that since many of the latest versions were released long before quad-core and octo-core PCs were in regular use, some don't manage it quite as efficiently as others. Even now some developers don't have octo-core test systems.

The DAWs

Reaper's Justin Frankel told me that he routinely does a lot of his development on a dual quad-core Xeon PC, so it's hardly surprising that the default Reaper settings work well with up to eight-core machines, typically offering over 95 percent utilisation of all eight cores. Reaper mostly uses 'Anticipatory FX processing' that runs at irregular intervals, often out of order, and slightly ahead of time. Apparently, there are very few times when the cores need to synchronise with each other, and using this scheme he can let them all crank away using nearly all of the available CPU power. Exceptions include record input monitoring, and apparently when running UAD1 DSP cards, which both prefer a more classic

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'Synchronous FX multi-processing' scheme.

Steinberg's Cubase SX, Cubase 4 and Nuendo all work decently on quad-core systems, scaling up well from single to dual-core and quad-core PCs. However, Cubase 4 and Nuendo 4 don't currently provide all the benefits they could at low latency with a dual quad-core system. Compared with the potential doubling of plug-in numbers from dual to quad, when you move to 'octo' you may only be able to run about 40 percent more plug-ins down to buffer sizes of 128 samples, while below this you may even get worse performance than a quad-core system.

Steinberg developers have already acknowledged the problem, which is apparently due to "a serialisation of the ASIO driver, which eats up to 40 percent of the processing time. Together with the other synchronisation delays, only 25 to 30 percent of the 1.5-millisecond time-slice can be used for processing. This is not very efficient." Steinberg have promised to address the issue in a Nuendo 4 maintenance update, and have hinted that it may also result in changes to the ASIO specification.

Cakewalk's Sonar does seem to scale well, sometimes giving a better percentage improvement when moving from a quad-core to an octo-core PC than the current version of Nuendo/Cubase 4, but the jury still seems to be out on whether choosing ASIO or WDM/KS drivers gives better results; with some systems ASIO is a clear winner, while in others WDM/KS drivers move significantly ahead.

Digidesign have a reputation for being slow but thorough when testing out new hardware to add to their 'approved list', and as I write this in early November 2007 their web site states that Intel Core 2 Quad processors and Intel Xeon quad-core have not been tested by Digidesign on Windows for any Pro Tools system.

Nevertheless, Pro Tools HD/TDM users started posting recommendations for rock-solid systems featuring twin dual-core Opteron processors (four CPU cores in all) in mid-2006, and there are now loads of Pro Tools LE users successfully running both quad-core and even a few octo-core PCs in advance of any official pronouncements (there's lots of specific recommendations on both quad-core and octo-core PC components in a vast 126-page thread on the Digi User Conference at http://duc.digidesign.com/showflat.php?Cat=&Number=988224). Despite the lack of official 'qualification', all Pro Tools systems seem to scale well on quad-cores, happily running all four cores up to 100 percent utilisation, and many users are very pleased with their quad-core 'native' CPU performance.

Like various other audio applications, even the latest Mac version of Logic Audio doesn't yet fully benefit from having eight processor cores at its disposal, but for die-hard PC users of Logic the situation is rather more serious: Apple discontinued development and support for those using Logic on the PC back in 2002, so most recent version (5.5.1) is now some five years old. Although it's a multi-threaded application, Logic 5.5.1 for Windows is not really optimised for multiple processors, so only one of the cores is likely to get much of a workout. However, there's a partial workaround, using the I/O Helper plug-in available from Logic version 5.2 onwards, which can force any plug-ins on a track with it inserted to run on a second core, so that you can use lots more plug-ins/instruments overall (there's a more detailed description on Universal Audio's web site at www.uaudio.com/webzine/2003/may/index5.html). Logic Audio 5.5.1 also has a problem if more than 1GB of system RAM is installed (see http://community.sonikmatter.com/forums/lofiversion/index.php/t8032.html for some suggestions on this one), and also has problems running some VST plug-ins. It's unlikely to benefit from a quad-core processor at all, and I wouldn't recommend running it on a new quad-core PC, so its shelf-life is looking increasingly limited.

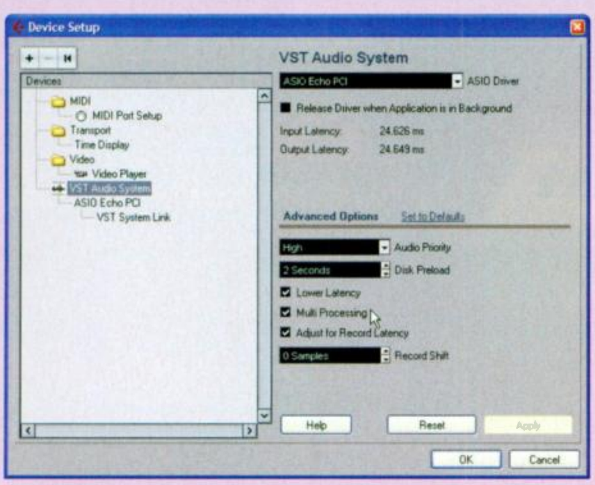
Overall, getting the best out of a multi-core PC generally means a little detective work from the user. You need to make sure you have the most appropriate audio application settings (which might be different if you run DSP cards), and you also need to be cautious when running heavy-duty synths or plug-ins that might consume one of your cores in a single gulp. Keeping an occasional eye on the Windows Task Manager may also help, since the CPU meters provided by most sequencers are becoming rather less useful now that they are monitoring so many individual cores. SOS

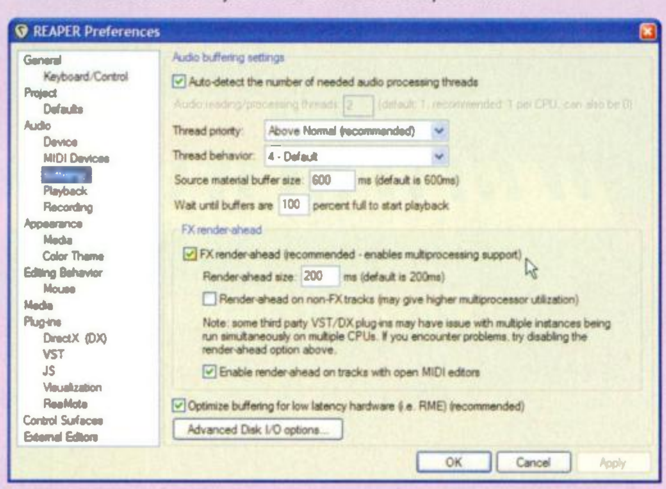
Special Settings

Before coming to any conclusions about the multi-core performance of your particular sequencing package, make sure you have any appropriate parameters set correctly. For instance, in the case of Cubase/Nuendo you'll need to tick the 'Multi Processing' box in the Advanced Options area of the Device Setup dialogue, while for Sonar the tick-box labelled 'Use Multiprocessing engine' is the one to check. With these settings deactivated you'll only be using one of your cores, and performance will plummet.

In Reaper, most multi-core users will need to tick the the 'FX render-ahead' option in the Audio Buffering dialogue to enable the full benefits of native plug-in multi-processing. Universal Audio UAD1 owners should leave this option un-ticked, however, because of current UA driver issues.

Audio applications (such as Cubase 4 and Reaper, shown here) tend to have specific tick-boxes to allow you to enable multi-processing support, so make sure these are activated if you want to achieve the best performance.





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Native Instruments Kontakt 3

Software Sampler For Mac & PC

It's over five years since Native Instruments released the original version of their flagship soft sampler, and its third incarnation takes the Kontakt concept even further, with a streamlined user interface, a new waveform editor and a massive sample library.

Nick Magnus

The jump from Kontakt version 1 to version 2 brought a huge number of improvements to all areas of Native Instruments' software sampler. Amongst the enhancements were a more attractive cosmetic appearance, easier modulation routing, more flexible effects routing, a variety of new effects including a convolution reverb, a searchable database, instrument banks, universal file import and the KSP script engine, to name just a few highlights. To fully appreciate the differences, it's worth catching up with the reviews of Kontakt 1 and Kontakt 2 in the August 2002 and July 2005 issues of *SOS*. The increment to Kontakt 3, whilst not being quite as all-encompassing as the last major integer change, brings some perhaps less dramatic but nonetheless beneficial improvements over Kontakt 2. Included in the roster of changes are numerous workflow enhancements, a substantially reworked and turbo-charged waveform editor, an improved mapping editor, a speedier and more elegant database, some new effects, another cosmetic makeover and a generous 33GB sound library that spans five DVDs.

The program installs alongside any previous versions (should you have any), ensuring that projects using Kontakt 1 or 2 will continue to function correctly. The Kontakt 3 library can be installed at the same time, or at a later date, to your destination of choice. You can also opt to install the entire shooting match, or just specific parts of the library. See the 'Kontakt 3 Library' box later in this article for more information.

Product activation is via the NI Service Centre application included on the installation disk, which allows for both on-line and off-line registration. Kontakt 3 itself works on both PC (under XP and Vista) and Mac, as a stand-alone program and a plug-in that is compatible with VST, Audio Units and RTAS (for Pro Tools 7). Interestingly, NI have chosen to abandon the DXi version.

Who's A Pretty Boy Then?

Kontakt 3's new cosmetics are immediately apparent. Much as I liked the slightly dour, militaristic look of K2, K3 has an altogether more modern, friendly appearance and a fresh, neutral colour scheme that helps clearly differentiate one section from another. Before we go any further, it's worth noting

Kontakt 3 Library

Although part of the K3 library duplicates that of K2 (principally the excellent VSL Orchestra collection), if you have K2 and if you have plenty of spare disk space, it's worth retaining the entire K2 library alongside it. The difference is 'only' around 7GB (I can't help but smile wryly as I type that!). While K2 instruments will load into K3, K3 instruments are not backward compatible, so if you delete the K2 duplicates but wanted to return to using K2 for whatever reason, the K3 library instruments would be unusable in that version.

The 33GB K3 library consists of around 1000 instruments organised into six categories; although space restrictions make it impossible to describe the instruments in detail, an overview of each category below gives an idea of what to expect. All instruments in the library make use of Performance Views, the extended 'skinned' panels below each instrument that contains a set of appropriate performance controls. These controls can affect anything from cutoff frequency and resonance, to effects parameters, keyswitchable articulations, legato and hammer-ons, harmonising and more. All Performance View parameters can be automated simply by dragging controllers onto them from the Browser's Auto pane.

- Band: everything you need for a virtual band line-up, including the constituent parts of a horn section, acoustic and electric pianos, loads of organ registrations sampled from a real Hammond, guitars, basses and drum kits.
- Orchestral: extracts from the famous VSL library, pianos, pipe organ, harpsichord and orchestral percussion. This is an almost identical collection to that supplied with Kontakt 2, but with some variations — and, of course, the Performance Views offer enhanced playability.
- Synth: an eclectic range of electronica, but not just the usual pads, leads and basses. Well, they're here, but you also get a diverse range of musical automata, including arpeggiators, mini-sequencers and synth-style drum beatboxes that make extensive use of KSP Scripts. These come ready loaded with groovy patterns and beats, but they can all be reprogrammed to do as you wish.
- Urban Beats: rather like the beatboxes of the synth category, these groove generators offer complete drum-based backdrops in a distinctly hip-hop/R&B flavour. Each instrument also includes a fully playable kit of parts with detailed control over each sound's effects, levels and articulations. The groove loops have their own level and effects controls and are also reprogrammable, making each Urban instrument a very flexible toy, and fun into the bargain.
- Vintage: ranging from classically trendy to shamelessly camp, this category features raw, unprocessed samples from a variety of vintage battleaxes. Some are respected household names (Minimoog, Memorymoog, Mini Korg 700, Logan String Melody II, RMI Electrapiano, Crumar Orchestrator, Linndrum, TR808 and 909) whilst others are less familiar, even humorous choices. The Electronic Toys folder, for instance, includes a variety of Casio Rapman sounds, Suzuki Tronichord, Yamaha Handy Sound, Mattel Bee Gees rhythm machine, a Casio SK1 drum kit and even a complete story told by a 'Droopy The Dragon' speech synthesis toy!
- World: it may not scale the same heady heights as Quantum Leap's RA virtual instrument, but K3's collection of World instruments delivers an impressive range of flavours from around the globe. Flutes, recorders, reeds, metallophones, stringed instruments, accordions and percussion make up the menu. My personal favourites are the bagpipes, including Uilleann and Highland versions, complete with drones and chanters.

A PDF manual for the library installs with K3, providing detailed descriptions of every instrument and full instructions on how their Performance Views work. It should be noted that the Performance Views were made using Kontakt's KSP Script engine, which means that the creation of customised Performance Views is reserved for those souls adventurous enough to learn KSP Scripting. I had hoped that NI would make this process as easy as drag'n'drop, but alas no; something perhaps for the future. Oh, and be warned that some of these library instruments can be quite CPU-hungry!

that K3's hierarchical architecture is identical to that of K2, which is as follows: samples are placed onto a keymap within what are known as Zones; any number of Zones can be placed inside a Group, which can be viewed as a sub-instrument and offers parameters such as filters and envelopes for everything within it. Groups are contained within Instruments, which can be seen as being rather like the patches in a synthesizer. Instruments are, in turn, stored within Banks, much like you would find on a hardware synth. Finally, up to 16 Instruments can be loaded into a Multi page, and each can be assigned its own MIDI channel. Each instance of Kontakt can have four Multi pages. Kontakt 3 offers numerous improvements to enable you to navigate all these different levels at far greater speed than before.

The main 'view' controls are clearly laid out at the top of the main screen: of the two groups of icon buttons across the top, the left group allows you to show or hide the Browser, output section, virtual keyboard and Master Control strip. The right-hand group of icons access loading/saving tasks, Options, global sample Purge and main window size. The View button at the far right collapses Kontakt to a compact version that shows just one loaded instrument at a time, which is useful for conserving screen space. A new 'i' button reveals a contextual help strip at the bottom of the main K3 window. This provides extremely useful information when you're not sure what something does; just point the mouse at a knob, button or other screen element, and the strip displays a short description of its function to guide the way.

The Master Control section, which deals with tempo sync and master tuning, has a welcome new addition: a master volume control. This affects the total output level of all Kontakt's outputs, and is a lifesaver when the combined output of all your loaded sounds takes your DAW into the red.

One of the principal new navigation aids is the Instrument Navigator pane below the two Browser panes at the bottom left. This can be shown/hidden using the 'Instr Nav' button along the top of the Browser; all three of these panes can be resized by grabbing and moving their divider strips. The Instrument Navigator helps in two ways: firstly, the instrument rack can fill up very quickly, especially when using K3 library instruments, as all of these include Performance Views that eat up a lot of screen real estate. Previously, you would have had to manually search through the rack to find the instrument of interest, but now, these are listed in the Navigator. Just click on an instrument name here, and the rack scrolls automatically to that instrument. Secondly, and most usefully, this works when editing instruments too. Using the Navigator, you now simply click on the name of the next instrument to be edited, and you're there — no closing and re-opening of editing views is necessary. Kontakt even helpfully maintains the same relative position within each instrument's editing views, so you can instantly compare, for example, the envelope settings of all your instruments with ease. The Navigator also provides instrument ▶

SOUND ON SOUND

Native Instruments Kontakt 3
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pros
- Endless creative possibilities.
- Greatly improved navigation.
- Intuitive, versatile Wave Editor.
- Enhanced mapping tools.

cons
- Some of the more complex library instruments are quite CPU-hungry.
- Creation of custom Performance Views requires knowledge of KSP Scripting.
- Not yet 100 percent bug-free.

summary
On the surface, Kontakt 3 might not appear to be a huge step forward from Kontakt 2, but dig a little deeper and you discover a much sleeker and faster user interface, great new effects, some amazingly creative new tools in the Wave Editor, and a sound library that on its own could easily cost more than the price of the upgrade.

solo and mute buttons, so there is no need to scroll down the main rack looking for these.

Further features in the main editing window help speed up your work: firstly, at the top of the window is a Group list which provides one of several quick shortcuts to selecting Groups. When lit in solid red, the 'Edit All Groups' button warns that any edit will affect all Groups; clicking on the drop-down menu cancels the 'Edit All' function and allows selection of one Group at a time for editing. Secondly, moving between modulation source routing panels and their respective destination modules is made faster thanks to the new Modulation Quick Jump buttons, which accompany all modulation routings. Instead of having to scroll manually between the two, which is often quite a distance, clicking a Quick Jump button scrolls directly to the relevant module, which itself has a corresponding button to take you back to the source routing panel. Thirdly, the Modulation Shaper now makes light work of creating smooth modulation response curves with its new 'envelope view'. Like Flex Envelopes, the curves allow for multiple break points with adjustable curve smoothing.

Test Spec

- NI Kontakt v3.0.1.
- 2.4GHz Pentium 4 PC with 2GB RAM, running Windows XP.
- Tested with Cakewalk Sonar 6.2.1 and 7.0.1.

Browser Tabs

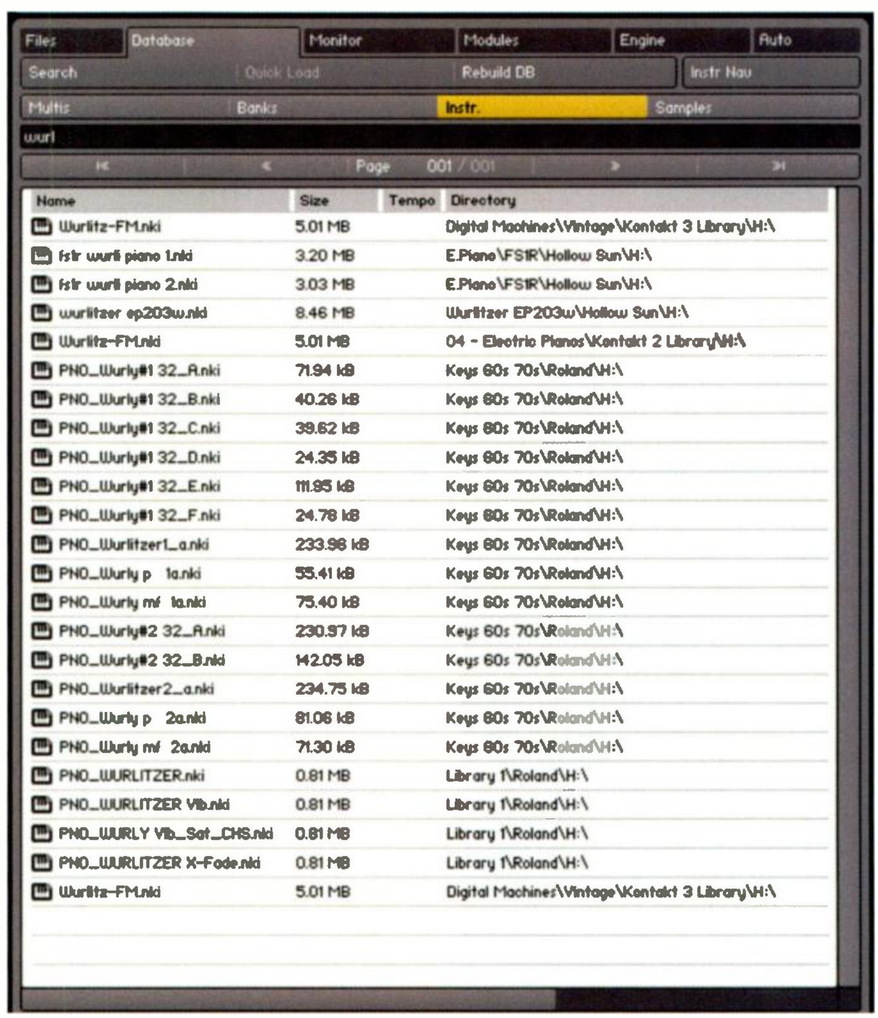
Still on the navigation theme, a Monitor button has been added to those at the top of the Browser pane, offering three tabbed views: Group, Zone and Parameter. Group view provides an overview of all Groups that make up the Instrument currently being edited, as well as being another convenient means of selecting Groups without needing to open the Group Editor in the main window. This even has its own search function, which is handy for filtering a lengthy Group list down to specific types. Zone view shows a list of all Zones within an Instrument. It's also searchable by name, and any combination of zones can be selected from here for editing. Single-clicking a Zone name automatically selects that Zone's Group, whilst double-clicking on a Zone name opens up the waveform editor in the main editing window, just as it does in the Mapping Editor. Parameter view shows the value of the last-touched parameter across all Groups of the current Instrument, or across all Instruments in a Multi if you're not in Instrument Edit mode. This is very handy for side-by-side comparisons, and displayed parameters can be edited from here by clicking and dragging their values' text with the mouse. (See the composite screenshot of all three tabbed views above.)

A composite of the three different tab views in the Browser's new Monitor page.

Not to be confused with Modulation Quick Jump, the Browser View tab also incorporates a Quick Jump function, which allows you to 'tag' up to 10 frequently visited folders in your library and go to them with a single mouse click. This is very handy if you have a large library with instruments stashed away inside many embedded folders.

The Engine tab adds one new feature: CPU Profiling Mode. When this is activated, constantly updated CPU usage is displayed in each instrument's header, and is also superimposed upon every active effect slot when the edit window is open. This makes it easier to identify which elements are responsible for excessive CPU usage.

Kontakt's searchable database has also been improved both in appearance and operation, and now performs searches at lightning-fast speeds. Simply choose the search level (Multi, Bank, Instrument or Sample) and enter a search term (the list updates instantly as you type). The screenshot below shows the results for



The improved Kontakt database enables faster searching at all levels.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a search for Wurlitzer pianos, using the search term 'wurl'. For me, the greatest improvement is that the parent directory of each search result is now also shown, making it far easier to identify suitable candidates, and to discount any ambiguous results the search may return.

Mapping Editor

New to the Mapping Editor are six tools which can assist in the laying out of key Zones. Auto-Spread Zone Key Ranges and Auto-Spread Velocity Ranges automatically fill in any holes there may be between key and velocity Zones by extending the Zones horizontally or vertically until they touch their neighbours. Root does the same for key Zone ranges, but keeps the root keys in the centre of Zones to ensure the minimum transposition from the root key in either direction. Resolve Overlapping Key Ranges and Resolve Overlapping Velocity Ranges both do the opposite to Auto-Spread; in other words, wherever overlaps occur, ranges are reduced to make the Zone transitions clean.

Auto is the most elaborate tool, and attempts to create a Zone map based upon the samples' names. This can only work if samples' names contain meaningful clues to their pitch range and/or velocity, such as 'Bass_ 64-100_C2.wav'. The names are broken down to a series of 'tokens', to which certain conditions can be applied (the example just given would break down to three tokens: name, velocity range and pitch). Using this information, Kontakt makes a stab at placing the sample where it ought to be. There is also an option to read the root key from embedded sample metadata, if it exists. Unsurprisingly, if the sample names are less than informative, like 'Piano #31' or 'sample 374', the Auto tool is unlikely to offer much assistance!

Also newly implemented is 'rubber band zooming' — this allows you to zoom in on the Zone Mapping area by holding down the Alt key and lassoing with the mouse.

Lastly, it is now possible to drag samples from the desktop, or any directory browser, directly onto the Key Mapping area. Sonar users will be pleased to know that this includes dragging samples from Sonar's Loop Explorer window, obviating the need to locate and load them from Kontakt's own browser window.

Wave Editor

Of all Kontakt 3's features, the Loop Editor has received the most substantial reworking. Now known as the Wave Editor, this offers a marked improvement over its previous incarnation, with additional tools to aid looping, beat-slicing and more. The Wave Editor can be called up manually via its own button, and also opens when any Zone is double-clicked, showing that Zone's waveform. Four sets of 'tasks' are laid out across four tabbed views, and we'll look at them each in turn in a moment.

To the right of the screen, the Grid control panel is always visible, offering two options: Fix and Auto. Fix divides the waveform into equal slices, or Zones, based upon selectable musical time values ranging from 1/1 to 1/64. These may not necessarily line up precisely with the individual beats of a drum loop, and so Fix is not the best approach to take in this case. However, there is a perfect application for Fixed slices, which we'll come to in due course. The second option, Auto, slices the loop using transient detection methods, and is more appropriate for beat slicing. The sensitivity slider adjusts the density of slice markers, and here the improvements are immediately apparent: K3's beat detection is manifestly superior to that of K2. The 'Min Slice Duration' parameter allows beat detection to ignore any low-level transients that fall within the set duration,

The Sample Loop mode of the Wave Editor.

Version 3.0.1

Version 3.0.1 had just become available for download when the review copy of Kontakt 3 arrived, so this review is based on that version. Issues addressed include cosmetic improvements, updates to the Library and a number of bug fixes. Notably, Cubase users had complained of crashes, particularly when running Kontakt 3.0.0 alongside third-party plug-ins. There were also reports of major problems using Kontakt 3.0.0 with Sonar. These included total failure to load the plug-in, and frequent crashes. I've now been using version 3.0.1 for two weeks in both Sonar 6.2.1 and 7.0.1, inside busy projects running numerous third-party plug-ins, with no obvious problems. There have been occasional crashes — twice whilst using the Wave Editor, and twice in Stand-alone mode — but these were not repeatable and the reasons remain unclear. The only other issue I found is that K3 fails to pick up Sonar's 'zero controllers when play stops', causing notes that were held by the sustain pedal to continue sounding when the sequencer is stopped.

giving precedence to 'valid' beats. I rarely had to delete or move any markers using the default 50ms setting — a far cry from the endless adjustments required in the previous version! Markers can, of course, be added and deleted manually, and locked to prevent further accidental movement. Now, let's examine the four tabbed views:

Sample Loop: up to eight different loops can be specified, offering the usual one-shot, forward, alternate and loop-in-release options. Note that alternate looping only works on samples using the 'sampler' engine mode, but not DFD, Beat Machine or Time Machine engines. The start and end points of loops (highlighted in orange) can be dragged to position, and will snap to the grid lines if Grid is active, or can be positioned freely if the Grid is disabled. Dragging sideways within the Loop area moves the entire loop. The Loop Edit button displays a highly magnified waveform view of the loop point, which can be finely adjusted (when the Grid is disabled) using the numerical Start and End values.

Sync/Slice: this determines how a sliced loop behaves when triggered via MIDI. Kontakt can take its tempo from either the host DAW or its own internal clock — either way, there are four options. The first, Time Machine, uses time-stretching, allowing the pitch of the entire loop to be altered without affecting tempo. The second, Beat Machine, triggers each slice as a discrete sample, producing cleaner audio across a wide tempo range. Thirdly, turning both Time Machine

The Sync/Slice mode allows you to determine how loops respond to MIDI triggering.

and Beat Machine off forces the loop to play back at its original tempo, instead of in sync with the DAW or Kontakt's own internal clock. The final option, Drag MIDI to Host, will be a familiar concept to Spectrasonics Stylus users: simply click on this button and drag directly to a MIDI sequencer track. Kontakt creates a new Key Zone for every slice, and drops a 'driver' MIDI part into the sequencer to play them at the project's tempo. Every nuance of timing (as determined by the slice markers) is preserved faithfully; this is so much easier and faster than the old K2 method of exporting MIDI files and re-importing them to the sequencer. If you wish to MIDI-slice more than one loop, selecting Auto Find Empty Keys ensures that each loop's newly generated Key Zones don't overlap each other, although the limit of 128 notes (and therefore Zones) per Key Map limits how many loops you can 'MIDIfy' this way in a single Kontakt instrument.

Zone Envelopes: new to Kontakt 3, this highly creative feature allows you to freely draw envelopes onto a waveform. Up to 16 envelopes can be assigned to control any modulatable parameter that exists in the loop's current Group. Just click on the parameter for which you want to create an envelope, then click Add Last Touched in the Zone Envelope area. Nodes are automatically placed at (and snap to) every Zone's Grid line, and can be moved, deleted or added to. Nodes can also be moved freely in time by disabling the Grid. Sub-nodes (red dots) enable the shaping of smooth curves between nodes, and the envelope itself can be looped using loop points completely independent from those in the actual sample loops. This is where the Grid's 'Fix' setting comes into its own; for example, any non-rhythmical sample, such as a pad sound, can be given 'rhythmic envelope' treatment by selecting, say, a eighth-note fixed grid. Any Zone Envelope applied to this Grid will now modulate the sample in sync with the host DAW. It's virtually identical in concept to Kontakt's tempo-sync'able Flex Envelope, but the beauty of using Zone Envelopes is that every sample that makes up a multisampled instrument can have its own independent modulations (or none) whilst still being part of the same Group. Similar effects could conceivably be achieved using multiple Groups and Flex Envelopes, but would be infinitely more complicated and time-consuming to set up.

Sample Editor: samples can be directly edited here, without the need for an external wave-editing program. After drag-selecting a region of the sample using the mouse, a number of processes can be applied to the selection: cut, copy, paste, duplicate, crop and delete are all to do with re-ordering a sample's content or otherwise altering the sample's length, whilst fade-in, fade-out, silence, reverse, normalise and DC removal are 'transform' processes. Kontakt applies these changes destructively, but to a backup copy (which Kontakt subsequently references) that's written to a special folder alongside the original sample on your hard drive, so the source sample remains unchanged.

Effects

Kontakt's already varied list of effects gains four new members in v3. First up is Rotary, a (you guessed it) rotary speaker simulation sounding not unlike the ones found in NI's B4 and B4 II instruments. Although it's not as editable as those, it offers control over acceleration and deceleration for both horn and bass rotor, along with horn/rotor balance, mic distance and wet/dry balance.

Next we have Skreamer. Although this is primarily intended for producing lead guitar sounds, NI would do well to include this effect in their B4 II organ. Warmer and smoother than Kontakt's Distortion effect, it sounds much closer to an overdriven Leslie preamp than B4 II's present Overdrive effect. It runs very hot, though, so you'll need to substantially reduce the gain at its input if you want subtle distortion!

Twang is a retro-oriented guitar-amp simulator capable of producing clean through to highly distorted tones. The controls are simple: input gain with bass, mid and treble EQ, a Bright switch to add that extra bite, and the oddly named 'Polyphonic' button which, when active, processes each side of a stereo signal separately.

Cabinet (what, not 'Kabinet'?) is a microphone/speaker cabinet simulator featuring 11 different cabinets, ranging from small combos to a Leslie 122 cabinet and 4x12 configurations, some with a choice of on or off-axis mics. Controls here are cabinet

The new Zone Envelopes mode lets you draw enevelopes to control any modulatable parameter.

The Sample Editor page allows you to edit samples without having to use an external waveform editor.

size, treble and bass EQ and 'Air', which adds convincing-sounding early-room reflections.

Conclusion

Existing Kontakt 2 users who are happy simply to load and play third-party libraries straight out of the box may not be seduced by the enhancements Kontakt 3 has to offer. However, anyone who views a sampler as a tool for creative sound design and manipulation surely won't fail to appreciate the ways in which this new version extends the sonic possibilities and speeds up work. The new Wave Editor is simply streets ahead of the old version, with pride of place going to its Zone Envelope tools, providing endless ways in which to warp, mangle and generally mess with your samples. The new effects are fine additions to the roster, and in fact, the full complement of Kontakt effects would make a very respectable suite of plug-ins by themselves.

An honourable mention must also go to the user manual; it's excellently written in a friendly, conversational style, and in perfect English (might we therefore infer that the author is not English?) The installation also includes a number of helpful and informative video tutorials.

To anyone yet to jump on the soft-sampler bandwagon, Kontakt 3 should certainly rank highly on your list of contenders. SOS

information

- £ £289.99 including VAT; upgrade from previous versions 129 Euros; upgrade from Kompakt, Impakt, Keyboard Collection or Kontakt Player 249 Euros.
- T Arbiter +44 (0)20 8207 7880.
- E mtsales@arbitergroup.com
- W www.arbiter.co.uk
- W www.native-instruments.com

NuGen Audio Plug-ins

Visualizer Analysis Tools & SEQ Linear Phase Equalisers For Windows

We test a selection of impressive and affordable Windows mastering tools from NuGen Audio.

Martin Walker

NuGen Audio is a new name to me, but over the last couple of years the company have been quietly building an enthusiastic following for their range of elegant and sophisticated VST plug-ins, building on the success of their first three products. Stereoizer enhances the width of both mono and stereo inputs, Stereoplacer is a parametric equaliser whose bands can be individually placed in the stereo field, and Monofilter can highlight and correct a wide range of bass-end and phase problems to give your mixes more definition and solidity.

The three products under review here are the SEQ1 seven-band parametric EQ plug-in and the SEQ2 'spline' EQ, which are both linear-phase designs, and an extremely comprehensive audio analyser tool called Visualizer that can help you highlight and solve many mix-related problems. The range is currently PC-only, but beta testers are currently being sought for Mac versions.

Visualizer provides a comprehensive set of functions with some unusual options, such as the Mid/Side spectrum analysis and stereo spectrogram, shown here above a pair of more traditional correlation and level meters.

Metering Matters

Audio metering and analysis plug-ins are not glamorous, but can contribute greatly to the quality of your mixes, and help pick up on problems such as subsonic rumblings and phase issues. Many audio editors and DAWs now bundle spectrum analyser plug-ins, but in my experience these tend to be fairly basic affairs that can be frustratingly inflexible. Visualizer is in a different league altogether, being incredibly versatile yet easy to use.

There are seven main analysis options in Visualizer's arsenal: Level Meter, Spectrum Analyser, Stereo Spectrum Analyser, Spectrogram, Stereo Spectrogram, Vectorscope and Correlation Meter. By highlighting the appropriate icons in the right-hand View Selector column, you can enable these in any combination to be viewed in the main display area. As you enable or disable each option, any others on screen re-size themselves to make room, so you always end up with the optimum window arrangement with the minimum of fuss. There's also a global switch between Compact and Large sizes for the entire plug-in. The 10 supplied presets give a good overview of possible arrangements. A set of Edit Control Panels allows you to tweak various aspects of the displays, such as digital or channel options, ballistic response and colour choices, so you can customise things to your preference and intended use.

I was really impressed with the click-and-drag scrolling and zooming functions offered by the four spectral displays, since this makes homing in on problem areas really easy. A Link button ensures that the frequency ranges of all visible spectral displays remain perfectly aligned with each other. You can also use the A Store and B Store buttons to capture displays for later comparison.

The stereo Level Meters can be displayed horizontally or vertically, and offer peak, RMS and K-Scale options, as well as margin and clip indicators. They cover most eventualities, although a M/S meter option would be useful. However, the versatility of the four spectrum displays more than make up for this lack, especially since the Spectrum Analyser provides a Mid/Side display option of its own, alongside more traditional ones such as left, right, maximum and average. The response options let you adjust the attack/release times of the display movement, while the Peak Hold has variable decay or infinite hold, which is useful when you want to build up

With seven bands of traditional knobs, plus extras like overall depth and M/S mode, the SEQ1 linear-phase plug-in offers a lot for the money.

a picture of what's going on during an entire track.

The Stereo Spectrum Analyser display is one I've not come across before, but within a few seconds I was really appreciating its stereo-differential views: you can quickly see how the stereo bias varies with frequency across the entire frequency range, with frequencies biased to the left appearing above the centre line, and those shifted to the right below it. A classic application for this meter would be spotting skewed bass sounds that would ruin a vinyl mastering session.

I've long enjoyed Spectrograms, which display a scrolling sonic fingerprint of your frequency spectrum over time, with the level of each frequency indicated by a spread of colours (typically 'hotter' levels are red, while quieter ones shade down to blue. Pure continuous tones appear as straight lines, which makes it easy to spot unwanted background whistles and hums, while drum hits produce wider bands of information. Because the scrolling display may well encompass a minute or more of your track, you can examine how compression is affecting the spectrum of your drum hits over time, for example, and spot masking problems when two or more instruments are fighting for the same part of the spectrum. There's also a Stereo Spectrogram that displays how your frequency spectrum is spread side to side in the stereo image over time, although I didn't find as many uses for this.

All four spectrum displays can be switched from high-resolution to octave, third-octave, sixth-octave and chromatic options, plus various psychoacoustically derived Bark/Mel scales, with narrower bands where the ear is more sensitive. I found the third-octave setting made it easy to spot audible problem areas without getting lost in hundreds of individual peaks and troughs.

The Vectorscope display illustrates the stereo width and phase characteristics of your mixes, and also offers some handy extras. There are both the more familiar Lissajous (oscilloscope) and Polar (like Waves' PAZ) view options, and there are manual and Auto Zoom functions for the Lissajous display, so even low-level mixes can provide full-scale displays.

The final display option is a Correlation meter with an optional scrolling History, which can help you spot phase problems. If, for instance, you spot a dip in the meter each time a particular drum is hit, its close mic is out of phase with the others.

Visualizer is completed by a Stats/Setup panel that provides a readout of various

Sound On Sound: NuGen Audio plug-ins

pros

- Visualizer offers virtually all the audio analysis options you'll ever need, including some you won't find elsewhere, within one colourful yet easy-to-use window.
- SEQ 1 and 2 provide more clarity than conventional EQs, but still offer a little character too, with high-resolution options that allow narrower bass-end EQ tweaks than some competitors.

cons

- Like all linear-phase EQs, SEQ 1 and SEQ2 have high latency when used at higher-resolution settings.

summary

All three of the NuGen Audio plug-ins reviewed here are professional and extremely versatile. They offer various useful features, such as Mid/Side options, that are not found in many competing products, yet they remain very reasonably priced.

Test Spec

- NuGen Audio Visualizer v1.3, SEQ1 & SEQ2 Master v1.1.
- Intel Conroe E6600 2.4GHz dual-core processor, Intel DP965LT motherboard with Intel P965 chip set running 1066MHz system bus, 2GB RAM, Windows XP with Service Pack 2.

clipping parameters, and lets you add various weighting (including A, B, C and D curves) to the spectral displays and change various FFT parameters. By right-clicking anywhere on a spectrum display you can also generate a sine test-tone at any frequency and level, which I found really useful in confirming what notes or harmonics caused particular spikes in my mixes.

I already have a small collection of meter and analyser plug-ins, but I can honestly say that within a day of receiving NuGen Audio's Visualizer I was using it almost exclusively. At $89 it's excellent value for money, given all that it offers.

Drawing in a suitable EQ curve using SEQ2 is easy and painless, while the integral spectrum analyser helps you spot problem areas that need treating.

SEQs Lives

Compared with conventional EQs, linear phase designs can achieve more transparency, preserve transients better and retain a sharper stereo image, but they do this at the expense of increased processing overheads and latency, so they are more suitable for mastering than mixing.

Nugen Audio currently offer two linear-phase EQs, each available in Standard or Master editions. SEQ1 provides five parametric bands, two with shelving options, plus high- and low-pass filters. SEQ2, by contrast, is a 'spline' equaliser. This means you can draw in your own freehand curves, although the high- and low-pass filters are also present. All four offer 64-bit internal processing and support sample rates up to 192kHz. Some parametric EQs can suffer from lopsided frequency curves at the top of their range, but NuGen's are claimed to avoid this.

Nice user interface touches include multi-stage undo/redo buttons and A/B memories so you can easily compare two different EQ curves. SEQ1 has a useful 'solo' light, so you can hear the effect of any single band in isolation, a neat Order button that shuffles the bands into frequency order if you happen to have dialled in your settings randomly, and a very useful Depth control that can make the entire response more extreme or more subtle without changing its overall shape. In both, a large graphical display offers the same versatile click-and-drag zooming and scrolling functions as Visualizer.

SEQ2 is simplicity itself to operate: you just draw in the desired curve, with fine changes greatly eased by the horizontal and vertical zoom functions. The number and distribution of control points is determined by the drop-down Banding selection, which offers the same wide range of options as Visualizer (chromatic, thirds, sixths, plus Bark/Mel alternatives). There's also a Depth control similar to that of SEQ1 that provides global 'compression/expansion' of the EQ curve, and a Contour setting that 'sharpens' or 'smooths' your curves. I found this particularly handy for ironing out the inevitable kinks in a hand-drawn response. In SEQ2, the graphical display can optionally act as a spectrum analyser, so you can immediately see the results of EQ changes in your audio output as you change the curve.

A natural limitation of most linear-phase EQ designs is that as you move down the spectrum, the maximum Q of your notches or peaks drops. If you need to dial up something 'tighter' to cure a hum or roll off the low end more sharply, you may need to increase the resolution. Unfortunately, each doubling in resolution doubles the latency of the entire EQ, so you need to choose wisely. NuGen's Standard models offer three of these 'quality' settings, while the Master Editions offer nine, enabling much tighter low-end tweaks if you need them, but at the expense of more sluggish audio feedback. Other Master Edition features include additional 64-bit dither options, a frequency range of 10Hz to 30kHz (Standard offers 20Hz to 20kHz), and a more accurate WYSIWYG graphical display.

The Master Editions also offer an extra Mid/Side stereo EQ mode, which lets you do clever things such as frequency-dependent stereo-width adjustment, or reduce the noise in an FM broadcast signal, which mostly occurs in the Side signal. You can EQ the left and right channels separately, or define different initial responses but then add further linked changes. Overall, the SEQ family is remarkably versatile, and I look forward to trying the eight-band SEQ3 with individual band placement in the stereo image, designed for restoration of elderly mono or poorly recorded stereo material.

Let's Phase The Music

All three linear-phase designs in my collection offer noticeably more clarity than conventional EQs, but there are still definite differences between them. To my ears, Waves' LinEQ has an uncanny knack of being able to alter spectral balance without adding any character of its own; if you dial in more bass, for instance, you get more bass, but the sound somehow doesn't seem to be any 'warmer'. This absolute neutrality is ideal for some mastering applications, but not universally enjoyed. PSP's Neon HR, by contrast, has its own definite character, a sort of analogue-esque 'sheen'. To my ears, SEQ1 and 2 fall somewhere between Neon HR and LinEQ, with a noticeable but very subtle character. I particularly liked the M/S options; Neon HR also offers these, but few other EQs do, in my experience. Like all linear-phase designs, the response to changes in settings can become sluggish once you increase the resolution for higher-Q bass tweaks, but I do think the Master Editions are worth the extra money for their various other options.

Overall, I'd say Nugen Audio's SEQ1 and SEQ2 have a very pleasing yet reasonably transparent sound, and are about the most versatile pair of EQ plug-ins I've ever used. This means it takes slightly longer to get your head around all the options, but once you do, the SEQ series can turn its hand to jobs that many other EQs couldn't attempt! SOS

information

£ Visualizer $89; SEQ1 and 2 $119 each; SEQ1 and 2 Master Editions $219 each. Various discounted bundles also available.

W www.nugenaudio.com

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Digidesign Pro Tools 7.4

Mike Thornton

Digital Audio Workstation Software For Mac & PC

Nowadays, we expect software to be able to change the tempo of recorded audio, but the latest version of Pro Tools integrates this functionality in a new and impressive fashion.

Often, the most exciting new features in Pro Tools appear first in the high-end HD version, and there's a frustrating wait before LE and M-Powered users get them — if they ever do. So it's a welcome change to find that nearly all the new features in Pro Tools 7.4, including the big one, are for the HD, LE and M-Powered platforms.

The biggest new feature in the 7.4 upgrade is something Digidesign call Elastic Audio. The idea behind Elastic Audio is that it allows you to work with audio in the Edit Window as if it was MIDI. All the things you take for granted with MIDI — moving and stretching notes and phrases, conforming loops to new tempos — can now be done with audio Regions.

Behind The Scenes

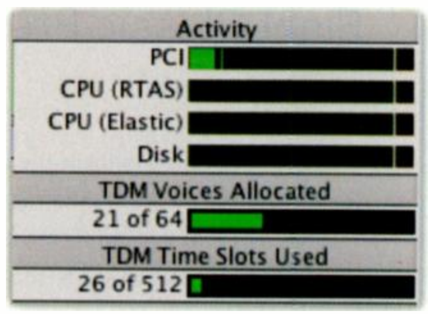
Elastic Audio uses a combination of transient detection, beat and tempo analysis, and time-stretching and pitch-shifting algorithms. These are not new technologies, but what is new is the way Digidesign have integrated Elastic Audio into the normal Pro Tools working environment to make using it as easy as possible. Major changes have been made 'under the hood' to ensure that Elastic Audio works as seamlessly as possible, but these are only apparent in a few places; one such is the System Usage window, which has a new CPU Usage indication for Elastic Audio.

Elastic Audio is accessed via a new track mode in the Edit Window, which offers one of four real-time algorithms that you select to best suit the type of audio content on that particular track. Three of these are time-stretch algorithms designed to keep pitch constant whilst varying speed: they are Polyphonic for multi-instrument loops or complex instruments like pianos and strings, Rhythmic for drums and anything percussive, and Monophonic for single-pitch instruments including vocals. The fourth algorithm is Varispeed, which produces a tape-machine-like varispeed effect where pitch and speed are changed together.

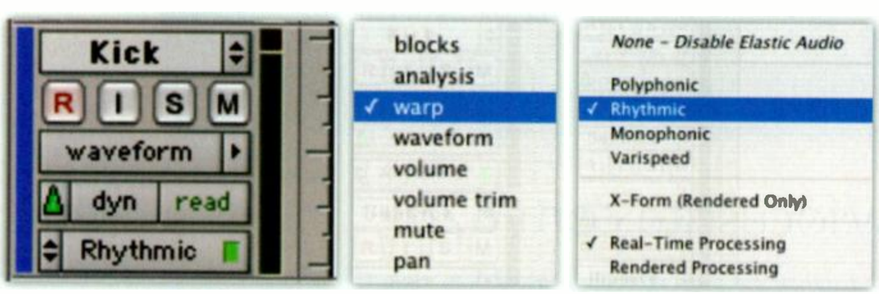
You can choose whether Pro Tools will handle all the time compression and expansion in real time, which is quicker but puts more load on the computer; or render, which means you have to wait for the files to be processed each time you change something, but has the advantage of not increasing the load on the computer. The rendered option gives you the ability to use Digidesign's X-Form algorithm instead of the four types listed above.

Going Into Analysis

When you change an audio track to Elastic Audio mode, Pro Tools automatically takes the audio off-line for a while to analyse it for transient events. These might be drum hits, the beginning of a sung note or the start of a chord played by a guitar. These detected events then serve as control points for what Digidesign call 'warping' the audio. Pro Tools can warp audio events either automatically, such as when conforming audio to the Session



The System Usage window displays Elastic Audio activity separately from RTAS plug-in overheads.



Elastic Audio introduces two new track modes, Analysis and Warp, and four analysis algorithms to choose from.

SOUND ON SOUND

Digidesign Pro Tools 7.4

pros
- Elastic Audio is a life-saver and very neatly integrated into the program.
- At last, a volume control in the Digibase Browser!

cons
- It would have made my life so much easier if it had been released a month earlier...

summary

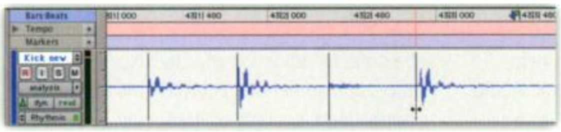
The technology behind Elastic Audio is familiar, but the way Digidesign have incorporated it into Pro Tools makes it something special.

tempo or quantising audio events to a grid, or manually, using the standard editing tools in the new Warp track View.

Once analysed, the file comes back on-line and you can then swap to Analysis View to see where Pro Tools has put Event Markers. You can correct Pro Tools' automatic transient detection by manually moving, editing or deleting Event Markers, in much the same way as you can in Beat Detective. When you're happy with the way the track is sliced up, you can have it automatically conformed to Session tempo or a groove (see below), or switch it to Warp View if you want to edit its timing manually.

In Warp View you can see three types of Markers: the Event Markers created by the analysis, Warp Markers and 'Tempo Event Warp' Markers. The last are not editable, and only appear on tick-based tracks, indicating where Elastic Audio has been used automatically to conform the audio to events on the Tempo Track. If no quantising has taken place (see below), there will initially be no Warp Markers in Warp View, but the Event Markers will be there. If you click to grab one of these and move it around, you will stretch the whole Region, with the start remaining fixed (unless you Alt/Option-click, in which case the end stays fixed).

Add a single Warp Marker somewhere in the Region before using the Grabber tool ▶

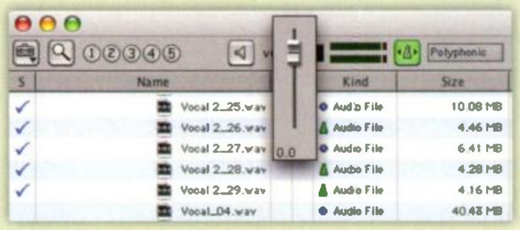


Elastic Audio analysis automatically detects transients and places Analysis Markers, which can be moved manually.

Digibase Browser Pumps Up The Volume

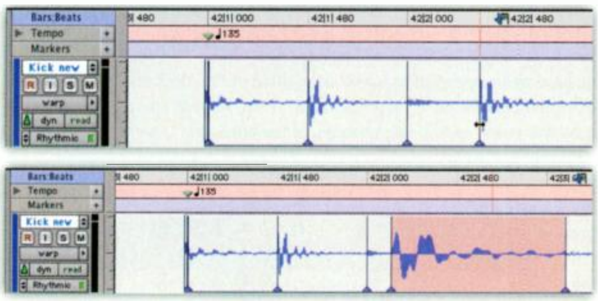
Various other aspects of Pro Tools have been improved to make integration of Elastic Audio as smooth as possible. Many of these enhancements are found in the Digibase Browser. At last, there is a volume control so you can turn down the volume when auditioning samples in the browsers. I, along with many others, am fed up of being blasted when auditioning loud samples and files, so this is very welcome.

A new Conform to Tempo option means that files being auditioned in the Browser will be analysed and then will play at Session tempo rather than their native tempo. You can also audition loops while the Session is playing, in which case Pro Tools will wait for the downbeat and play the samples in sync with the Session. The volume control enables you to 'mix' in the sample and hear it in context with the rest of the session.



You can now right-click files and folders in the Digibase Browser and analyse them in advance, to save you time when looking for files for sessions. Analysed files have a big tick against them and display a duration in bars and beats. They can be dragged straight into your Session Edit Window, whereupon Pro Tools will create new tracks with Elastic Audio already turned on and the files conformed to the Session tempo.

When you select one of a pair of Split Stereo files or a group of multi-channel files, the entire file now previews together unless you press the Shift key when you start previewing.



Switching to Warp Mode allows you to place Warp Markers. This example shows what Digidesign call a 'range warp', where audio between two fixed points is stretched by clicking and dragging with the Grabber tool.

to drag an Event Marker, and the results are similar, except that the Warp Marker rather than the start or end of the Region is treated as the fixed point. If you have two Warp Markers in your Region, clicking and grabbing affects only the audio between these markers; alternatively, you can make a selection that encompasses a number of Warp Markers, in which case the two outermost ones within the selection will be the fixed points. If you move a Warp Marker so far that the amount of time compression and expansion is deemed to be excessive, Pro Tools warns you by turning that section red.

Using Elastic Audio

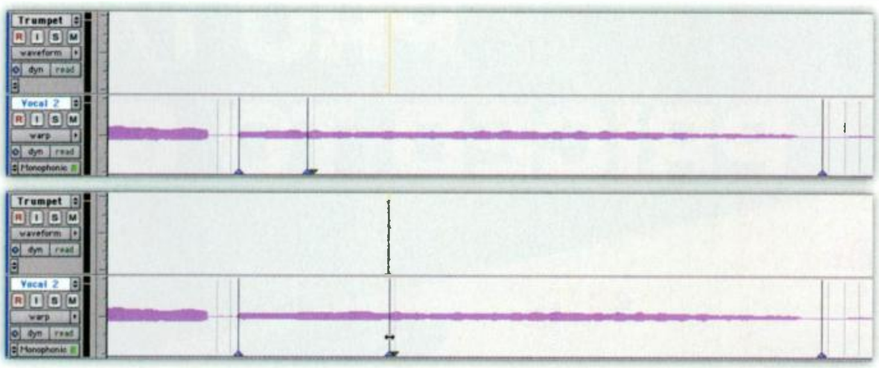
The screenshot below shows a real-world example where I used Elastic Audio to bring vocals into time. The sync mark (with the green arrow) shows where the singer placed a syllable change, while the yellow line in the track above is the locator point, which shows where the instruments changed their chord. The two should be in the same place, but aren't. So I switched the vocal track to Elastic Audio, went into Warp view, placed Warp Markers at the start and end of the word to act as anchors, and then placed another at the syllable change point. Then I simply dragged that point to the right to line up with to the locator point, and the singer was in time with the instruments — simple. All of this was in a Session with no click and no tempo grid to work to, and all tracks were sample-based rather than tick-based.

If you need to change the tempo significantly — especially if a significant slowing down is involved — you may find the sound has been compromised, but Elastic Audio has another feature to help counteract this. Clicking on the Elastic Audio Mode button opens up the Elastic Audio plug-in window for that track, and you can adjust the Decay control to bring the sound quality back into line. Sometimes you can also improve the results by switching to a different algorithm: for instance, if you are stretching kick-drum sounds to get a longer, richer thud, it is worth trying Polyphonic mode rather than the usual Rhythmic, and increasing the Decay value.

Prior to Elastic Audio, producing believable varispeed effects has been difficult. It could be done by adjusting the clock speed of Pro Tools and somehow recording the result; then came along a couple of Audiosuite plug-ins, including Wave Mechanics Speed, Waves Soundshifter and Serato's Pitch n' Time, which could render good varispeed effects, but none was especially cheap. Now, however, Pro Tools can do it out of the box: simply make sure your tracks are set to use the Varispeed algorithm and draw your slowdowns and speed-ups on the Tempo track.

Conformism

You can use Elastic Audio on sample-based tracks, but if you want to automatically conform a Region to the Session tempo, you'll need to switch your track to the tick-based timebase. You can then right-click the Region and select Conform to Tempo, and Pro Tools instantly moves all the events so they line up with the tempo of the track. It seems so much



Here, Warp Markers are being used to move a syllable change within a word so that it coincides with a musical event.

Playback Engine

Another area of the program that has changed in v7.4 is the Playback Engine, and some of these changes are reflected in the Playback Engine window. The DAE Playback Buffer size is now displayed in milliseconds as well as the usual 'levels', and there is now a Cache Size drop-down menu for determining the amount of memory DAE allocates to pre-buffer audio for playback and looping when using Elastic Audio.

If Digi's Structure sampler is installed, the Playback Engine window now allows you to set a Plug-in Streaming Buffer Size to determine the amount of memory DAE allocates for sample playback, and has an option to 'Optimise for Streaming Content', for improved playback of sampler plug-in samples from your audio drives.

Other Improvements

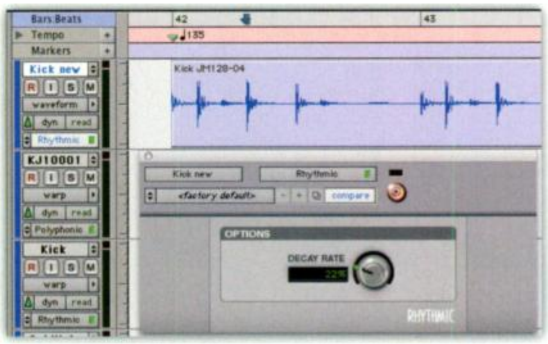
Elastic Audio is the big story in Pro Tools 7.4, but as ever, there are a number of less eye-catching yet useful enhancements:

- **REX file handling has been improved, and you can now import REX files as Region Groups.**
- **It is now possible to change the waveform vertical zoom level on individual tracks by Ctrl (Windows: Start)-dragging up or down with the Zoom tool. If you zoom an individual audio track, or group of tracks, the waveform display on that track remains offset when you zoom all tracks, unless you reset the waveform height.**
- **If there are no tracks in the Session, and you import a tick-based audio file, Pro Tools will give you the option of importing the tempo from the file.**
- **In Pro Tools HD 7.4, stereo surround panners are now linked by default, so when you create a stereo track that is routed to a multi-channel output, the left and right pan controls are linked, as well as the Front Inverse, which inverts Left and Right pan control linkage across front and the Rear Inverse, which inverts Left and Right pan control linkage across rear. The Front/Rear Inverse pan control remains unlinked by default.**
- **When you open a Session with missing files, the Missing Files dialogue now includes the option to Regenerate Missing Rendered Files Without Searching. This is great when you know that the missing files are only fades and so on that can easily be regenerated.**
- **Pro Tools now supports receiving MIDI over Rewire, allowing you to route the MIDI output from a client application such as Reason to the input of a Pro Tools MIDI or Instrument track. This is especially useful for recording changes to parameters in a Rewire client application onto a Pro Tools MIDI track.**
- **Pro Tools 7.4 LE now has on/off footswitch support for 003, 003 Rack, 002, 002 Rack and M Box 2 Pro interfaces, and the ability to punch in and out with a footswitch has been added to the M Box 2 Pro.**

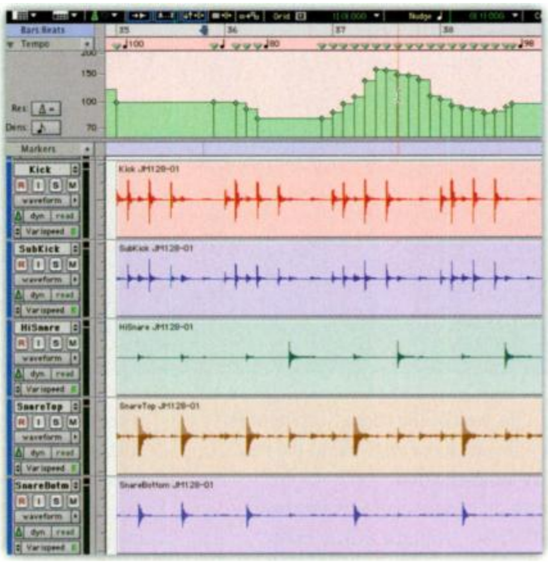
New Video Features

There have been a significant number of Avid video-related improvements in all the versions of Pro Tools 7.4. Pro Tools systems with Avid Mojo now have access to all the video resolutions previously supported by Pro Tools with Avid Mojo SDI or AVoption V10. Support has been added to enable the co-installation of Avid Xpress Pro 57.x with all versions of Pro Tools 7.4 on the same computer. Digidesign have added a number of previously HD-only features to the LE version when used in conjunction with Avid Mojo SDI hardware and 003 or M Box 2 interfaces.

For HD users, Avid Media Station PT 2.7 Software with the Video Satellite Option eliminates time-consuming video exports and removes all video-processing burdens from the Pro Tools system by synchronising playback with Media Station PT on a separate, dedicated computer.



The Elastic Audio plug-in window enables you to fine-tune the time-stretching.



At last it's possible to create realistic tape-style varispeed in Pro Tools, by switching tracks to Varispeed mode and editing the tempo map.

more 'intelligent' than Beat Detective, which may well be down to the improved transient detection. You can also quantise audio to a groove, rather than the strict tempo of the Session. This is accessed from the Event Operations option in the Events menu; if you choose the Quantise option in the sub-menu, you can select a groove from the drop-down menu. Make sure you have Elastic Audio Events selected at the top — if you have Audio Regions selected, Pro Tools will move the complete Region rather than the elements within it.

Does Elastic Audio replace Beat Detective, then? In many ways I think it probably does, although Digidesign have added an improved analysis option to Beat Detective, which suggests they think there is life in it yet. The key difference is that Beat Detective 'cuts up' files and moves small regions around to conform events, whereas Elastic Audio uses time compression and expansion. I had expected that Beat Detective would be the only way to be sure you could guarantee absolute phase coherence when moving around groups of events across multiple tracks. However, I understand that one of the reasons why version 7.4 took longer to release than Digidesign had hoped was that they wanted to be sure that if you used Elastic Audio across multiple tracks within an edit group, it would all remain phase-coherent. Certainly, judging from the content I have played with personally and heard in demonstration, Elastic Audio is more than good enough at re-conforming multitrack events without creating undesirable side-effects. I am sure I will not be the first to be saying "I wish I'd had Elastic Audio for the last job I did. It took me ages to sort out and with Elastic Audio it would have been done in no time at all!" SOS

information

- **£** Upgrade from v7.3 £52.88 (HD) or £29.38 (LE/M-Powered); upgrade from earlier versions £129.25 (HD) or £47 (LE/M-Powered). Prices include VAT.
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Studio SOS

This month, we perform some audio alchemy on a home studio room that also has to double for other purposes.

Ray Stroud's studio desperately needed acoustic treatment to help him achieve the guitar recordings he wanted, but he was reluctant to cover the walls of the multi-purpose room with acoustic foam. Unobtrusive Advanced Acoustics panels provided an answer.

Paul White & Hugh Robjohns

Ray Stroud is a very talented singer and guitarist who has more than his share of gigs under his belt. Since taking early retirement, he has started to do a bit more gigging, as well as some recording. However, he hasn't been satisfied with the quality of his recordings, so he asked us to cast an eye (and a couple of ears) over his studio, which is set up in a spare room within his Worcestershire home and based around a PC running Cubase.

Studio Stroud

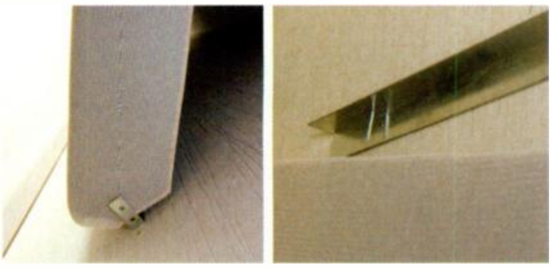
The room is a rather compact 9 x 10 feet, and when we arrived Ray had his studio gear set up along the longest wall, which meant that his chair was close to the centre of the room when mixing. This should usually be avoided, as the bass end tends to disappear in the exact centre of a small room, especially if it is nearly square, as in this case. Some kitchen worksurface was set up all the way along the 'studio side' wall of the room, with another piece on the adjacent wall taking the surface round as far as the door.

The studio setup itself was reasonably straightforward, with the PC's in-built soundcard being used to get audio into and out of Cubase. The monitoring comprised a domestic hi-fi amp driving a fairly small pair of Mordaunt-Short hi-fi speakers, and these were set up on the fabric rings normally used to cover Ray's tablas when not in use. By way of microphones, Ray has a Rode NT2 and a pair of cheap TakStar capacitor 'stick' cardioid models that he bought on eBay, as well as an LD Systems 1011 back-electret condenser, a Shure SM57 and a pair of Sennheiser E345 dynamics, for live work.

A cursory listen to some commercial material played back over the speakers confirmed that the bass was unpredictable, the imaging very poorly-defined, and the sound generally coloured — which is not surprising, given that the wallpapered walls had no acoustic treatment. Ray had bought a couple of used office dividers in the hope that these would provide a suitably dead corner for him to record in, but they turned out to comprise a fairly thin layer of fabric over a hard sub-surface, which meant that they weren't very effective as absorbers. We decided to get rid of these, apart from a couple of narrow panels that Ray had joined together to create a notice board. We decided to move this in front of the radiator at the back of the room, as it would be marginally less reflective than the radiator itself and it looked a bit more inviting. However, some serious acoustic treatment would be needed to get the room up to the standard where Ray could make good recordings in it.

Advanced Treatment

Fortunately, Advanced Acoustics had offered us a room treatment kit to use on a Studio SOS project, and because their panels blend in visually rather better than traditional sculpted foam, it meant Ray could still use the room as a guest bedroom without it looking too



Ray suggested mounting the Advanced Acoustics panels using some thin aluminium stripping, which he had lying around, in order to minimise the cosmetic damage to the walls — something which worked as well in practice as it did in theory.

studio-like. We received four 2 x 4 feet wall sound trap panels, two 2 x 4 feet corner traps and a further narrow corner trap, again four feet long.

The wall traps contain solid slabs of two-inch thick, high-grade acoustic foam with an open-weave fabric covering, and a backboard made from 6mm MDF to keep the panel in shape (and also, in some mounting situations, to augment the low-frequency absorption). The foam and covering meet the relevant British and European safety regulations, so flammability shouldn't be a major concern. The corner sound trap panel, which is of a similar construction, but with four-inch foam and angled edges, has an NRC (noise reduction coefficient) of 0.93, providing an absorption coefficient of 0.76 at 50Hz when mounted diagonally across a corner. While these are essentially conventional foam absorbers, the fact that they aren't sculpted means that their average thickness is equal to their full two (wall) or four (corner) inches, whereas shaped foams have an effective thickness somewhat less than their maximum thickness. As with all similar products, the thicker the foam, the more effective the low-frequency absorption.

Before fitting the panels, we persuaded Ray to change the orientation of the studio, such that he would be working down the length of the room with his back facing the window — where we hoped the window blinds would help break up reflections. This would place his mixing position a little distance forward of the centre of the room and hopefully avoid the worst of the bass suck-out experienced in the centre. He'd end up sitting slightly to the left of centre, due to the positioning of the door but we felt this would be an acceptable compromise.

After some thought, we settled on using two wall panels on each side wall, with one large corner panel in one rear corner. We found the space between the wall and window at the other side was too small, and even the narrow panel was too tight a fit, so we used that in the front left corner. The remaining large corner trap we decided to hang horizontally, so that we could use it as a wall panel in front of the mixing position.

Though the panels come with velcro strip for attaching to walls, we didn't want to ruin Ray's walls by gluing anything to them, so after a bit of head-scratching, Ray came up with the simple idea of fixing lightweight aluminium angle strip to the walls above and below the panels, then drilling holes through these, through which could be inserted short masonry nails (taken from some spare phone cable clips) to pierce the foam just in front of the MDF backboard. Ray already had some aluminium angle left over from a kitchen refit, so we tested his idea, and when it worked he went out and bought a little more to finish the job. He also bought some small angle brackets to hold up the corner traps. Ray then lined up the tops of all the vertically-mounted panels and the end result looked very tidy.

Sonically, the difference between the before and after results was dramatic. Before treatment, the room had a very pronounced coloration, but afterwards it sounded really well controlled, without being oppressively dead-sounding. Checking out material over the Mordaunt-Short speakers showed the imaging to be much more pronounced, and the bass was now quite tight (albeit limited from such small speakers). In fact, it was only after fitting the acoustic treatment that Ray realised he'd need some better monitors (the amp was also a touch underpowered) and before we returned to take the final photos, he'd bought a set of Alesis Monitor 1 Mk2 active monitors, which proved ideal for that size of room, given Ray's limited budget. These were set up on a pair of Auralex MoPads to isolate them from the worktop, and to ensure they pointed directly at Ray's head when he was seated in his normal monitoring position.

Since a computer upgrade, Ray had been unable to get his Tascam US122B audio interface to work, so he'd been using his computer's built-in soundcard, with less than ideal results. Installing the latest Tascam drivers did the trick.

We also installed Ray's Tascam US122 USB audio interface — something he'd had around for a while but never been able to get to work properly since having his computer fixed. Reinstalling the latest driver, then going through the setup routine in the manual ▶

This picture shows the studio after we'd sorted the main problems. We'd rotated everything 90 degrees so that the speakers fired lengthways down the room. This, combined with the acoustic panels, Auralex MoPads, and the diffusion provided by the blinds on the window to the rear of the room, made a huge difference to the sound. Now we're ready for the boutique acoustic guitars...

reinstated the device, which was then selected as the I/O for Cubase 4. I got Ray to do this while I stood on a pile of dry newspaper, wearing a welding visor and rubber boots filled with Vaseline — you know how I feel about messing with PCs!

With the Tascam interface up and running, Ray now had a means of controlling the active monitor level and setting up latency-free monitoring. The interface has a pair of mic preamps built in, along with phantom power for his capacitor mics, and it also provides a decent headphone feed. Ray's Roland XP30 keyboard was connected via the Tascam's MIDI interface and his system was up and running.

Acoustic Guitar

Before finishing up, we made some test recordings, partly to test the system, but also because Ray was unsure how he could get the best sound out of his collection of rather nice boutique acoustic guitars. Now we had an Advanced Acoustics panel across one of the rear corners, it seemed a good idea to try recording with Ray sitting in front of this. We also put an SE Reflexion Filter behind the mic to further reduce room reflections. Ray knew the Reflexion filter could help with vocals but he hadn't considered its use for guitar. One reason I like to use them for acoustic guitar is that it gives me chance to experiment with using omni-pattern microphones, rather than cardioid: I find that omnis always seem to give a more natural sound. Without some type of acoustic screening, omnis can pick up too much room tone, but with a Reflexion filter behind them they are far easier to control.

Previously, Ray had been advised by an engineer friend to use an X/Y pair of cardioid mics to capture the guitar in stereo, but although this can work well it also introduces problems that can sometimes outweigh the benefits. Firstly, my own experience is that omnis give a better sound and, of course, you can't use these in an X/Y coincident pair. Secondly, with any sort of stereo array that's set up fairly close to the instrument, any movements made by the player cause the stereo image to shift. In recent years I've always recorded acoustic guitars in mono and then used a stereo ambience reverb to add width where I think it is necessary.

For the first test we set up Ray's Rode NT2 mic and made some guitar recordings with it switched to cardioid mode, then recorded the same piece in omni mode. As usual, we moved the mic around to find the sweet spot, and because Ray was sitting over a piece of exposed wooden floor, we got some useful reflections from that source to add life to the sound. Both recordings sounded fine, but Ray and I felt that the omni version had a more natural, open feel to it. Next, we tried the same thing using one of his budget TakStar cardioid pencil mics, which had cost him little more than the price of a set of strings! This turned in a very respectable performance too, given its modest leanings, but it exhibited the somewhat congested character of a low-cost cardioid capsule, and it also had a bit too much presence boost — which may work very well over a drum kit, but made the guitar sound a hint on the grainy side. With care, we were able to get very usable results out of it, but comparing it with the Rode NT2 soon highlighted its shortcomings.

Paul explains to Ray how to achieve a good recorded sound from a single omni-pattern mic.

Finally, we tried my own Rode NT55, with the omni capsule fitted, and if anything that gave the best result of all — though not sufficiently better than the NT2 to persuade Ray he needed to buy another mic. It was an interesting exercise, though, and I think it gave Ray the confidence to experiment in order to find out what works best for each guitar. During the course of our mic positioning experiments, we also pointed out that placing the mic below, in front of and slightly to one side of the guitar, then aiming it up in the general direction of the bridge, can not only give you a great sound but, if you're recording vocals at the same time, also puts useful distance between the two mics. This can be important, not only in reducing spill, but also in minimising the phase errors that occur when the same sound source is picked up by different mics at different distances.

Ray's Reaction

Ray: "The room had been decorated and equipped to function as an office, studio, and part-time bedroom, and suffered from the inevitable compromises that entailed. I'd spent quite a lot on recording software and hardware but balked at the thought of covering the walls in grey foam — I didn't want to have a 'cell' or 'bunker' feel to my creative environment and, let's face it, it is hard to get as excited about foam as a new piece of 'kit'. However, the lack of treatment negated confidence in the results I got, and my backlog of compositions remained unrecorded.

"I'm delighted with the Advanced Acoustics panels and with the Auralex MoPad speaker platforms. The aural impact on the room is amazing, and the soft, non-oppressive appearance of the panels blends tastefully with the decor. Fixing them with three-quarter-inch aluminium angle means the panels can be removed in seconds, and if I move house I can take the angle off the walls, leaving holes smaller than picture nails. All round, it is a great result: my thanks to Advanced Acoustics and to *Sound On Sound*.

Ray Stroud, at home in his revamped studio.

On Reflection

Having the Advanced Acoustics panels really helped us address the main acoustic issues in Ray's room, without the end result looking oppressive, and the difference they made was dramatic. Moving the monitoring to fire down the length of the room also helped resolve the unreliable bass-end issue.

Once again, we'd managed to show that a simple recording system, using modestly priced but well-chosen mics, can produce excellent results, as long as you're prepared to pay enough attention to the acoustics of the recording space. SOS

Steinberg Groove Agent 3

Virtual Drummer Instrument For Mac & PC

Martin Walker

It's been a long time coming, but has the latest version of Steinberg's Groove Agent been worth waiting for?

Since it was first released, back in 2003, Steinberg's Groove Agent has come to the aid of many musicians who either don't want to or can't program drums, yet prefer not to rely on static sample-based loops. Groove Agent is essentially a virtual drummer VST Instrument that can play drum styles crossing many decades of history, each with a range of complexities, from laid back to busy, but with the flexibility to add accents and fills on demand.

By the time I reviewed Groove Agent 2 in *SOS* May 2005, it offered many more rhythm style options (81 instead of v1's 54) including grunge, punk and trip-hop, plus nine new kits, as well as up to eight stereo outputs available for more refined mixing options. Unfortunately, its interface provided few opportunities for yet more growth, and since the original programmer had moved on to pastures new, Steinberg decided to rewrite Groove Agent 3 from the ground up, with an ambitious new list of options. These include the Special Agent, with 15 drum styles of sample-based recordings; the Percussion Agent, with eight groups of live-recorded percussion; a new Dual Mode, offering a choice of any two of the three available modules; and other additions such as 27 new Groove Agent styles, three new acoustic drum kits and an enhanced effects section for each of the 12 outputs.

Unfortunately, Steinberg ran into difficulties during GA3 development that prevented it being released for a further year, but they managed to smooth a lot of ruffled feathers by starting a development timeline and associated blog (www.steinberg.net/1346_1.html) to keep everyone informed about progress. Despite this, many musicians gave up waiting and investigated other alternatives, such as EZ Drummer and Jamstix, before GA3 was finally released in August 2007. Has it been worth the wait? Let's see.

Installation & Compatibility

As with Groove Agent 2, the supplied GA3 activation code requires a Steinberg dongle (not supplied); you can either use an existing one (I already had one for Cubase) or buy one. While it isn't mentioned anywhere in the manual or on the Steinberg web site, GA2 users should also accept the default installation destination, which places GA3 in a fresh folder alongside any existing installation, so you can use them simultaneously. If you already have songs that use GA2 you'll need to leave it installed anyway, since you can't load GA2 presets into GA3 (I do wish developers would include such information in their manuals!)

Another issue with a VST Instrument that pushes the boundaries like GA3 does is compatibility. Groove Agent provides an elegant 'Live to host' feature, to output in real time whatever MIDI data you generate from its changing patterns (plus fills and accents) onto a separate MIDI track, so you can further edit it. However, not all hosts can handle MIDI output from a plug-in, so there's a 'Record to file' option for applications that can't use the 'Live to host' feature.

Unfortunately, many musicians who bought previous versions of Groove Agent experienced difficulties in using it with hosts other than Cubase SX3 and Cubase 4, so some at least must have been hoping that the GA3 delays would ensure wider-ranging compatibility in this latest ▶

Steinberg Groove Agent 3 £169

pros

- Provides a huge amount of virtual drumming and percussive talent in a single package!
- Dual Mode is a clever and versatile way to combine multiple Agents.
- You can now create your own kits using the sample import function.

cons

- Much longer delay when switching between styles than with GA2.
- Groove Agent 'Live to host' option may not work with some host applications.
- Special and Percussion Agent performances can't be exported to a MIDI track and further edited.
- I feel that some of the Agent combinations are overwhelming for many types of music.

summary

Steinberg's Groove Agent 3 is an ambitious upgrade containing more styles than you can shake a stick at, plus an impressive new 'live' drummer and percussion grooves, but it does have a few teething troubles, and there are question marks over compatibility with some host applications.

version. Sadly, this doesn't seem to be the case (even Steinberg's own Cubase SX1 and SX2 haven't yet been officially tested), so I do hope that Steinberg soon publish a list of GA3-compatible host applications on their web site, along with advice on how best to set up GA3 to work with them.

Back To The Classics

I began by revisiting the original Groove Agent display, now known as Classic Mode. First impressions were good — although the timeline of previous versions (with different drumming styles laid out chronologically from 1950 to 2000 and beyond) is gone. This was already getting unwieldy in GA2, given the huge number of styles, so in GA3 Steinberg have grouped the even greater number of styles into 15 main categories: Jazz, Latin, Moods, Blues, Country, Pop, Dance Floor, Rock, World, Music Academy, Heavy, Hip Hop, Electronic, Modern Pop and Club. This makes it far easier to find what you're looking for.

Most of the interface layout remains the same, although GA3 has been given a 'walnut dashboard' make-over, and there's an enhanced auto-fill control that, in addition to offering an optional fill every time you change the drum pattern, now offers a very handy option to play automatic fills every second, fourth, eighth, 12th or 16th bar. The 'under the hood' controls for tweaking individual kit sounds are also largely the same, but there's a handy additional Speed control with half, 1x and 2x settings for those times when (for instance) you've played in your song at 50bpm but accidentally left the sequencer tempo set to 100bpm.

If you want to create your own kit sounds, you can now import user samples into Groove Agent. Global compressor and graphic EQ options are also now available.

There are lots of new styles — so many, in fact, that the manual devotes a full 24 pages to describing them all in detail, but given that we don't have that kind of space available, here are some highlights. From the Moods collection there's 'Free Form' jazz improvisation and the evocative 'Old Squeaky', with snare rolls that sound like creaky doors, while the more ethnic feels are catered for by the skin drums accompanying 'Mandela' and the agogo bells of 'Senegal'.

Other sections include nods to specific tracks and artists, including 'Wonderland' (Stevie Wonder), 'Jillie Bean' (Michael Jackson), 'Madish' (Madonna-inspired) and 'League' (Human League), while more genres, such as Acid Jazz, Jungle and Irish

Live Drumming Feel With Special Agent

Special Agent features 15 extra styles covering such genres as rock, jazz and Latin, with tempos ranging from 60 to 120bpm, and each with 25 variations and fills, just like Groove Agent, but using live recordings edited into slices rather than pre-programmed MIDI triggered samples.

This means you get the feel of a live drummer, but with lots more flexibility than you would using standard drum loops. However, you have to be more careful to stick within the recommended tempo range for each style, because using an SA style faster than intended can sound a little unnatural, while going slower can result in silences between beat slices (interesting for weird, sliced effects, though!).

You also have to be careful when changing from a variation involving cymbals to one without, because as the new one starts, any cymbals currently sounding will suddenly be chopped off in their prime, instead of continuing to ring on as they would in the real world. However, you can largely mask this anomaly by enabling the fills, as these are layered over the patterns, so when you play a fill any cymbals used continue dying away while the normal pattern continues.

As a control freak already in love with the way I could modify each individual drum sound within Groove Agent kits, I was initially a little sceptical about Special Agent, but the more I discovered workarounds for its little foibles and ways to push it in unexpected ways, the more I grew to like it, and the more I appreciated its undoubtedly natural feel. This module has been cleverly programmed, and since both dry and ambient samples are used, SA offers a 'pre-delay' control that lets you move the ambience samples forwards and backwards, which also offers some strange and chaotic effects.

There are a few bugs — I occasionally heard double (flanged) kick drums on the first beat of a fill, and SA only plays hi-hats if you start playback before choosing a new style, but Steinberg have already acknowledged these and 10 other bugs that are destined for cure with a GA 3.0.1 update expected by November 2007 (http://forum.cubase.net/phpbb2/viewtopic.php?t=79928). Ultimately, I suspect that if you enjoy the 15 available 'live' styles you'll enjoy Special Agent, and I found it a useful if perhaps not a must-have addition to the team.

Special Agent offers the 'live' feel of a real drummer, sliced up and served at varying tempo and complexity.

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ECHO

Rock, are covered. There are also some more unusual time signatures, such as 5/4 Fusion and 7/8 Funk, as well as several styles based on paradiddles, including a classic Heavy Metal version for those double bass-drum figures. Each style includes 25 variations and fills, plus half-tempo versions.

There are three new acoustic kits and a range of new electronic ones, such as the Linn LM1 (heavily featured on the LM Ballad style). To my ears, many of the previous kits seem to have had the relative levels of their component drums and cymbals slightly rebalanced, as well as having their pan position and ambience levels tweaked, giving a more punchy and lively sound, while the new kits feature alternating hits for kick drum, snare, hi-hat, tom and cymbals, making rolls rather more realistic and faster ones less like machine guns.

Even better for those who know exactly what drum sounds they want, you can now import samples. There's a dedicated Import & FX page where you can load up to 27 different pairs of sounds, allocate each of them volume, pan and balance values, and then save them as User kit presets.

There are also two new global effects on this page that are applied globally to all agents — an easy-to-use compressor and a nine-band graphic EQ, plus lots of presets. I'm sure most musicians will already have plenty of plug-ins that can perform both these duties, but they are nevertheless a handy addition and a very quick way to radically alter kit sounds when required. On the other hand, if you want to add plug-in effects to individual sounds, you can now increase the number of GA3 stereo outputs to 10 from the much more comprehensive setup page.

The only significant down side of the new Classic engine that I noticed is that each time you change style the new kit is loaded from disk, which can take quite a few seconds in some cases. For me, the instant changeover of GA2 meant that I often accidentally discovered styles that worked well in a song, but sometimes found it difficult to locate a particular style in its cluttered timeline: with GA3 you get a much better-organised timeline, which makes finding a particular style a lot easier, but the loading times do disturb your flow when searching for the best style to fit a song. Definitely a case of swings and roundabouts.

Any two of the three available Agents can be combined in Dual Mode, with global transport controls provided along the central strip. Notice the simplified Groove Agent display in the top half of this screenshot.

The Dual In The Crown

For many people, the most significant new features are going to be the two new Agents

Percussion Agent

Percussion Agent provides you with the sound of up to eight simultaneous percussion players, and has a very similar screen layout to the Classic Agent sound edit controls. Each of the eight horizontal strips holds your choice of the 39 available instrumental grooves, covering various tambourines, shaker, cabasa, triangle, guiro, woodblock, cowbell, hand claps, congas, bongos and cajon.

Each groove offers five variations with increasing complexity, and you can fine-tune their sound with shuffle, tune, ambience, pan and volume controls. However, far more creative possibilities are offered by the Groove Offset controls, which let you shift the start time of individual instruments in eighth-note increments to create polyrhythms. Changing the groove offsets in real time creates all sorts of shifting rhythms, and the sounds are all well recorded. However, there are no accents or fills available (although sometimes these happened accidentally with the random buttons activated), and it would be handy to have an option to automatically mute the percussion parts when fills are triggered in the other Agents. I was also a little disappointed at the lack of more Classical percussion, such as timpani and orchestral drums, plus Eastern percussion such as Japanese Taiko drums, Chinese gongs, and so on, and since the Sample Import options only apply to Classic Agent there's no way to incorporate your own new percussion sounds.

One thing that sets both the Special and Percussion Agents apart from the original Groove Agent is that, although you can still automate your moves, you lose the option to capture the note data to a MIDI track for further tweaking; instead, their contributions must be captured like a regular VST Instrument, using audio export. This is perhaps understandable for the loop-based Special Agent, but not for the often complex rhythms generated by Percussion Agent, and I found it frustrating not to be able to further refine them. Nevertheless, Percussion Agent is undoubtedly a very welcome new feature, especially when used in conjunction with the other Agents.

With up to eight players and some very effective Groove Offset features for complex polyrhythms, Percussion Agent is an excellent addition to the team.

that accompany the original Groove Agent, and the Dual Mode that allows any two of the three to run simultaneously. To enter this mode you click on the Dual Mode button that appears at the top left of Classic Mode, and a new screen appears. This has a central horizontal control strip where you can load any of the three Agents in either the upper or lower half of the display, and have separate or joint control over the Stop, Run, Accent, Fill and half-tempo feel, plus overall control of Level and Balance between the two Agents loaded.

In Dual Mode, Groove Agent appears with the lower half of its Classic display largely intact, but with a much smaller horizontal timeline, and you must make style and kit choices from drop-down boxes. I adapted within a couple of minutes and was soon enjoying the option of having two Groove Agents playing simultaneously with different styles and kits. Some lateral thinking generated lots of other interesting combinations, since it's possible to mix individual drum patterns from two styles, using, for example, the kick and drum parts from one style with the hi-hat and cymbal parts from the other.

However, the more obvious Dual Mode combinations involve Percussion Agent and Special Agent (see boxes for more details on each of these). It's easy to over-indulge in Dual Mode, and unless you want the massed sounds of a carnival parade it's better to remove some components from each active Agent. One obvious coupling is the Groove and Percussion Agents, although I found this combination often sounded more effective if I automated the various solo instrument buttons to drop elements in and out during the course of my song.

Combining Special Agent and Percussion Agent is more challenging for 'traditional-sounding' drum parts, since you can't drop individual drums out of the former, making it very easy to create rather an over-the-top feel. Using Groove and Special Agents together is an ideal pairing for anyone who wants to explore the experience of having twin drummers in their virtual band, but another, more subtle, approach might be to employ the Dual Mode balance control to morph from one to the other, and use them individually in different parts of the same song.

Final Thoughts

Overall, there's nothing else on the market that's quite like Groove Agent 3, but despite the long wait for the new version, it does still have some rough edges, with a list of initial bugs that's already been worked on, a few unexpected limitations, and some doubts over compatibility with well-known DAW hosts such as Logic and Sonar. Let's hope that these issues are soon resolved, and that more information becomes available to prospective users of applications other than Cubase SX3, Cubase 4 and Nuendo 3.

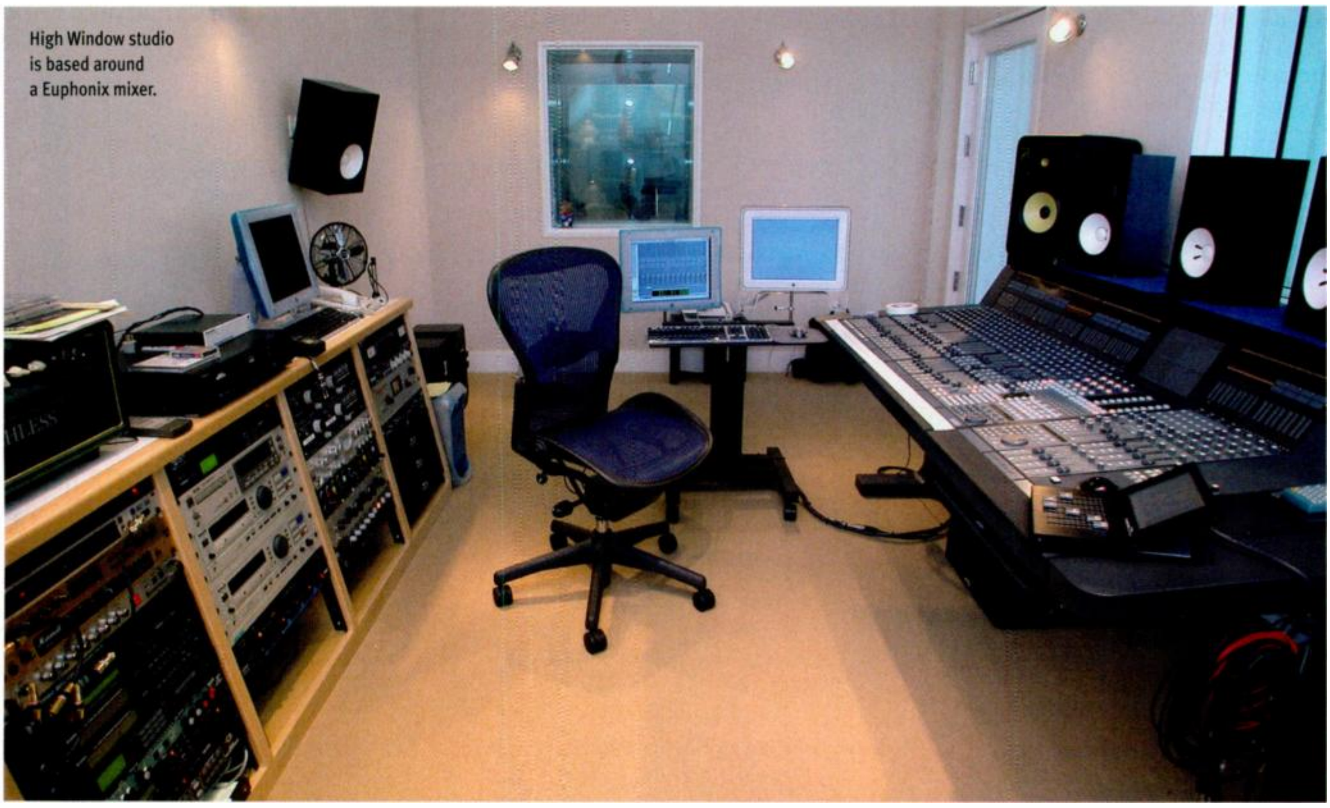
Nevertheless, GA3's developers are to be congratulated on an ambitious product, and, with an upgrade price of just £69 for existing GA2 users, I suspect few will resist the urge to add percussion and a versatile 'live' recorded drummer, in the shape of the new Agents. The full £169 asking price also seems very reasonable, considering all that's on offer. SOS

information

- **£** Groove Agent 3 £169.99; upgrade from GA2 to GA3 £69.99. Prices include VAT.
- **T** Arbiter UK +44 (0)20 8207 7880
- **E** mtsales@arbitergroup.com
- **W** www.arbiter.co.uk
- **W** www.steinberg.net

interview glen ballard

THE ART OF RECORD PRODUCTION



High Window studio is based around a Euphonix mixer.

your imagination can conjure up. It can be accomplished with you and a computer.

"I started off writing charts, but I don't do it as much anymore because it's much easier to put it into Sibelius, or have the Logic program do it. The hours and weeks and months that I used to spend writing arrangements, while it is very important, is also very time-consuming. And now all of that can be accomplished in a fraction of the time. In addition, you can basically get any sound you want and use it, when previously it might take days to create one sound."

Know Genius When You Hear It

However, the most important skill in the producer's armoury is the ability to recognise a brilliant song. "I think a producer's job involves many aspects. The most important thing that a producer can bring is judgment of material. A great song or a great piece of music is very hard to mess up. Your skills as a producer will only be elevated by the level of the material you are working with. If you are working with mediocre material, it takes a lot of work to make it palatable. Most of us have spent time on material that probably didn't deserve that much time.

"For me, it starts with that and knowing where to spend your time. It all goes back to developing a critical faculty in terms of understanding what really works. While it's intangible, it's something that you can develop an instinct about. I start with material. That is the first thing. If you are going to produce a record, what are the songs? If you don't have any songs, that is your first order of business. If you don't have anything that makes you excited, then you must find it.

"The most important thing you can do after you acquire a basic skill set is to really develop your taste in music. To me, that is the future for all of us in the somewhat fragmented marketplace. Your taste will be more important than anything. Being able to define that in a way that is distinctive is very important now. Work on the kind of music that you like to make and you will probably find an audience. If you are a creative musician, you are in better shape now than ever before."

Be Committed

The music business is in the midst of enormous upheaval at present. How does this affect the role of the producer? "In recent years, there has been a lot of doom and gloom in the business. It has a lot to do with everybody aiming for the Top Ten, because that was where you could make a living. That was where all the glory was, and all the money. But that has all changed. There has been a devaluation of the currency of music as we have known it, its remunerative value. It's a shocking development for a lot of producers, like me, but there is an up side of this decentralisation, and the fact that you don't get paid as much as you used to, for a variety of reasons.

Studios In Boxes

Despite the boom in home studios and the availability of every instrument under the sun in software, Glen Ballard is a firm believer that professional recording studios still have a place in music. "For the last 40 years, all of us have been recording sounds," he explains. "We set up microphones and worried about the sounds, and spent an inordinate amount of time making music sound good. Fast forward to 2007 and it seems like all that work has been done and is essentially available, whether it's in sample form, or it's someone else's record that you put a beat over, or whatever — the artifacts of recording history have been made available to people in their box at home.

"But the continuation of creating new music and new sounds requires well-built and well-maintained recording studios, like the Record Plant here in LA and many others, where people actually have great knowledge about how to record real instruments. I can't tell you how many people I know, younger producers or engineers, who don't really know how to record. There is a great art and a great science involved with recording and understanding how to record something well.

"I've always thought that Los Angeles was the place most conducive to recording because so many people out here are not only interested in recording, but are inventing gear. In LA's San Fernando Valley about half of the garages have people in them making new gear. It's great that this place still exists, because the things that you can do here can't be done anywhere else."

Learn With Ballard

Berkleemusic.com, the on-line arm of the Berklee College of Music, recently created the Glen Ballard Scholarship as part of the Celebrity Online Scholarship program, which is designed to reward and assist outstanding students studying in certificate programs at Berkleemusic. "Berklee has always had the reputation of not only being an excellent music school, but also one which is really in touch with contemporary music and all other kinds of music, too," he remarks. "There has been a fundamental difference in the way that Berklee approaches and differentiates itself from a lot of other great schools. It's the practical application of what you are learning, and the sense that it isn't just theoretical."

"At this point in history, we have the accumulated recordings of everything that has happened up to now, and it is available on some level to us to use as components of what we are calling 'new music'. This is a great moment for musicians; it is also a scary moment. Now we have a diverse universe of music, but the audience is still looking for the music they love. They may not find it on the radio, but it's out there.

"We, as musicians should create the music, first of all, that we like. We have to be confident enough to make the music that should be made. We have the ability to do that now, at a fraction of the cost. I think that is what's happening everywhere, whether it's in jazz, classical, different forms of pop music, world music, or other forms. There is an audience for all of it. Because of the new changes in technology, people can get to it from all over the world. It's a great time to be a musician, a producer, an engineer.

"You have an opportunity to make the music that you really want to make, or which you really have an affinity for. You may not get paid as much for it, but I tell you, you will have more fun doing it. We must have realistic expectations of what music can bring us. It's got to be more than just the money. You can still make money in this business, but it's a defining moment for those who are into music because they love it, and for those who just want to make money. You will make money if you love it enough to put your heart and soul into it. But if you don't, and you just want the royalty cheque — and maybe it's not coming this year — maybe you shouldn't be in music.

"You start off in music not getting paid. You do it because you love it. And somewhere along the way you get paid, but it's now a time when we are reorganising how we can be compensated for the music we create. It may call for some lowered expectations in terms of what monetary riches and fame music can bring you, but instead call for more of a commitment to making great music. And, with that in mind, I believe you can make a living from it."

Prepare To Succeed

In closing, Ballard looks again at the importance of properly preparing for a career in music. "If you go into a profession with a strong skill set you have a much greater opportunity to be successful at it. While you are acquiring the skill set, you should really figure out what music you really love and what music you really have an affinity for. When you do that, you have found your place in music.

"You have a much greater chance of success if you are working with music that is natural to you. It shouldn't be a stretch, and it's not about trying to recreate something that is already there. It's really about developing what you already have. Whatever it is that turns you on, that is what is most important. Don't ignore that. That is your key." SOS

Software Pianos
A Buyer's Guide

If recording a real piano isn't for you, there's no shortage of very convincing software alternatives.

Mike Senior

Nothing can quite match the sound of a real grand, and elsewhere in this issue we explore methods of recording one, but in practice a lot of home studio owners find sample sets or virtual instruments a more practical option. There are lots of contenders to choose from, each with its own slant favouring certain applications. This buyer's guide aims to make the task of choosing a suitable instrument quicker and easier, as well as providing a Jargon Buster (see page 182) to get you up to speed with the necessary piano-related technical terms.

4Front Technologies True Pianos

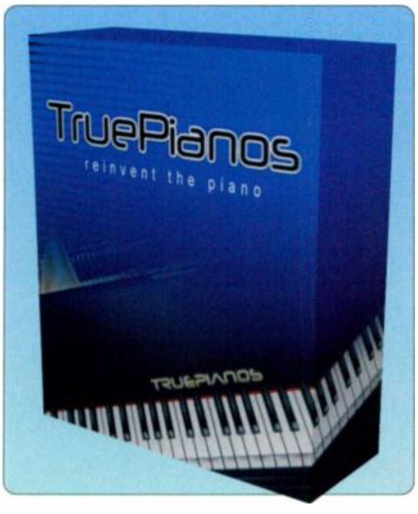
True Pianos offers three basic piano sounds of unspecified provenance ('Diamond', 'Emerald' and 'Sapphire'), with excellent polyphony and modelled sympathetic resonance, despite the slim file size (75MB for Diamond). Recommended CPU specs are a 2.5GHz single-core or 1.5GHz multi-core PC, or a Power PC G5 or Intel multi-core Mac, with neither needing more than 256MB RAM. The interface is simple, with each basic piano sound ('Module') offering a number of sonic variations ('Presets') along with straightforward keyboard dynamics, tuning reference and note release controls. The output is free of room ambience, so you can add processing according to your needs. Included in the price is a year's worth of downloadable Module updates: the Sapphire Module was new in October 2007, and further new Modules are promised. Audio examples and 40-day demo can be downloaded from the company's web site (see 'Pricing' box).

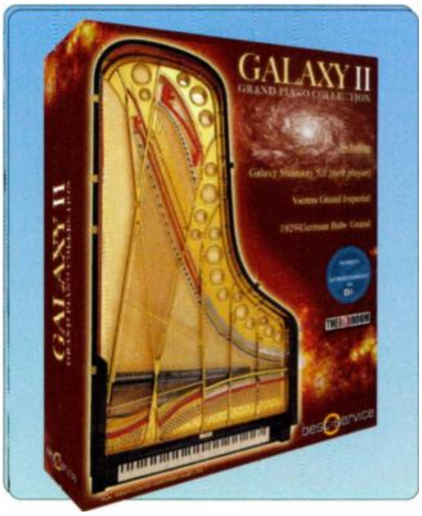
Best Service Galaxy II

The successor to Galaxy Steinway 5.1, Galaxy II offers three sampled grand pianos: the original Steinway Model D 270, recorded in Belgium's Galaxy Studios, a nine-foot 96-note Bösendorfer Imperial 290, recorded in Germany's Hansahaus Studios; and a 1929 Blüthner Model 150 baby grand. Each was copiously multisampled, and the Steinway, as before, was miked up for surround as well as stereo, resulting in an overall library of about 29GB Special features are a convolution reverb, a programmable pad-synth layer, and a Warp section of four sound-design effects. A full review of the library is in progress as I write this, so keep your eyes peeled. In the interim, here are Dave Stewart's thoughts on the Steinway sound from the Galaxy 5.1 review in *SOS* March 2006: "The piano sounds great... Dynamic transitions remain smooth right across its seven-octave range; the beautifully transparent high notes are responsive to the most delicate of touches and the bass notes can deliver weight and power... Classic-sounding without being too 'steely' or overbearing, the instrument is subtle and expressive enough for jazz and classical music, and strong and bright enough to cut it in a pop mix."

Blüthner Digital Model One

This library is based on samples of a Blüthner Model One recorded by Dan Dean & Ernest Cholakis on the scoring stage at Lucasfilm's Skywalker Sound studios. Each note of the piano was recorded at 12 velocity levels, with a phenomenally low noise floor. The sound was captured very dry, in order to eliminate, as far as possible, the resonant character of





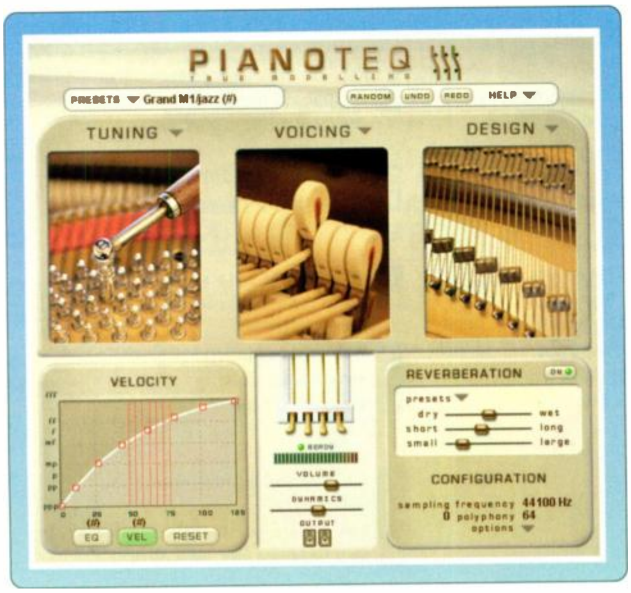
the piano and the recording room. These elements are instead recreated using convolution processing, which effectively allows you to change the construction of both piano and room using the large built-in library of impulse responses. All that convolution requires at least a 2.8GHz Pentium 4/Athlon PC or a 1.8GHz Apple G4/G5. Other unusual features include support for a variable sustain pedal (for a more natural playing response) and a facility to retune just intonation on the fly to suit whichever key you're playing in. In *SOS* November 2007, Mark Wherry was impressed: "When I started to play the Digital Model One I almost forgot that I was sitting in front of an electronic instrument... If Dan and Ernest provided just their original source samples in the Digital Model One, it would already have been a great instrument, but the flexibility and tonal possibilities you can get from using the [*convolution processing*] are tremendous and really add another dimension to the sound."

Modartt Pianoteq v2.2

This instrument uses physical modelling rather than the more common sample-based approach — hence the software's tiny 15MB file size. The two main built-in piano sounds aren't based on a particular brand of piano, although since Pianoteq's launch the company have used the same technology to capture the sounds of seven museum-piece keyboard instruments, and these add-ons are free downloads for registered users. The power of the physical modelling approach is evident in the characteristics that can be adjusted in Pianoteq — changes in hammer hardness, amount of sympathetic resonance and piano size, for example. Other unusual features include a variable sustain pedal; an intriguing added 'staccato sustain' pedal, which provides sustain-pedal resonance without affecting note lengths; microtonal and variable stretch-tuning; and a unison width control, which adjusts the tuning accuracy of the individual strings for higher-register multi-string notes. Delivering 256-note polyphony at up to 192kHz, with built-in reverb and EQ processing, Pianoteq's CPU load is understandably pretty high (Modartt recommend a 3GHz Intel Pentium 4 CPU or any of the new dual-core processors to get the best out of it), but a modest 128MB RAM is consumed. In *SOS* January 2007, Dave Stewart gave a resounding thumbs-up: "I compared Pianoteq to one of the best of the recent crop of sampled grands, and the difference was pretty dramatic. Both pianos sounded fine on simple pop/rock styles... But when it came to expressive improvised music, Pianoteq was much more playable — its notes sounded more connected than those of the sampled instrument, giving fast runs and phrases something of the silvery cohesion of a real piano." Audio examples and a 45-day demo are available at the Pianoteq web site.

Native Instruments Akoustik Piano

Four pianos are provided by Native Instruments' Akoustic Piano: a Steinway Model D, a Bechstein Model D 280, a Bösendorfer 290 Imperial, and (for a very different character) a Steingräber 130 vintage upright. Each note was recorded at 10 velocity levels, and there are samples for note releases, sympathetic resonance, and key/pedal noises, all of which contribute to the 15GB install size. The pianos respond to sustain, sostenuto, and Una Corda pedals, and you can specify whether the virtual piano lid is open, half-open, or closed. A wide range of tuning systems is available, including such exotic-sounding schemes as Pythagorean, meantone, Werckmeister, Kirnberger, Vallotti and quarter-tone. Three-band EQ and a convolution reverb help you fit the pianos into your mix.



Sampletekk

Sampletekk's own-brand libraries use extensive 24-bit multisampling, with pedal-up, pedal-down and release samples at numerous velocity levels. Available for recent versions of Gigastudio, Kontakt, Halion and EXS24, the computer requirements vary depending on polyphony and the complexity of the specific piano. Sampletekk recommend a fast hard drive and a 3GHz Pentium 4 PC with 2GB RAM (or similar Mac) to run their largest piano without compromises. The libraries offer 'lite' patches with fewer velocity levels, and some titles are also available as 16-bit versions, which reduce the strain on your computer. Cheaper, cut-down versions of many of the libraries are also available.

Sampletekk also sell the libraries of their former competitor Post Musical Instruments (PMI). These take a similar multisampling approach, but incorporate programming refinements designed to increase realism. For example, both sustain-pedal-up and sustain-pedal-down samples play at once — the sustain pedal simply crossfades between them, allowing more convincing re-pedalling effects. Body-resonance and pedal-noise samples are layered under the note samples in some cases, for a more natural sound. Impulse responses for the piano's string/body resonances and original room ambience are often also provided, for use in recent Kontakt and Gigastudio versions. Sampletekk's grand piano libraries are listed here, but their range extends to upright pianos and historic keyboards, which are detailed on the web site. The site also features audio demos.

SOFTWARE PIANOS

- **PMI Yamaha C7:** A Yamaha C7 grand piano, sampled at 16 velocity levels with two miking rigs, one close and one ambient, which can be mixed together in real time. In *SOS* October 2003, Nick Magnus had some concerns about the mid-range group of pedal-down samples, which sounded 'choked' to his ears, and overall he felt that the C7 was "best-suited to an accompaniment role... this piano is most likely to appeal to those people doing rock and pop productions."

- **SG88 MkII:** A Malmsjö Concert Grand sampled at eight velocity levels, with release samples added from Sampletekk's White Sister library (which uses the same piano). This library is only available at 16-bit resolution, but at only 570MB it is easily downloadable.

- **The Big One:** Another Yamaha C7 grand, the largest in Sampletekk's range (and, apparently, 'The Largest Sampled Piano In The World'), sampled at 31 velocity levels. A cut-down (but still weighty) version, The Small One, is also available.

- **White Grand:** A nine-foot Malmsjö Concert Grand sampled at 16 velocity levels. In *SOS* May 2004, Dave Stewart found that, while the "bold, assertive" sound would work for pop/rock styles, "it's hard to imagine jazz and classical pianists warming to its slightly middly tone and rather percussive attack." Sampletekk's response was to release White Sister, a more ambient version, which comes in at the same price. Together, the two form the WG II bundle, which costs $208. There's also a $49 version with half the velocity levels.

Steinberg The Grand 2

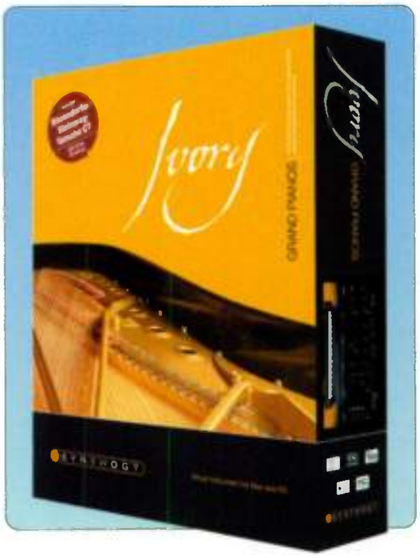
Two different pianos (rumoured to be a Kawai and a Steinway) were recorded dry, using an anechoic chamber, allowing you to shape the sound later via the built-in surround-capable room-ambience processing. All three pedals are supported (with mid-note re-pedalling possible), and hammer-release, key-action, damper and pedal noises can be mixed in. A number of sonic variations on each of the two pianos is available to suit different musical styles, and there are both equal temperament and stretch tuning options. You'll need at least a 1.6GHz Intel/Athlon PC or a 2GHz G5 or 1.5GHz Intel Core Solo Mac with 1GB RAM to run this instrument, despite several built-in facilities to trade off polyphony and sonic subtleties for more efficient performance. Look out for a full *Sound On Sound* review, coming soon. Meanwhile, here's what Martin Walker had to say about its predecessor in *SOS* March 2002: "I was certainly impressed by the quality of The Grand, and with the breadth of sounds on offer... The Grand is an instrument to be reckoned with."

Piano Jargon Buster

- **Audience Perspective:** A stereo presentation of the piano's pitch range to match the view from the audience, with high strings more on the left and low strings extending more to the right.
- **Damper Noise:** The soft felt-on-wire sound of the string damper moving away from and back onto the string. This is most noticeable when the sustain pedal operates all the dampers together.
- **Equal Temperament:** The most widely-used tuning system for pianos, which involves all intervals being very slightly out of tune, but in a manner that is barely noticeable and, more importantly, consistent in all musical keys. This tuning system is also best when combining the piano with other MIDI instruments, most of which also use equal temperament.
- **Hammer Release Noise:** The subtle mechanical noises the hammer mechanism makes just after a note is played, as it bounces off the string and comes to rest.
- **Just Intonation:** A tuning method associated with Renaissance and Baroque music. It provides a more in-tune sound when playing in closely-related keys, but quickly sounds out of tune as you move into more remote keys.
- **Key Noise:** the subtle mechanical noises emanating from the key and hammer mechanism as the key is pressed and released.
- **Pedal Noise:** The mechanical noise associated with the action of any of the piano's pedals.
- **Physical Modelling:** A way of recreating the sound of an instrument by mathematically modelling the characteristics of each small element of it, and then allowing those modelled elements to interact. The advantage of this approach is that it gives you much more scope to shape the final timbre, but the model needs to be powerful to faithfully reproduce the complexities of natural piano sounds.
- **Player's Perspective:** A stereo presentation of the piano's pitch range to match the view from the playing position, with low strings on the left and high strings on the right.
- **Polyphony:** The number of notes that can be played back simultaneously. Although 10 notes of polyphony might initially seem enough for us humans, the sustain pedal dramatically increases the number of notes that can sound at once and repeatedly-pressed keys will trigger overlapping samples. Reproducing this many notes (and other samples of mechanical noises and sympathetic resonances) can demand a lot of the host computer, and if it can't cope then you'll probably find some previously played notes disappearing to make way for new ones — an effect often called 'note stealing'.
- **Room Ambience:** The sound of the acoustic space in which the piano was recorded. The type and amount of ambience is recorded with the samples by some developers, sometimes with a choice of sample sets using different mic positions. Others record the piano as dry as possible so a chosen room ambience can be added artificially.
- **Sostenuto:** A third pedal on some grand pianos which sustains only those notes which are being held when it is pressed — all other notes can still be played staccato.
- **Sympathetic Resonance:** When any piano note is played, the piano's casing and all the other strings also vibrate to some extent, especially if the sustain pedal is down. A lot of developers record separate pedal-up and pedal-down samples to reflect this, but some capture the strings as dry as possible and then recreate the resonance effects synthetically.
- **Una Corda:** Literally 'one string', this is the term given to the left-hand pedal on a grand piano, which shifts the playing mechanism to play only one of the strings for upper-register notes, giving a more muted tonality.
- **Convolution:** A type of digital processing which can extremely accurately reproduce the response of any resonant system, such as a piano's soundboard or a concert hall's acoustic. A specially recorded file (called an impulse response) needs to be loaded into the convolution processor for each resonant system to be recreated. Because convolution is quite processor intensive it can place heavy demands on your computer's CPU.
- **Release Samples:** A sample that typically incorporates some residual room ambience decay or a little damper noise, and is triggered at the ends of notes for extra realism.
- **Stretch Tuning:** A tuning system based on equal temperament, but with all the intervals slightly stretched so that octave intervals in particular sound more in tune. This gives a slightly purer tone for solo pieces, but can sound out of tune when layered with other MIDI instruments using equal temperament.
- **Sustain Pedal:** The rightmost pedal on a grand piano, which lifts all the dampers from the strings, irrespective of which keys you play.

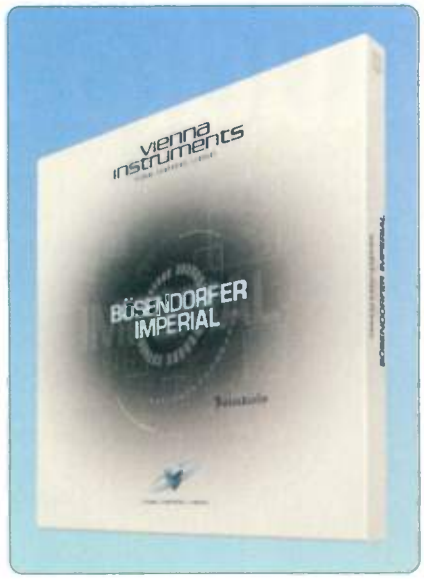
Synthogy Ivory

Three different grand pianos (a nine-foot Steinway Model D, a Bösendorfer 290 Imperial, and a Yamaha C7) are sampled at 10 velocity levels for this 40GB title, and a further 19GB expansion pack adds a 10-foot Italian Grand. Separate Sustain and Una Corda pedal samples help increase realism, as do the DSP-based sympathetic string-resonance modelling and sampled key noises. There's a range of timbral controls, including chorus, EQ, and reverb processing, as well as a pad synth layer you can mix in. Both equal temperament and stretched tunings are available. In *SOS* March 2005, Paul Wiffen said "It's great when the first sound you open is the one you have been looking for for years — I must have played the Bösendorfer for at least 20 minutes before I even thought about calling up another preset... the Steinway and Yamaha are also both excellent."



Vienna Symphonic Library Bösendorfer Imperial

A nine-foot Bösendorfer 290 Imperial grand was copiously multisampled at seven velocities to produce a whopping 54GB of samples packed up into a 37GB virtual instrument. Sustain resonance, key noise and multi-velocity release samples help the realism, and VSL have also taken the unusual step of recording special repetition samples to take account of the difference in sound when an already vibrating string is hit again. Two miking positions offer an ambient audience perspective and a drier player's perspective. In SOS May 2007, Dave Stewart remarked: "This piano is characterised by an open, clean and stately sound and a very clear attack which is discernible at all dynamics — a great asset... for making sure the instrument stays audible in a full orchestral score." SOS



Audio & Design DMA2

Digital Mic Preamp

Hugh Robjohns

This unassuming device has a price that makes it look like a bargain alongside competitors of similar quality.

Audio & Design Reading (A&DR) are one of those small British specialist manufacturing companies who have been around seemingly for ever, but always in a fairly low-key way. The company started in the mid-1970s, making highly regarded solid-state limiters and compressors — such as the Compex — for the professional recording studio market and broadcasters. Amongst their eclectic product range were also several very specialised Ambisonic surround sound products, including UHJ transcoders, pan and rotate units and more.

With the introduction of digital recording in the early 1980s, the company started making specialist digital interface products as well as modifying semi-pro equipment for professional use. In fact, I bought an A&DR-modified Sony PCM701 digital processor in 1986, which was my first venture into the world of digital audio.

These days, in addition to the manufacture of a small range of bespoke products, the company has evolved to become more of a consultancy and systems installer for the Broadcast IT market, with recent projects including the installation of a multi-terabyte archive system for the BBC World Service. The manufacturing side mainly comprises a range of broadcast transmission limiters, grade 1 video and AES digital clock reference generators, sample-rate converters, digital audio distribution and management processors, compact desktop digital mixers and high-quality digital mic-amps. It is the last of these that forms the subject of this review.

Technicalities

The DMA 2 is a two-channel microphone preamplifier with built in 24-bit, 96kHz A-D conversion. It is designed to offer all the essential facilities in a convenient package without any unnecessary frills. The sound is clean and transparent, rather than coloured or tailored — so this is a product aimed at cost-conscious professionals who want precision, reliability and accuracy.

Housed in a simple, 1U rackmounting box, the DMA2' has front-panel controls that are all push-buttons with associated LED status lights. The case itself measures only 153mm deep and weighs just 1.65kg, and the unit is powered by an internal universal mains supply, which accepts mains voltages between 90 and 250V AC.

The rear panel features a pair of female XLR sockets for the microphone inputs, plus another for an external AES11 reference clock input. The digital output is provided on a male XLR connector. There are no analogue outputs and no word clock input or output. An RS232 serial port is provided for full remote control and there is an option for an RS485 interface if required. The mains on-off switch is adjacent to the fused IEC mains inlet.

Internally, the unit is built to high standards, with a main circuit board covering most of the floor of the box and a subsidiary one behind the front panel to carry the switches and LEDs. The surprisingly large switched-mode power supply is in a separate case at the right-hand side.

The circuitry is a mixture of conventional and surface-mount components, with the analogue stages being based around very low-noise SSM2019 preamp chips, supplemented with a couple of discrete transistors at the front end. The gain control and audio switching is performed using a combination of CMOS analogue switches and a digital volume control chip, and the A-D converter is a Crystal device.

The technical specifications are impressive, with an effective input noise of -124dB (with a 200Ω source) at the maximum gain setting of 70dB. Distortion measures 0.003 percent at 30dB of gain, and the digital output has less than 0.5ns of jitter when running from the internal crystal clocks.

Operation

The front-panel controls are very logical and straightforward. Each of the two mic-preamp sections features five push-buttons and an array of LEDs. The preamp gain is set using a pair of up/down buttons and an associated ladder of LEDs shows the nominal input level in 5dB steps from -70 to +10dB.

The three other buttons switch in a high-pass filter (18dB per octave from 100Hz), phantom power and a polarity reversal — and each has a different colour of status LED, so that the configuration is obvious even from a distance.

SOUND ON SOUND

Audio & Design DMA2 £582

pros
- Simple user interface.
- Remote control options.
- Accurate resettability.
- 70dB gain available.
- Clean, accurate sound quality.
- Very attractive pricing (in the UK).

cons
- No word clock in or out.
- No analogue outputs.
- Phantom power voltage slightly low.

summary

A very cost-effective and high-quality dual-channel preamp that delivers a natural, uncoloured sound, with integral 24/96 A-D conversion and simple, no-frills interface.

There is no conventional level metering, just a trio of LEDs set to illuminate at -18, -10 and 0dBFS, corresponding to the EBU recommendations for nominal level, maximum permitted level and clipping. Clearly, as this unit provides only a digital output, the intention is to use the recorder's own metering for a more detailed analysis of signal levels.

Located between the two preamp sections are two more buttons. The first switches the sample rate and the second configures the output channel assignment. The sample-rate switching display is slightly unusual. With four LEDs available, most equipment would provide lights for the 44.1 and 48kHz modes, plus a double-rate light and an external input light. However, the DMA2 has lights for external input, 44.1, 48 and 96kHz — because those were the only rates provided in the original model. Subsequently, an 88.2kHz option was added, and this is now indicated by illuminating the 48 and 96 LEDs together. Admittedly, the front-panel graphics make this arrangement perfectly clear, but it is a little unusual.

The Channel Assignment button switches the output formatting. Normally, preamp 1 feeds the left channel of the digital output, and preamp 2 the right — normal stereo. However, it is also possible to switch the unit to provide mono (both inputs summed and routed to both outputs), or to route either preamp's signal to both outputs (with the other input muted). These facilities betray a broadcast requirement and will be rarely used by most, although the ability to generate a true mono signal is useful when lining up a stereo pair, especially given the polarity-reversal facility in the preamps.

The final front-panel facility is a slide switch to disable the front-panel buttons completely, as a security function. To avoid accidental operation, if a button hasn't been pressed for a while, the first press of any button causes its corresponding LED to flash, but the function is only activated if the button is pressed a second time. If pressed within the 'timeout window', other buttons can be activatec and their functions will change immediately — but once the timeout expires the next press will trigger the warning flash again. If a second push doesn't happen within a couple of seconds, the flashing will stop and the function rema ns in its original state. The only exception to this arrangement is the gain-control switching, which responds instantly at all times.

Every operational function can be controlled remotely via the RS232 port using a simple ASCII-based interface. RS232 can be extended over about 15 metres, but for more distant remote control an optional RS485 board can be installed, allowing connection over more than 100 metres.

In Use

The DMA2 is very simple to set up and use. The nice thing about the gain switching is that it is very accurate and totally repeatable. The double button-press business sounds a lot more complicated than it really is — and in practice it doesn't seem to get in the way at all.

In terms of performance the DMA2 really impressed me. It is a quiet, clean and very neutral-sounding preamp, with a huge amount of gain available. I'd compare the sound of the DMA2 with the likes of the Audient ASP008, or the DACS MicAmp — and it costs significantly less than either (although it has only a quarter of the channel count of the Audient, of course), while also providing the A-D conversion.

The gain markings seem to correspond accurately to the input level needed to achieve full modulation — which means that the highest input it will tolerate is +10dBu. This is fine for a microphone source, obviously — even a condenser in front of something very loud indeed — but it won't accept a full professional-level line input, so you can't use the DMA2 as a line-level A-D converter without padding the line source down first — although, having said that, you will probably get away with a semi-pro line source working at a nominal -10dBV.

Everything works as expected: the high-pass filter is nicely judged, and useful in reducing low-frequency acoustic or mechanical rumbles; the pola ity inversion does what it says on the box; and although the phantom power measured a little low at 45V, it is still within spec... just. The digital output is stable and the clock rates are accurate. The only disappointment here is the absence of word clock in and out, which makes it difficult to synchronise multiple DMA2s: you'd need to get a master clock unit that provided duplicate AES11 reference outputs, such as the Drawmer M-Clock, for example, and clock each DMA2 via their AES11 reference inputs.

In terms of sonic quality, this unit impressed me — but it does not produce an 'impressive sound.' By this I mean that it is clean, transparent and natural-sounding. It does not endow its outputs with the 'larger than life' character that so many aspiring high-end preamps do. Personally, that suits me down to the ground — I'd rather have an accurate, clean recording that I can tweak and shape later in post-production than a coloured master recording... but I accept that this approach doesn't suit everyone. At the current price, the DMA2 has to be considered something of a steal. SOS

Alternatives

There are a few other two-channel preamps with integral A-D converters: the slightly cheaper Sonifex RBDMA2 has similar facilities; the Audient MICO offers word clock and multiple digital output formats for a smilar price to A&DR's preamp; Apogee's Mini-Me is a bit more expensive; and the Neve 1073DPD is rather more expensive! Of these, only the Neve offers the repeatable switched gain facilities of the DMA2, and none has the remote-control features. Alternatively, in cost-per-channel terms, the four-channel Focusrite ISA428 is only slightly more expensive, and provides several additional facilities.

This is a dedicated mic preamp, with mic inputs via the usual XLR sockets but no line inputs. The digital outputs are offered on AES-EBU XLRs and there are no analogue outs. An RS232 serial port is also included on the rear panel, allowing full remote control.

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More Thor
In Reason 4

Following last month's introduction to the Thor synthesizer, we embark on a patch-building project and explore several of the key features of Reason 4's god-like new synth...

Simon Price

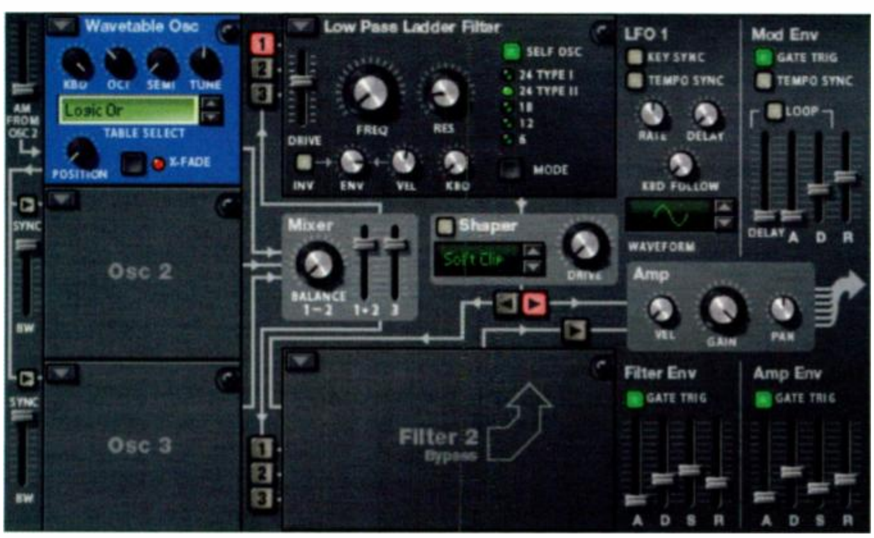
Last month we looked at the basic structure, routing controls, and modulation matrix in Reason's new Thor synthesizer. This time we're going to investigate the Wavetable Oscillator, Step Sequencer and Virtual Controls by following a series of steps towards a finished patch.

Getting Started

Begin by creating an instance of Thor in your Reason rack, then right-click on it and choose Initialize Patch; this will give you a simple starting point with a sawtooth oscillator, a filter, and no modulation assignments. Now click the Show Programmer buttons to see all of Thor's controls. Change Osc 1 to a Wavetable, by clicking on Osc 1's pop-up selector, as shown in the screen above. (For more information on Wavetable synthesis, see the 'What's A Wavetable?' box.)

The Wavetable Oscillator has a central display showing the current wavetable. Click on the display to bring up a pop-up menu, or use the up and down buttons to step through the available wavetables. For this patch we're going to use the 'Logic Or' wavetable.

Play some notes and slowly turn the Position knob to hear the sounds generated at different points in the table. Try experimenting with different settings for Filter Frequency, Resonance and Envelope, and the Decay stage of the Filter Envelope. As you'll hear, a wide range of timbres are available, even from this simple patch.



The basic settings for our patch. The rest of the sound will be programmed in the Modulation Matrix and Step Sequencer.

A Simple Step Sequence

Thor's step sequencer is located at the bottom of the panel, beneath the Modulation Bus Routing Matrix. No assignments are needed to play notes with the sequencer, so we can get stuck straight in. At the left of the sequencer panel you'll find the Run Mode controls. The first slide switch is set to off; change this to Repeat, press the Run button, and the sequencer will begin to play the default sequence of 16 middle Cs at a rate of 16 per bar.

The playback speed of the sequence is controlled by the Rate knob, just to the right of the Run Mode controls. By default, this is sync'ed to the song's tempo.

The rest of the panel is dedicated to the actual sequence, represented by a series of 16 knobs and buttons. A six-way selector knob (labelled Edit) reveals six separate step sequences. The first four sequences have pre-determined functions: Note, Velocity, Gate Length and Step Duration. The final two — Curve 1 and Curve 2 — are left unassigned, ready for you to use as you wish. Regardless of the pre-assignments, you can send any of the six sequences to any destination (including the external CV outputs) using the Modulation Matrix.

With the Edit knob set to Note, use the sequence knobs to create a simple melody. You can disable individual notes using the buttons beneath the knobs.

If you switch to the Velocity sequence, different settings here will change the brightness of the notes, because Velocity is mapped to Filter Envelope by default in the Filter modules.

Custom Sequencer Assignments

Our patch could use something to make it more interesting, so the next step is to give the sound some movement by assigning one of the Curve Sequences to Oscillator 1's wavetable position.

In the first slot in the Modulation Matrix, set the Source field to Step Sequencer / Curve 1 (as shown in the screen below); for the Destination choose Osc 1 Pos and set the Amount to 100. Now we need to create a sequence for Curve 1: start by using the Edit knob to display the Curve 1 sequence, then right-click anywhere on the Thor panel (or go to the Edit menu) and choose Random Sequencer Pattern. Now click Run to hear the results.

Set Osc 1's Position knob fully to the left: the sequencer's modulation acts positively on the current position of the parameter. This setting of Position is where the sequencer modulation will have the greatest effect on the sound.

Virtual Control Key Assignments

Each of the Virtual Control buttons on Thor's main Controller Panel has a small display to the right of it. Clicking and dragging upwards on these displays reveals that you can set a MIDI note value for each button. This is an unusual feature that allows you to momentarily (in the British sense of the word) activate a button (if it is off) by holding down the assigned MIDI note. Typically, you would assign a note to this that you won't need to play otherwise. You can then alter the sound mid-performance simply by holding down the assigned note.

The core assignments for our patch: Wavetable Position is modulated by the sequencer, and the sequencer is triggered and transposed by MIDI notes.

As with step sequences created using Reason's Matrix device, our Thor patch is currently limited to playing the specific notes set in the note sequence. Also, the sequence is triggered whenever you press Play on Reason's transport. It would be better if the sequence only played when you held down a MIDI note, and was also transposed depending on what note you played — so here's how to achieve that. Using the Modulation Matrix, assign MIDI Key / Gate to Step Sequencer / Trigger (as shown in the screen above). The sequence will now be triggered whenever Thor receives a MIDI note. Making this assignment also stops the sequence from being triggered by Reason's main transport.

Assign MIDI Key / Note to Step Sequencer / Transpose (also in the screen above). Now the sequence will play at different pitches depending on which key you play. If you play middle C (C3), the sequence will play back at its original pitch. Play any other note and the sequence will be transposed up or down by the number of semitones between C3 and the note you played. Finally, set the Keyboard Mode (on the main Thor panel) to Mono Retrig.

Virtual Controls

It's time to turn our attention to the Virtual Controls, the two rotary knobs and two buttons on Thor's main Controller Panel. These are used in a similar way to the assignable knobs (or 'macros') on many workstation synths, and also in Combinator. They allow patches to be set up so that choice aspects of the sound can be varied from a simple control panel, without needing to delve into the more complicated guts of the synth.

There is a significant difference between the assignable controls on Thor and those on Combinator. Combinator's knobs and buttons are absolute controls, effectively replacing the control which they are assigned to. Thor's rotaries and buttons are relative (or delta) controls, adjusting a value up or down from a base value. For this reason, the Virtual Controls are called Modifiers.

There are also two different ways in which they can be used. You can use a Virtual Control to adjust a particular parameter directly — a typical example would be to use a rotary to adjust filter frequency. Alternatively, you can use a Virtual Control to scale another modulation or routing assignment (as, in fact, we did in last month's column).

Let's create one of each of these types of assignment. To start with, we'll set Rotary 1 to act as a filter frequency control. In the next spare modulation slot, choose the first knob (Modifiers / Rotary 1) as the source and Filter 1 / Frequency as the destination and set an Amount value of 100. Now set

Assigning the Curve 1 sequence in the Modulation Matrix.

the Filter 1 Frequency control fully anti-clockwise. By setting the original Filter Freq control to zero and using a 100 percent control assignment, you've made Rotary 1 act as a direct replacement for the filter control. This is as it would be if you'd assigned a Combinator knob to the filter with full range. If you set the original control to somewhere other than zero and limited the Amount value, the Virtual Control would adjust the filter frequency over a limited range. You can also set negative values to reverse the polarity of the control.

An interesting aspect of Thor's design is that many parameters can extend beyond the range of their panel controls. This means that modulation sources, including the Virtual Controls, can take a parameter to values that would not be possible otherwise.

Next, we'll use the second assignable knob to scale an existing modulation source. In the first modulation slot (S.Curve 1 assigned to Osc 1 Pos), click the Scale field and choose Modifiers / Rotary 2 (as shown at the bottom of the screen above) and set the Amount to 100 percent. Rotary 2 will now act as a depth control for the wavetable modulation by Sequencer Curve 1. By setting the two Amount fields carefully in an assignment like this, you can finely tune how a control will affect the sound. If the first Amount was 50 percent and the Scale Amount was 100 percent, Rotary 2 would adjust modulation between zero and 50. You could use a number below 100 for the Scale Amount, and this case the rotary would have less effect, so there would still be some modulation at the minimum position.

The two assignable buttons work in exactly the same way as the rotaries, except that they can only toggle between two values. Let's look at a typical way of using a button to switch an effect on or off.

Switch on the Shaper, choose the Soft Clip mode and set Drive to zero. Then, in the Modulation Matrix, assign Modifier / Button 1 to Shaper Drive and set the Amount to about 80. Button 1 will now switch the Shaper Drive between off and nearly maximum. By adjusting the Shaper Drive's panel value and the button control Amount, you can define the two values of Drive for when the button is in or out.

Signal Routing With The Virtual Controls

For the finishing touch, we'll look at an advanced use of the Virtual Controls. As we mentioned last time, Thor's Matrix can be used to route audio signals, as well as modulation sources. By scaling this type of assignment with the virtual knobs or buttons, you can adjust or switch signal routings from the Controller Panel. We'll use our final available button to illustrate an example of this. In the Matrix, assign Osc 1 to Filter 2 / Audio Input and set the Amount to 75. In the Scale field, choose Modifiers / Button 2, and set an Amount of 100. Insert a Formant Filter into the Filter 2 module and press the red arrow button above Filter 2 to route its output into the Amp.

Now, when you press Button 2, audio will be routed from Oscillator 1 to Filter 2, mixing in a differently processed version of the sound. Alternatively, you could leave Filter 2 blank, which would blend in some of the clean signal from Osc 1. Another option is to use one of the rotaries instead of a button to scale the routing, as this would allow you to control the level of signal being routed to Filter 2. Given the range of possible routings, and the fact that you can scale multiple assignments from each Virtual Control, it's possible to create patches whose sounds can be radically altered using just the four panel controls.

Of course, there's a good deal more to explore and understand in Thor, but hopefully this example will give you some insight into some of the many possibilities available to you. SOS

Thor Sequencer Tips

- You can use Thor's step sequencer to modulate played (and sustained) sounds, rather than just triggering notes. Begin by switching off all the sequencer notes from the step buttons and then assign any of the sequence layers to modulate the desired synth parameters. Finally, you can use the assignment in step 15 of the sequence to trigger sequencer playback whenever you play a MIDI note.
- When triggering and transposing sequences from the keyboard, switch off MIDI in the Trigger section of the Controller Panel to prevent double notes when you play keys.
- The way you play affects how sequences are re-triggered. Playing legato will transpose the sequence without resetting the step position. Releasing MIDI keys between the notes will re-trigger the sequence from step 1.

The final patch, complete with Virtual Control assignments for easy tweaking.

What's A Wavetable?

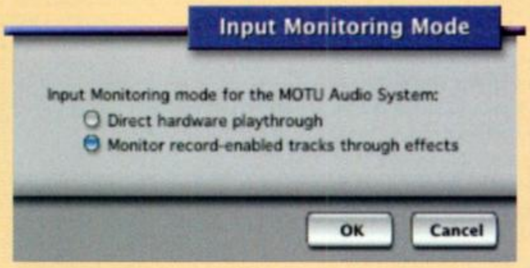
Each of the wavetables in Thor's wavetable oscillator consists of a series of single-cycle waveshapes. At any instant in time, the oscillator is 'looking' at one point in the table (the Position), and playing the waveform at that point. If the Position is between two of the waveshapes and the X-fade button is active, the oscillator will play a blend of the two waves. This gives you a vast range of waveshapes to use as the basis of a sound. If you then modulate Position, with an LFO, envelope, performance controller or similar, the sound will shift over time between the different waveshapes in the table. In most cases, the series of waves in the table change slowly from one wave to the next, resulting in a smooth, natural evolution of the sound when Position is swept.

technique digital performer

EFFECTIVE MONITOR MIXING

Direct Hardware Playthrough

If you use a compatible MOTU interface, you might find you've an extra option in the Input Monitoring Mode dialogue box (in the Setup menu): Direct Hardware Playthrough. It's a nifty feature too. When you select it (and Audio Patch Thru is turned on) DP makes temporary CueMix routings on your interface to route track inputs to their outputs, with zero latency, using the fader and pan settings you make in the Mixing Board. This means that for many recording tasks you can take advantage of CueMix without ever booting up CueMix Console. Direct Hardware Playthrough routings are made in addition to anything you might have already set up in CueMix Console or on the interface itself, so 'zero' those first to avoid confusion.



To prepare DP for the same guitar-then-vocals recording task, go Studio menu / Audio Patch Thru and select Off. This means that when you record-enable a track the input will not be routed to the output, so DP itself will never contribute anything to the monitoring process. Instead, monitoring must be set up as a separate mix, using your hardware mixer or audio interface. In my own project studio setup I use a MOTU Traveler, which has the CueMix Plus monitoring system. So my approach would be to fire up the CueMix Console application and use its on-screen digital mixer to get a live feed of my acoustic guitar. On a hardware mixer you might do your monitor mix via aux sends, or if you have another manufacturer's audio interface, use the zero-latency routing software that's made for it.

In DP, record-enable your guitar track and check input level. Nothing monitoring-related happens when you record-enable, as Audio Patch Thru is turned off. With the input level correct, go for a take. DP's metronome outputs as usual, mixing with your live monitor signal. If you need to adjust the relative level of metronome and live signal it's probably easiest to change the volume of your zero-latency live mix using your mixer or monitoring software. Finally, take the guitar track out of record-enable.

Next, set up a monitoring signal for your vocal track. Check the monitor mix by playing the guitar track while rehearsing the vocal track. If the relative levels are wrong, be prepared to adjust the guitar track fader in DP, the zero-latency mix level and even DP's metronome volume, along with the overall headphone output. Now record-enable the vocal track and go for a take.

It's worth noting that if you had a lot of audio backing tracks in DP and needed to reduce them all in volume relative to your live mix, the easiest way would be to create a Master Fader Track for your main audio outputs, then use that to change the 'from DP' audio with a single fader.

Combining Software & Hardware Monitoring

This is the hybrid approach, using a zero-latency monitoring system but adding reverb or other effects to your live mix using DP.

In DP, ensure your Input Monitoring Mode is set to 'Monitor record-enabled tracks through effects', to use DP's effects processing in real time. Set Audio Patch Thru mode to Blend again, so DP can route audio track inputs to their outputs (which we'll need for a bit of vocal reverb). Then set up a zero-latency monitor mix on your mixer or audio interface, ready to record the guitar backing track, and record-enable the guitar track.

Something undesirable happens at this point: we get two monitor signals, one via the zero-latency mix and one via Audio Patch Thru in MAS (which could have a little or a lot of latency, depending on buffer size): almost certainly phasey-sounding, and not very nice! This guitar track is best monitored via the zero-latency mix, so in DP mute the guitar track to silence the Audio Patch Thru output. Hit record and lay down your track.

Now for the vocals. Take the guitar track out of record-enable and un-mute it so it'll play back. Set up a zero-latency mix for the vocal take and record-enable the vocal track. Once more there are two monitor signals, one with latency and one without, but this time you can use this to your advantage. On the vocal track in DP, instantiate a reverb plug-in, call up a preset, and set wet/dry mix to 100 percent wet. Now the zero-latency mix is providing a dry monitor signal and DP is providing a reverb signal. Adjust reverb level by dragging the vocal track's volume fader in the Mixing Board. When the balance is right, go for a take.

As I mentioned before, there are multiple levels to juggle with this approach: the live monitor mix, backing track levels, the metronome click track, and temporary reverb levels. But once you've established a working method it's pretty easy to co-ordinate. Changing the level of backing tracks with a Master Fader is not so suitable for this approach, as it'll also affect the monitoring reverb level. But a good alternative approach is to create a Track Group of your backing tracks and configure it so that one track's volume fader controls all the others.

In Next Month's Episode

Monitoring is such a big and important subject that it just won't fit in a single copy of *SOS*! Next month, we concentrate on monitoring in a bigger studio, working with multiple monitor mixes, and how best to use DP5's Input Monitor and Aux tracks. SOS

News In Brief

- **Leopard & DP: Following the release of Apple's newest operating system, OS 10.5, or 'Leopard', MOTU have updated most of their software. DP is now up to version 5.13, and introduces no noticeable changes, except Leopard compatibility. MIDI and Audio interface drivers are also now Leopard-friendly, and MOTU's virtual instruments need no updates as long as you have the most recent versions. All are available from www.motu.com. Life in Leopard seems good, too — the Finder is better all round, the QuickLook feature is superb for browsing folders full of samples, and there seems to be no performance penalty, at least on my dual-2GHz G5 and MacBook. The multiple-desktop Spaces feature is also promising, but DP's Consolidated Window scheme isn't quite settled with it yet. More on all of this next month. (We also have an extended Apple Notes column all about Leopard starting on page 218 of this issue.)**

- **Waves plug-ins for Performer? The situation regarding Waves plug-ins and DP has been quite confusing in recent times, with no official support for Intel Macs, and potential AU/MAS mix-ups on Power PC machines. At long last, though, Waves have released a special beta version Waveshell for DP users, compatible with both Power PC and Intel Macs. Interestingly, it's in the Audio Unit format, so it looks like it's the end of the road for DP-specific MAS versions — perhaps no bad thing. Version 5.9.7 installs new Waves plug-ins and a single AU Waveshell, and requires that you're a Waves Upgrade Plan subscriber, as it relies on the installation of an Enabler on your Waves iLok. www.waves.com**

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Using Arpache In Cubase 4

John Walden

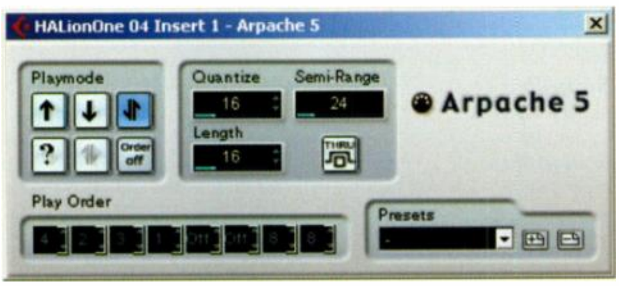
It's so easy to reach for the same audio plug-ins time after time — but MIDI plug-ins such as Arpache can bring something different and shouldn't be overlooked.

Given the extensive and sophisticated audio-processing functionality provided by all modern DAWs, it can be easy to overlook the powerful creative options offered by their less glamorous MIDI features, particularly MIDI plug-ins. So this month we'll concentrate on a couple such plug-ins — Arpache 5 and its more advanced sibling Arpache SX — and explore how you can use them to breathe life into your project.

Arpache 5

I don't know which wag at Steinberg came up with the name Arpache for the arpeggiator plug-in, but I'll set Shadows jokes aside for the moment (the editor has threatened me with something unpleasant if I stray too far into that territory...).

Although they may look simple, Arpache 5 and Arpache SX are both capable of producing excellent results. I'll start with Arpache 5, and as the PDF Plug-in Reference manual does a reasonable job of describing the basic controls, only the briefest of recaps is required here, before we move on to consider what it can bring to your music.

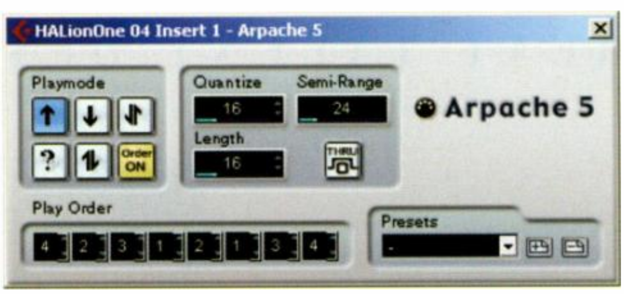


Arpache 5 with a setup suitable for a basic up-and-down arpeggio.

The Quantize, Length and Semi-Range settings define the basic properties of the arpeggio. Quantize controls the bar divisions at which the arpeggio notes will appear, with 32nd notes producing something quite manic at all but the slowest tempos, and a value of 1 producing one note from the arpeggio each bar. Both dotted and triplet versions are available for all time intervals between these extremes. The Length setting controls the length of the notes in the arpeggio. If note lengths are kept short (16th or 32nd settings), an almost staccato feel is created regardless of the synth sound source being used. At longer note lengths, the nature of the MIDI sound source is more significant, as the sustain and decay properties of the patch might come into play. Experimenting with the relationships between the synth sound and the Length setting can produce some interesting variations, although things can get a little OTT if you combine a short Quantize setting, a long Length setting, and a sound source with lots of sustain. Finally, the Semi-Range setting simply defines the range of semitones from which notes for the arpeggio will be taken, relative to the position of the lowest note being played.

The top three buttons in the Playmode section are straightforward, setting the sequence of notes in the arpeggio to play either up, down or up and down. The lower buttons are rather more interesting. The '?' button simply randomises the arpeggio note order. Depending upon the sound being used, this can create a nice variation on the straight-up or straight-down patterns. Perhaps more useful is the Order On button: with this engaged, the note order of the arpeggio is defined using the Play Order facility, which allows a sequence of up to eight notes to be specified. The number relates to the MIDI notes being played into Arpache 5 via the MIDI track, starting with the lowest note. In Play Order mode you can create some almost riff-like progressions (which work well for the usual mid-range keyboard parts) but it's also easy to create interesting bass lines.

Experimentation is the name of the game here, as it can take some time to work out just how the various controls interact with each other. Fortunately, for the Play Order mode there is a small number of presets for users to explore — and you can also save your own patterns as presets.



Play Order mode offers more control over the form of the arpeggio created by Arpache 5.

Wot, No Electro?

Of course, nobody would want to use Arpache 5 to create something suitable for a synth-based dance track... would they? Oh, alright then, if you want to, you can create the classic (clichéd?) synth chord patterns that will return you to a land of '80s pop, or place you very firmly into certain styles of dance music. Basic Apache 5 settings aside, all this requires is a suitable sound source, and the Halion One patch 'Polymood' makes a decent starting point — though there are plenty of other preset sounds in the various Cubase 4 VST instruments that you could put to good use.

There are also some less obvious applications for Arpache 5. For example, used with a suitable bass pad-like sound, a combination of slow Quantize (such as a setting of 4 to produce quarter notes) and a Length setting of 1 (so that each note in the arpeggio sustains for a whole bar) can be used to generate a drone-like bass part, which will have some nice timbral movement as the sustained notes are brought in and out of the arpeggio. Used with the right sound source (for example, Halion One's 'Close To The Edge' patch) and at a suitably slow tempo, this can be made to sit anywhere between a melodic bass pad and sound design. You could also add in some atmospheric percussion (such as the 'Storm' style from Groove Agent 3) — and things can start to get quite scary!

In fact, Arpache 5 can be very effective

It doesn't have to be just dance music: here, Halion One is providing a bass synth pad via a slow arpeggio from Arpache 5. I've also added atmospheric drums from Groove Agent, to create a rather unsettling feel!

with percussive and drum sounds, so let's look at two examples that provide a useful way of exploring this.

First, try a percussive synth sound using your VSTi of choice (the 'Djembe+Marimba Layer' patch in Halion One would be a suitable starting point) and, with a medium-to-slow tempo (70-90bpm), set a Quantize value of 16 and Semi-Range of 12 (the Length setting doesn't make any difference with this kind of sound). It is then a case of experimenting with the Playmode settings. Although the most interesting effects can be created by using the Play Order options, even a simple up and down configuration can produce some interesting rhythmic effects. By gradually adding and subtracting notes from the MIDI keys being held, you can change the rhythmic feel, adding movement and dynamics to the performance.

Exactly the same process can work equally well with straight drum-kit samples, and any GM-based drum kit could serve as a starting point. You're unlikely to come up with a traditional rock or pop drum-pattern using this method, but for a more abstract or experimental piece the approach can generate plenty of interesting material.

SXing Things Up

Arpache 5 has been around for a considerable time but (fortunately) Steinberg retained it when they introduced Arpache SX, as Cubase moved into its SX era, and both these plug-ins have been preserved in Cubase 4.

In terms of basic operation, the two are similar, but Arpache SX replaced the Play Order options of its predecessor with the more sophisticated SEQ mode. Again, the basic use of each control for Arpache SX is described in the Plug-in Reference PDF.

What this PDF doesn't do such a good job of is illustrating the potential of SEQ mode. The key creative element of this mode is the ability to define aspects of the arpeggio from an existing MIDI phrase. This is best done by recording a short phrase (one, two or four bars usually works best) into a MIDI track. This phrase is then simply dragged and dropped onto the box in the lower-left corner of the Arpache SX window.

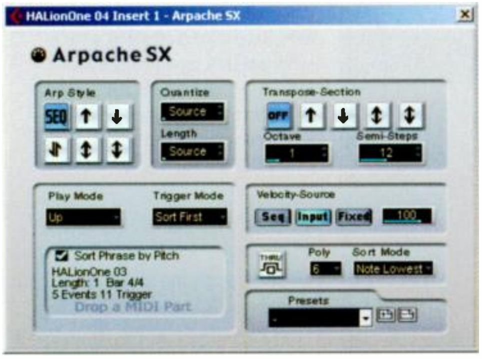
Depending on how the various settings are then configured, selecting SEQ mode allows the relative pitches, velocities and timing of the notes in the MIDI phrase to control how the arpeggio is created by the plug-in. It is also important to note that the number of different MIDI note pitches within the dropped phrase can have an influence upon how Arpache SX works its magic — and a phrase with a larger number of MIDI pitches will produce more complex (and more unpredictable) arpeggios. If you want to explore this further, simply drag and drop a suitable MIDI phrase, select SEQ mode as the Arp Style, and then work through the steps described below.

If Quantize is set to Source, then the timing of the notes in the arpeggio is taken directly from the MIDI phrase, and if the MIDI phrase contains 16th notes, it may produce a fairly standard-sounding arpeggio pattern. However, if the sequence has a few 'missing' notes in an otherwise 16th note pattern, a more interesting rhythmic element is created in the arpeggio. Incidentally, the Quantize value can also be set to something other than Source while in SEQ mode — this simply forces the pitch pattern contained in the dropped phrase onto a regular timing grid defined by the Quantize setting.

The pitches of the arpeggio that's created depend on the Trigger Mode setting, and it is here that the number of different pitches in the dropped MIDI phrase interacts with the number of different MIDI notes being held as a chord and fed to the Arpache SX input. Let's consider an example where the dropped phrase contains five different pitches, but only a three-note chord is being played into Arpache SX to drive the arpeggio. Arpache tries to match the pitches of the MIDI input to the relative pitches of the dropped phrase. If the number of pitches is not the same, then the plug-in needs to know how to deal with that and the Trigger Mode setting provides it with that information. If, for example, we wanted to create a more traditional arpeggiator-style result, the Sort First setting would provide a good starting point: as there are fewer MIDI notes being input than there are different pitches in the dropped phrase, the first input note is used repeatedly to fill in the gaps in the matching process — and this note therefore appears more frequently in the arpeggio. In contrast, if the Sort Normal setting is used, Arpache SX only matches pitches up to the number of input notes: no notes are substituted to fill the 'missing' pitches and, as a result, there are some gaps in the arpeggio. This isn't necessarily a bad thing, as it can create some unexpected and often interesting rhythmic effects.

As a quick aside here, if Trigger is chosen ▶

USING THE ARPACHE ARPEGGIATORS



The Arpache SX plug-in offers plenty of flexibility via its SEQ mode. A MIDI phrase can be dropped into the bottom-left box to provide the arpeggio source.

as the Trigger Mode, then Arpache SX simply triggers the original phrase contained in the dropped MIDI part. If just a single MIDI note is played into Arpache SX to create the arpeggio, this will be used as the root note for the dropped phrase, and when you play a different single note it will simply transpose that phrase. This provides a very simple way of triggering and transposing a riff, and it is also an obvious candidate for bass-line construction.

A final touch of dynamics can be added via the Velocity Source buttons. The three available options — SEQ, Input and Fixed — are fairly self-explanatory. In Fixed mode, the notes of the arpeggio all have the same velocity, and this obviously tends to produce a fairly static (in terms of volume dynamics) output. In SEQ mode, the velocity of the notes in the dropped MIDI phrase controls the velocity of the same steps in the arpeggio, and by editing the note velocities before dropping the phrase into Arpache you're able to add some very controlled volume dynamics — and considerable rhythmic interest — to the resulting arpeggio. With Input as the Velocity Source, the volume dynamics are controlled by the velocities of the individual notes being played into Arpache SX and, again, this allows the player to add some real dynamics to their performance.

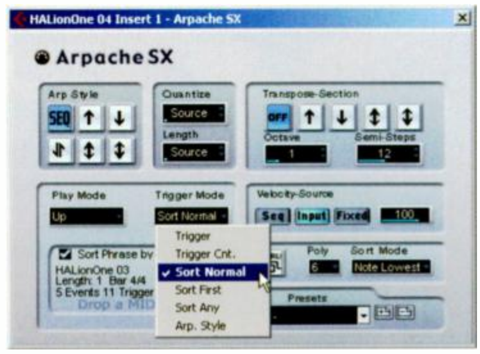
Although there's plenty of fun to be had by dropping any old MIDI phrase into Arpache SX, perhaps a more obvious (though no less interesting) way of using the SEQ mode is to import a MIDI element from elsewhere in your project to use as the arpeggiator source — and a short MIDI bass line or drum phrase can work very well here. The result is a synth arpeggio that is, in some way, rhythmically linked to the source phrase. This can be really effective in helping to generate a tight musical groove, while achieving an arpeggio with a much less 'robotic' feel.

Look What I Played!

Between them, the Arpache 5 and Arpache SX arpeggiators offer a bewildering array of possibilities that could be used in your projects, but occasionally there are times when you can't quite get the result that you want from them.

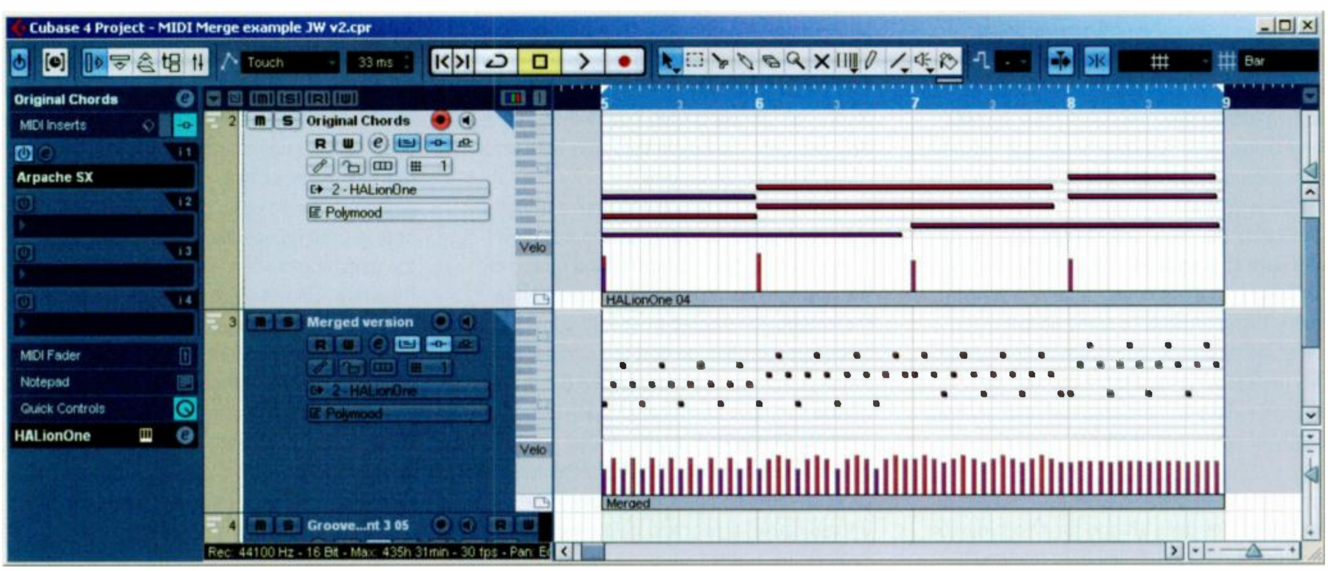
For example, perhaps you've got a great result, except for a few problem notes that simply don't work in the context of the musical arrangement in which the arpeggio is being played. Fortunately, it is possible to transform the output from either plug-in into a conventional MIDI part, containing all the notes from the arpeggio, and this part can then be edited using Cubase's standard MIDI editing tools. Whether you simply need to remove the odd note that is surplus to requirements, or change note velocities to control the volume dynamics, this means that you have complete control over the final performance.

To do this, you need to use the Merge MIDI In Loop function. First, you solo the MIDI track that contains the MIDI part you wish to process. The Left and Right locators must then be set around this part (if you select the part and then press 'P', which is the 'Locators To Selection' key command, the locators will automatically be placed around it). The MIDI / Merge MIDI In Loop menu option will



Trigger Mode influences how Arpache SX deals with matching the number of different pitches contained in the dropped phrase to the number of pitches in the chord arriving at the MIDI input. This setting can produce some interesting variations in the resulting arpeggio.

then bring up a small dialogue box, and if you tick the 'Include Inserts' and 'Erase Destination' options Cubase will obligingly replace the existing MIDI part (which will usually be the chords you are using to drive Arpache) with the actual notes that have been created by the arpeggio process — very neat! Once you're done with processing, though, make sure you deactivate the Arpache plug-in on the track, or you'll find yourself facing arpeggiated mayhem! SOS



The upper track, containing chords, was used to create an arpeggio with Arpache SX. The lower track shows the result of applying the Merge MIDI In Loop process. This new MIDI part can then be edited and brought back into Arpache as required.

New & Improved Effects In Logic Pro 8

Logic Pro 8 has not only introduced a major interface redesign, but also brings powerful new effects, alongside welcome improvements to a number of pre-existing ones.

Stereo Mike

Version 8 of Logic Pro brings with it a number of completely new effects, namely Delay Designer, Echo and Microphaser, which add to the impressive list of effects that were included with Logic 7. Alongside these new additions, stalwarts such as Space Designer, Stereo Compressor, AutoFilter, Linear Phase, Channel and Match EQ have received an overhaul, with new features and updated interfaces.

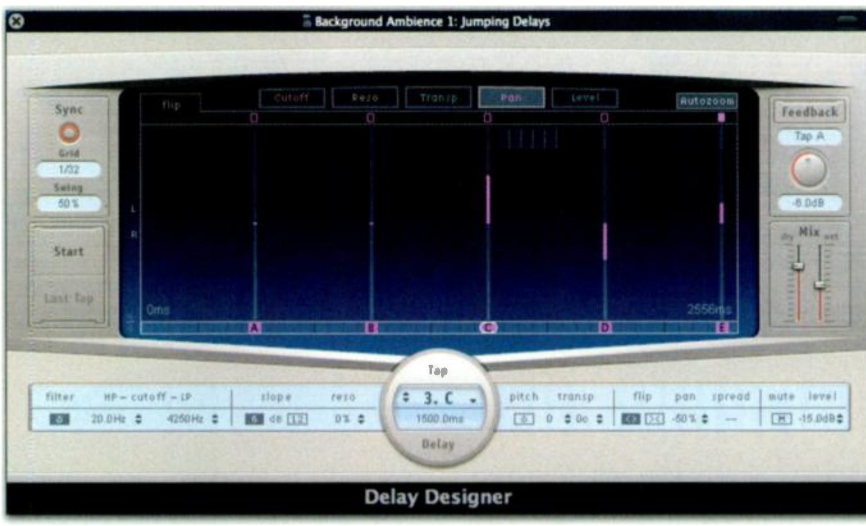
Across the board, plug-ins now feature a streamlined new header and menu with Previous, Next and Compare buttons. Assigning plug-ins from the channels is easier, since menu hierarchy has been rearranged to display plug-ins by type (such as dynamics and modulation, for example), then by name, with a final choice of mono, stereo or mono to stereo, depending on the status of the channel in question. Audio Units plug-ins are amalgamated into the first layer of the plug-in menu, so you don't have to consciously search for plug-ins by format type. In the case of multi-channel surround versions of plug-ins, additional buttons become available in the header, giving access to routing and configuration alternatives.

Let's start exploring the new Logic effects and enhanced features by first taking a tour of the newcomers.

Designer Delay

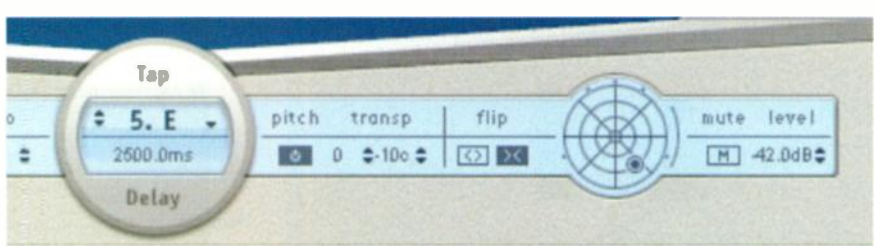
One of the major new additions to Logic's arsenal of plug-in effects is the Delay Designer, an advanced multi-tap delay with an interface that's rather reminiscent of the Space Designer at first sight. This quickly points to the fact that Delay Designer has been developed as the flagship effect in the delay family, just as Space Designer is for the reverb crowd, due to its unique convolution-based features. The key characteristics of Delay Designer are its multi-tap functionality, its ease of use and its programmability during setting up and shaping of the different 'taps'.

Taps are the separate repeats of a multi-tap delay and, in Delay Designer, these can be programmed to a grid, synchronised to the project's tempo and even manually 'tapped' by clicking the mouse rhythmically on the special Tap pads in the user interface. Furthermore, each of the taps can be shaped, using parameters such as timing, filter cutoff, filter resonance, pitch transposition (over a two-octave range), panning and level. The number and types of individually variable parameters for every tap mean that the potential delay patterns can go way beyond classic echo effects, allowing for intricate sound design and the creation of interesting musical patterns that can kick-start sonic or melodic inspiration. Furthermore, the graphical interface is efficient and intuitive in providing a user-friendly alternative to numerical editing of the parameters, which is what you find with some other delay plug-ins. And with 26 separate taps possible — alongside further global parameters — there are quite a few variables to consider.

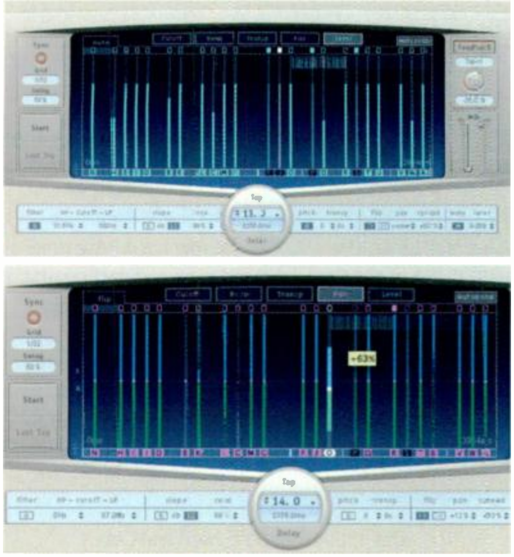


Delay Designer is Logic's new flagship delay. Up to 26 different taps can be programmed, and each one can be individually shaped.

The Delay Designer interface provides five different areas to help in dealing with the numerous design possibilities: the Sync section, tap pads, tap display, tap parameter bar and master section. The Sync section, on the left of the GUI is where the effect can be



The Tap parameter bar is especially useful for entering parameter values when working on individual taps, while the graphical editing methods possible with the Tap display prove more beneficial for work over several taps. Notice how the Pan and Spread areas of the parameter bar are replaced by a surround panner when a surround instance of Delay Designer is being used.



Graphical tap editing in the Tap display is made easy, with lines representing the taps and their lengths representing parameter values. Notice the difference in how Level and Pan parameters are displayed — after clicking the respective Level and Pan View buttons — when a stereo instance is used, with the bright stereo balance dot also representing range, not just values. Option-Command-clicking a Toggle button mutes the tap (as seen on the Tap parameter display), despite the Level view button having been selected.

synchronised to project tempo. Drop-down menus control musical subdivision choices and the swing value percentage, which can be used to great effect when designing syncopated rhythmic delays relevant to the style of your project. For example, you could set it up so that your electronic hi-hats repeat in straight or swung sixteenths, or you could assign a dub triplet repeat to a vocal, for that Massive Attack sound.

The Tap pads (below the Sync section) allow you to 'record' a mouse-click pattern, creating a new tap for each click. Alternatively, taps can be created by just clicking at the desired position of the Identification bar in the tap display, but the former method is more intuitive, allowing the incorporation of a live feel into the timing of the repeats. The latter approach can be used correctively or simply to efficiently adhere the taps to the visual/time grid.

The Tap parameter bar, which spans the bottom of the GUI, displays numerical values for every parameter of the selected tap. It is here that you can get creative with the filter, pitch, pan and level controls. The master section, on the far right of the plug-in, gives access to overall delay feedback and wet-dry mix. These are the only two global-level parameters.

Of course, the effect could be used as a channel insert or set up on an Aux channel, then treated as a send effect, but the latter method is less CPU-demanding when used over numerous tracks, and it also means that the same patterns can be used on various elements in a mix, thus providing a 'locked' overall rhythmic feel.

Tips, Tricks & Indicators

Putting the Delay Designer to practical use reveals just how well the graphical interface works. It's fully possible to shape every tap and the overall rhythmic pattern even if you never leave the main Tap display and, in fact, that's most likely the quickest method when shaping multiple taps for the effect. Operating the plug-in gets even easier once you're familiar with its indicators.

Clicking on a tap in the main display or the relevant tap letter in the identification bar renders it brighter to identify the selection, while click-dragging or shift-clicking allows you to rubber-band or select non-adjacent multiple taps respectively. Clicking a specific parameter's View button and click-dragging vertically on the selected tap's bright line will modify the selected parameter's value, while doing the same for multiple selections will enable relative modification.

You can even command-drag in the tap display to draw in values for multiple taps as an alternative method, which feels very familiar to Hyper drawing. Option-clicking a tap will reset the chosen parameter to a default value. For parameters with two defining values, such as low- and high-pass frequencies for filter cutoff, the bright line will represent a range of values, therefore it can be dragged from the middle to move the whole range or reshaped at the top or bottom of the line to alter the high- and low-pass cutoff frequencies.

With panning, however, things are dependent on the input channel configuration. With mono to stereo configurations, click-dragging the bright line up or down pans the tap left or right, whereas with stereo instances, the pan parameter takes the form of a stereo balance dot with the width of the line representing the spread. Again, click-dragging from the middle of the line will alter the whole stereo balance, while click-dragging on either side of the dot will alter the stereo spread. Notice that, when a surround instance of the Delay Designer is selected, the pan percentage of this parameter bar is replaced by a surround panner instead.

The Toggle bar buttons available for each tap enable activation, de-activation or switching of specific parameters, depending on the selected View button, thus providing a readily accessible graphical switch at all times, while clicking any Toggle button with the Option and Command keys held down switches Mute on and off regardless of the ▶

Big Cat Diary: Leopard Lands

Apple's latest version of OS X has come hot on the heels of Logic Studio and, as is so often the case these days, some third-party manufacturers seem to have been caught with their (virtual) trousers down. The Apogee Duet audio interface I've been evaluating was updated for Leopard compatibility within a few days of the OS release, while my Metric Halo UL-N survived the transition intact. Other manufacturers, however, have been slower in producing new drivers.

A few plug-in designers seem to have been bitten by the new big cat too, but there's nothing like the problems which appeared when Tiger was introduced. Many of the early adopters have had problems running Logic under the new OS, while for others it seems to have gone pretty smoothly. Most of the problems seem to stem from incompatible audio-interface drivers, so if you're taking the plunge, make sure yours is compatible.

I found that Logic 8 ran smoothly under Leopard on my Macbook Pro, which I use for basic recording and editing. However, I've decided to keep my G5 on Tiger for the foreseeable future, as I've got too much legacy software and hardware on there, not to mention the fact that it's running pretty smoothly at the moment (if it ain't broke...).

Speaking of which, I recently spoke to a very successful post-production house who are using what amounts to geriatric computers loaded with what many would consider 'outdated' software. But they have a system that works for them, and they don't want to upgrade until it's absolutely necessary. This is a salutary reminder to those of us who are eager to upgrade as soon as a shiny new bit of software or hardware appears. As a reputable Irish stout producer states, "good things come to those who wait".

Back to Logic 8, and it's interesting that Apple chose to release the 32-bit Logic before the 64-bit Leopard — albeit in a version that does remove the restriction on how much RAM the EXS24 sampler can use, in advance of the extra RAM 'addressability' that would naturally come as part of a 64-bit Logic — should Logic ever become 64-bit. We'll have to wait and see if that happens, although I'm personally not convinced that it's even essential (or desirable) for a DAW. However, I realise that this is an extremely controversial subject, as the heated discussions at www.apple.com testify.

One nice side-effect of Leopard's release is that it seems to have cured the problem some users were having when using a Mackie/Logic control surface alongside some Apogee interfaces. It's nice to have a fix, but this doesn't give much succour to those who are sticking with the old OS. I hope a Logic upgrade fixes the problem for those who prefer stripes to spots. *Stephen Bennett*

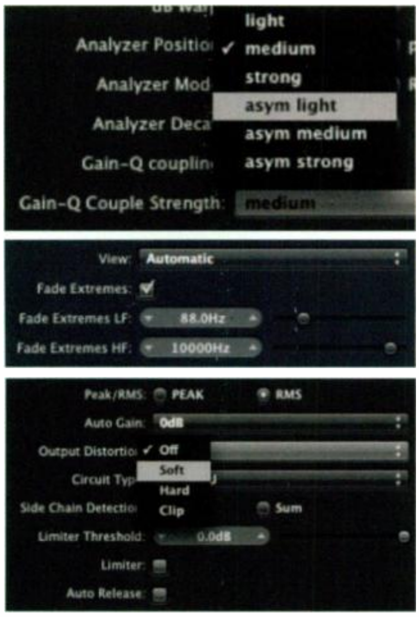
type of View button selected. Releasing the two keys resets the Toggle buttons to their default functionality. Control- or right-clicking a tap drops down a menu — just like the contextual ones available inside Logic's Arrange window — with various copy, paste, reset and delete alternatives for the taps.

What Else Is New?

Things are much simpler when it comes to the fast-and-easy Echo and Microphaser plug-ins, which provide simple delay and phaser effects with a selection of easily accessible and, most importantly, relevant parameters.

The Echo effect automatically synchronises to the project's tempo, while its Time parameter alters delay times in musical subdivisions, including triplets. The other controls, Repeat and Colour, determine the delay feedback percentage and the delay effect's harmonic content respectively, while the self-explanatory dry and wet parameters determine the mix of original and effected signals, and are especially useful if the Echo effect is used as an insert. Echo will come in handy when you're in need of simple delays but you don't want to have too many parameters to consider. However, and after spending a little time with it, it becomes evident just how effective Echo is in quickly achieving anything from an Elvis Presley-style short vocal echo to those Vangelis-like disappearing snare repeats.

Microphaser provides equally useful



A number of new features are lurking behind the Editor views of certain plug-ins (within the Control view, which is switchable from the plug-in header View button), such as the Gain-Q Couple Strength parameter of the Linear Phase and Channel EQs, the Fade Extremes parameters and frequency sliders of the Match EQ, and the Output Distortion modelling options for the Compressors. These can add musicality and analogue colouration to the respective effects and are well worth experimenting with!

functionality, with classic phasing effects, such as synth 'swooshes' and Rhodes or guitar phaser effects. The quickly accessible parameters for achieving these are the LFO Rate frequency, Feedback percentage and Intensity percentage, which determine the speed of the phaser LFO, the amount of the effect being fed back to the input, and the amount of modulation, respectively. With such a simple interface, it won't take you long to get to grips with its functions.

Moving on, the Autofilter, which is not a new effect *per se*, features a completely redesigned interface, with a clearer layout than the Logic 7 version, and some brand-new functions. The filter type is now switchable between high-pass, band-pass and low-pass modes of operation (the previous version was low-pass only), and as a result the whole effect is a lot more powerful in terms of sonic results. The effects achievable here can range from subtle synthesizer filter emulations to radical manipulations of the source audio, like those normally associated with using a Sherman FilterBank or Kaoss Pad, for example. Yet the Autofilter can go far beyond recognisable results associated with the source audio, proving itself as a powerful sound design tool. Again, I'd advise hands-on experimentation to fully discover the power of Autofilter.

Logic's famous Space Designer has also seen some additions, and now includes a four-band EQ with graphical editing options, an expanded impulse response library with new surround impulses, and the equivalent of a surround output mixer for surround instances. In such configurations, the Input slider turns into an 'LFE to Rev' slider, which controls the amount of LFE signal that's mixed with the surround inputs.

The Deconvolve button has been removed, as the Impulse Response Utility that's included in Logic Studio now carries all the necessary features (and more!) for de-convolution.

Compressor Models

Logic's compressors have also been enhanced in version 8. The most significant new feature is the circuit-modelling function, which allows you to change how the Compressor responds. Options include VCA, Opto and FET designs, with two rather more spuriously named models: Classic A_R and Classic A_U. The standard Platinum model, which has been a feature in Logic for many years, is also

The Surround Balancer plug-in equips Logic with a useful interface for making the most of surround signals.

available. Other new features include an integrated limiter, an automatic release-time option and, last but not least, new extended parameters such as side-chain filtering and analogue-style overdrive.

Another new feature is the option to change the Compressor's output distortion characteristics, with options for Soft, Hard and Clip. Additionally, you can control the characteristics of the side-chain, with band-pass, low-pass, high-pass, parametric and high-shelving filter modes, and sliders for altering the frequency, bandwidth and gain of the filter.

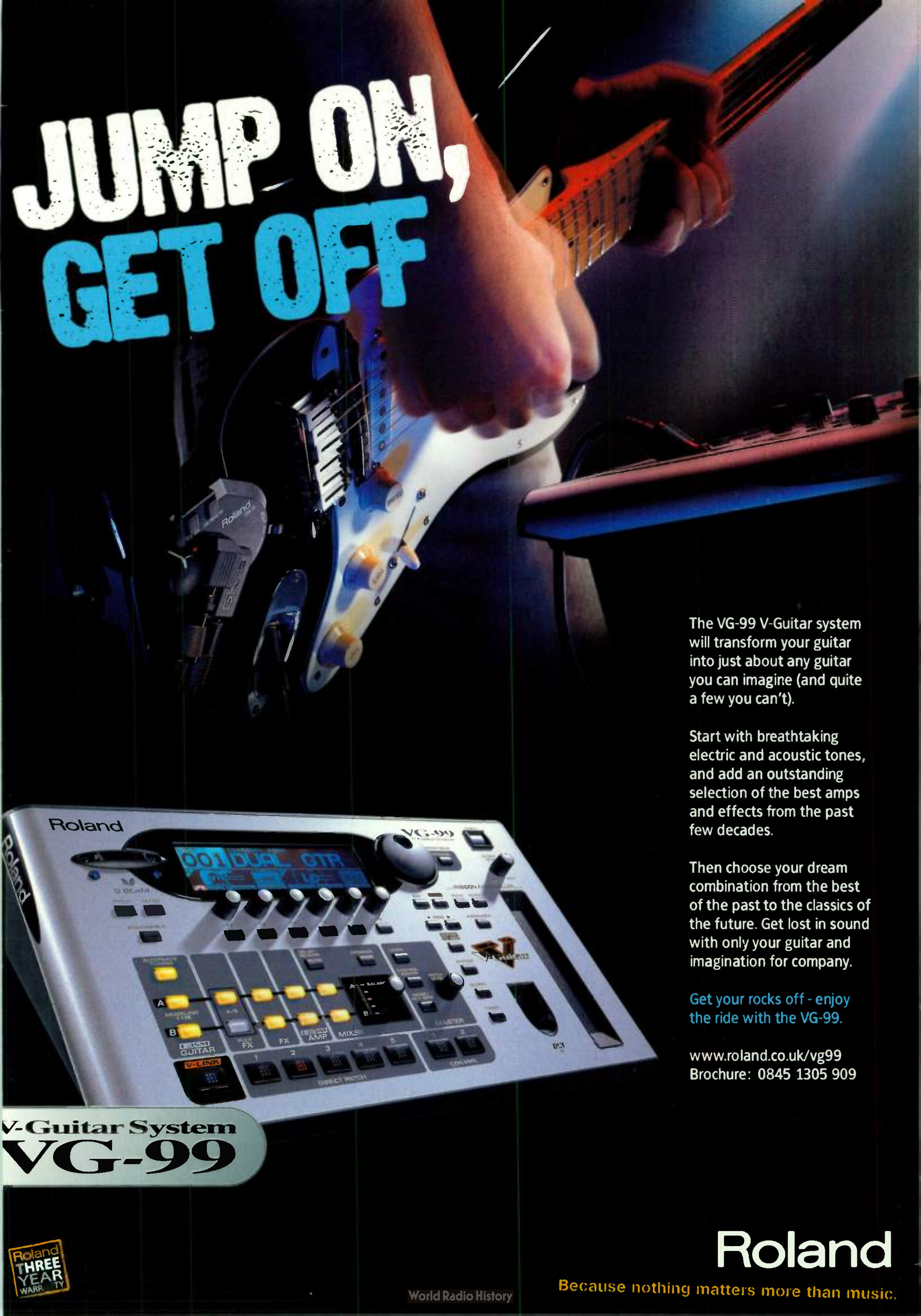
True Surround

The surround aspects of Logic Pro 8's effects have been greatly enhanced with new features for the Space Designer, new plug-ins with extensive surround functionality, such as the Delay Designer, and even completely new true surround effects. Furthermore, the LFO cycles for all Modulation effects are now locked and equally distributed across all surround channels, while the LFO Phase parameter and Distribution menu become additionally available for the Modulation Delay, Tremolo and Phaser effects. This provides further control over the phase relationships between separate channel modulations and over the way the phase offsets are distributed within the surround spectrum (for example in circular, random, front-to-rear and left-to-right distribution).

Likewise, the Spread parameter in the Ensemble effect allows for surround distribution of its voices, with the Down Mixer providing a handy stereo-check option if used on the surround master channel strip. The surround version of the Match EQ plug-in provides separate filter responses per surround channel, with independent editing functionality. And the interface redesign is topped off with the enhanced Surround Panner window, making it easier to maximise the new functionality. SOS

Current Versions

Mac OS X: Apple Logic Pro v8
Mac OS 9: Emagic Logic Pro v6.4.2
PC: Emagic Logic Audio Platinum v5.5.1


JUMP ON,
GET OFF
The VG-99 V-Guitar system will transform your guitar into just about any guitar you can imagine (and quite a few you can't).
Start with breathtaking electric and acoustic tones, and add an outstanding selection of the best amps and effects from the past few decades.
Then choose your dream combination from the best of the past to the classics of the future. Get lost in sound with only your guitar and imagination for company.
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Because nothing matters more than music.

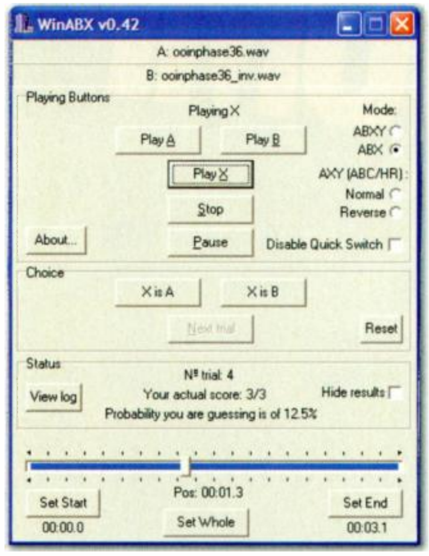
You might think you can hear that one sequencer sounds better than another, but are you fooling yourself? Your PC can help you find out. But first, if you're thinking of building a new computer, read on...

Martin Walker

Things change extremely rapidly in the world of PC components, so it's probably about time I updated the parts recommendations I gave in *SOS* February 2007 as part of my 'Specifying & Building A Dual-core Desktop PC' feature. At the time I ended up choosing an Intel E6600 2.4GHz dual-core processor, but I'd now recommend a Q6600 quad-core processor, with all four cores clocked at 2.4GHz. With most audio software the latter will yield up to double the performance of the dual-core processor, yet as I write this the Q6600 only costs around £30 more than the E6600, at just £170, making it an absolute bargain!

My choice of motherboard was Intel's DP965LT, due to its combination of competitive price and generous complement of PCI and PCIe slots, plus USB 2 and Firewire 400 ports. However, while this motherboard does support the Q6600 processor (courtesy of a BIOS upgrade), I'd recommend that anyone building a DIY PC from scratch today should consider Intel's DP35DP. As its name suggests, the chip set used is Intel's newer P35, the 'replacement' for the P965. Compared with the DP965LT, the DP35DP provides the same complement of three PCI slots, three PCIe x1 slots, one PCIe x16 graphics slot and two IEEE 1394a (Firewire 400) ports, but ups the complement of USB 2.0 ports from 10 to 12.

Both Intel motherboards support up to 8GB of system memory, and both specify that you fit 1.8V DDR2 SDRAM, such as Corsair's Twin 2X2048-6400C5 DHX (Dual-path Heat Xchange) with the new Dominator Heat Sinks for maximum cooling, which you can currently buy for about £125 for 2GB. It is possible to save some money by installing Corsair Twin 2X2048-6400C4 RAM, which is significantly cheaper (around £50 for 2GB in the UK), but some people have had problems booting the DP35DP motherboard with this installed. It's up to you whether you want to take this risk.



If you want to find out whether one sequencer or plug-in is truly better than another, try an ABX test (see next page for more details).

When choosing a graphics card you could stick with my previous recommendation of a fanless Nvidia Geforce 7300 model, which is still perfectly adequate for the musician, as well as being very reasonably priced. However, a card such as the GigaByte 8600 GT silent model might be a better bet if you want to future-proof your PC, as it adds support for DirectX 10.

DirectX 10 (Microsoft's latest graphics technology, offering new levels of 3D realism and detail) is only currently available to Windows Vista users, and not as an upgrade for Windows XP. This has enraged games players (arguably the people that are likely to benefit most from the improvements), because, according to on-line surveys, less than 10 percent of them have moved over to Vista, with the knock-on effect that few game developers are bothering to support DX10. Nevertheless, if you intend to move over to Vista during the life of your new PC, ▶

PC Snippets

• **Free Drums:** If you're into real drum sounds but have no money, point your browsers at the new Blue Noise web site and download the currently Beta version of My Drumset, written by Norwegian musician and studio owner Bjarte Ludvigsen. This PC-only free VST Instrument was reviewed in last issue's Plug-in Folder and is very much in the style of BFD and EZ Drummer, with outputs for each drum, plus overhead and ambience mics. The sounds are 24-bit and were captured from Bjarte's own Ludwig kit using high-quality mics and preamps. Even in this early version the sounds are good, although the download itself is larger than many, at 129MB, due to the 24-bit audio files employed (which also require lots of system RAM). Other goodies to try out while you're visiting the Blue Noise site include the Theremin instrument and Mountain Echo plug-in. www.bluenoise.no

The 'My drumset' VST Instrument from Blue Noise features 12 channels of well-recorded 24-bit Ludwig kit recordings, yet at the moment is a free download. Generosity indeed!

• **Keyboard & Mouse Sharing:** Do you fancy sharing your mouse and keyboard between multiple computers without needing a KVM (Keyboard Video Mouse) hardware switching box? Synergy is a shareware utility that runs on Windows 95, 98, ME, NT, 2000 and XP, as well as Mac OS 10.2 and higher and Unix across a TCP/IP network. You still need a separate monitor screen for each computer, but swapping from one to another is as simple as moving your mouse off the edge of one screen and onto the next, since all the monitors form a single large virtual screen (screens can be arranged side by side, above and below, or in any combination of orientations). Meanwhile, anything you type on your keyboard is sent to the screen containing the 'active' mouse cursor. Clipboards are combined, so you can cut and paste data from one machine to another, and if you have an active screensaver this can kick in on all machines simultaneously. Many thanks to *SOS* reader Jayne Drake for letting us know about this handy utility! http://synergy2.sourceforge.net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having a DX10-compatible card might be worthwhile.

I still stand by the hard drive recommendations I made in my February 2007 article (Seagate, Samsung and Western Digital) but most manufacturers, including the three that I've just mentioned, now make very quiet drives at very low prices. I bought 250GB models, but with 320GB drives now selling at an absurdly low £50, you might want to install this slightly larger capacity (especially since sample libraries seem to be getting bigger by the day). I'm still very pleased with the Pioneer DVD burner I fitted, and with my UK-built PaQ case (www.paqt.co.uk).

Double-blind Audio Tests

There still seem to be loads of musicians who confidently claim that one sequencer sounds better than another when mixing together the same set of audio files, or that one plug-in clearly sounds better than another, using the same settings. Sadly, it's easy to fool yourself that a new product sounds better than the old one, especially if it features better graphics or you've just spent a lot of money on it!

To remove any preconceptions or unintended bias from audio comparisons, you need to listen to each in turn without knowing which one is which (a blind test). You could ask a friend or colleague to switch between the two, but they might inadvertently do or say something that sways your decision. The most reliable way to compare two audio plug-ins, sequencers, or any other audio products is, therefore, to perform a double-blind test — one in which you have no idea which of the products you're listening to, and neither does the switcher.

Implementing double-blind tests is an ideal task for the computer, and there's a very good freeware utility, named ABX, that does exactly that (www.kikeg.arrakis.es/winabx/winabx.zip). I first mentioned ABX way back in PC Notes November 2000, but since then its interface has been enhanced and there are now lots more example files to download from the associated web site to test out your powers of audio discernment.

All you do is launch the ABX utility and choose an 'A' file and a 'B' file in WAV audio format. Then you can start testing. You can audition either file at any time using the 'Play A' and 'Play B' buttons, and switch between them at will while they are playing, but the trick is to then click on the 'Play X' button and decide whether it's 'A' or 'B'. Once you've done at least five such comparisons, the statistics readout will show whether you're guessing or can reliably tell a difference.

There's a Training Room (www.pcabx.com/training) where you can download tests for tone quality, loudness differences and harmonic distortion, and a Technical Listening Room (www.pcabx.com/technical) with files of differing bit-depths and sample rates, with polarity inversion, added jitter and low-pass and high-pass filtering, all of which will enhance your audio discrimination.

To test out different sequencers or plug-ins, just choose a suitable audio file or files as an input signal (from commercial tracks or your own songs), export the output signal from each sequencer or plug-in in turn as a separate WAV file, and then load them into ABX to see whether or not you can really tell one from the another, or whether you've been fooling yourself. You might even discover that you prefer the sound of a freeware plug-in to an expensive commercial alternative!

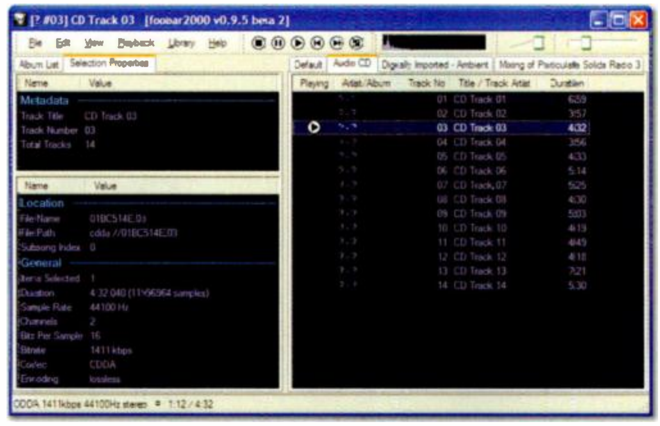
Audio Players

To simply play back some audio files from hard drive, many PC musicians reach for Microsoft's default Media Player, which plays a wide variety of video and audio file and playlist formats. However, there are plenty of other software media players that are either more versatile (supporting a wider selection of audio formats or offering lots of plug-in extras) or easier to use. You can find a representative list for a range of operating systems on Wikipedia (http://en.wikipedia.org/wiki/Comparison_of_media_players).

For example, for playing back RealMedia files I've abandoned the rather bloated Real Player with all its background processes, registration forms and advertising, in favour of the much slimmer and less demanding Real Alternative (www.free-codecs.com/download/Real_Alternative.htm), which includes the Media Player Classic graphic interface.

The most popular alternative player must be WinAmp (www.winamp.com), although the huge 2655 graphic skins and 3576 plug-in extras currently on offer can be rather overwhelming. Nevertheless, once you wade past the host of plug-ins offering basic effects such as EQ, compression and reverb, there are some that can be of genuine use to the musician, including binaural and surround offerings for improved headphone listening. However, just as with RealPlayer, some WinAmp users are beginning to find the latest versions of the player rather bloated.

If you fancy an audio player with a minimalist and easy-to-use interface that doesn't have graphic bells and whistles, Peter Pawlowski's foobar2000 (www.foobar2000.org) could well suit you. It supports a wide range of both high-resolution and compressed audio formats (although not RealMedia, as yet), there are options for dithering and noise shaping, so you can play back exotic files through more mundane audio interfaces, and it's also hugely configurable.



While its interface might be bland, foobar2000's audio options include Kmixer and ASIO driver support, plus a software DTS surround decoder, making it of particular interest to the PC musician.

However, it's the various additional downloadable components (http://wiki.hydrogenaudio.org/index.php?title=Foobar2000:Components_0.9) that make foobar2000 specially interesting to the musician. These include Kernel Streaming support, which, as I explained last month, lets you bypass the Windows Kmixer when using Wave/MME drivers, thus avoiding 'behind the scenes' sample-rate conversion. Another option is ASIO support (rare in mainstream audio players), which not only avoids SRC but also offers much lower latency, and there's also an Impulse Response Convolver, so you can add captured reverb and other effects to audio playback.

Perhaps the most unusual offering is a software decoder for DTS (Digital Theatre Systems) surround-encoded audio files (www.saunalahti.fi/~cse/foobar2000/foo_input_dts.zip). Foobar2000 even includes ABX comparator functions, as discussed earlier, with the added advantage that you can compare different audio file formats, such as WAV and MP3 (to see if you can reliably hear what's been discarded by the conceptual codec), or different MP3 bit rates. Foobar2000 is now my default player for CD audio, WAV files and Internet-radio listening. SOS

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Mark Wherry

Leopard is finally with us — but while it promises improvements for general Mac users, will it offer anything to musicians and audio engineers other than incompatibility?

Apple originally announced the successor to Mac OS 10.4 'Tiger' at their August 2006 Worldwide Developer Conference (WWDC). At the time, they hoped to be shipping Mac OS 10.5 'Leopard' a year later, at the 2007 WWDC; but with the effort required to launch the iPhone, Leopard's release was pushed back to October and this year's WWDC saw the unveiling of the remaining core features Apple had developed for Leopard.

And so, on the last Friday in October at 6pm, Leopard went on sale, and those who visited an Apple Store that night — myself included, I must confess — were rewarded with a Leopard t-shirt, which, if nothing else, is great for those days when you're doing laundry. Leopard itself comes in Apple's new, smaller-style packaging previously used for iLife/iWork '08, and has a snazzy purple space hologram on the front. Inside is a DVD containing both Power PC and Intel versions of the new operating system; the installer will automatically take care of making sure you end up with the appropriate version. There's also a small, 80-page manual that gives you a visual overview of the new features.

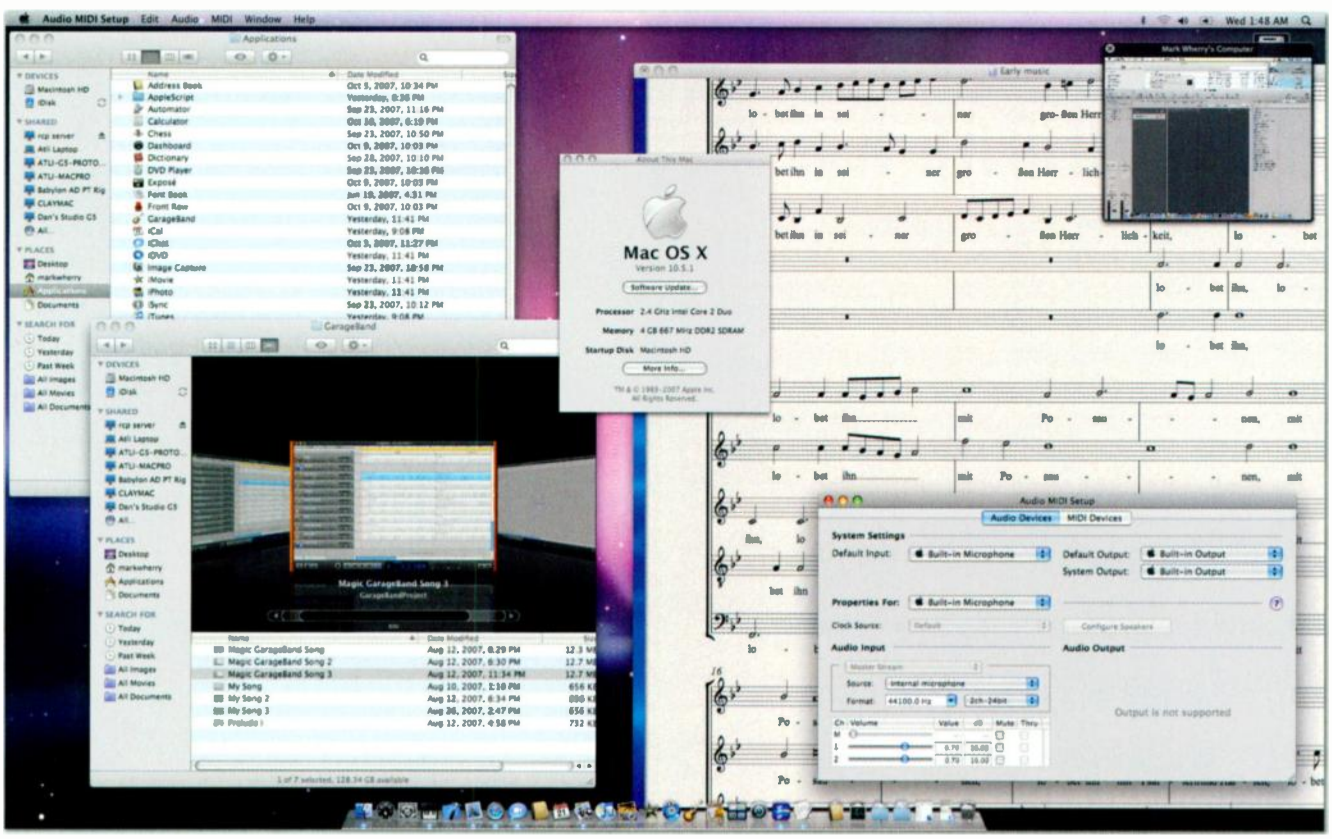
It seems to be a trend with Mac OS X that even-number releases add more music and audio-related functionality than odd-number releases. Mac OS 10.2 (Jaguar) introduced the Audio MIDI Setup utility and was arguably the first release of the operating system that became widely adopted by musicians and audio engineers, while 10.4 included a new Network Device in Core MIDI, along with the ability to use many audio interfaces as one by creating a Core Audio Aggregate Device. So in Leopard, while Core Audio has been improved in certain areas, there aren't any significant new features for music and audio work, although musicians and audio engineers will still arguably benefit from many of the improvements Apple have made in the user interface and at the core of the operating system.

User Inexperience

The most obvious changes in Leopard are those to the general user interface, specifically the new Dock, Menu Bar and Finder. The Menu Bar has lost its friendly rounded edges (even though menus themselves now have rounded edges), and instead gained a degree of transparency, allowing it to blend more seamlessly with your desktop background. This looks quite pleasant when you're looking at your desktop, but after a while I found I had to use a greyscale image on the desktop because it looked a bit odd running a full screen application like Logic Pro and seeing a colour-tinted Menu Bar from the desktop background.

The Dock has been given a new 3D perspective, and the icons now sit upon a shelf that reflects its contents and whatever windows are hovering above. Instead of a clearly visible arrow under a running application's icon, the Dock now displays a blue light, which is hard to see when you set the Dock to display smaller icons. However, if you position the Dock to the left or right of the desktop, it returns to a more 2D, Tiger-like appearance.

In addition to the questionable aesthetic improvements, the Dock also offers a new feature called Stacks. In previous versions of OS X, when you dragged a folder to the Dock, the icon for that folder would be added. You could then either click on it, whereupon the



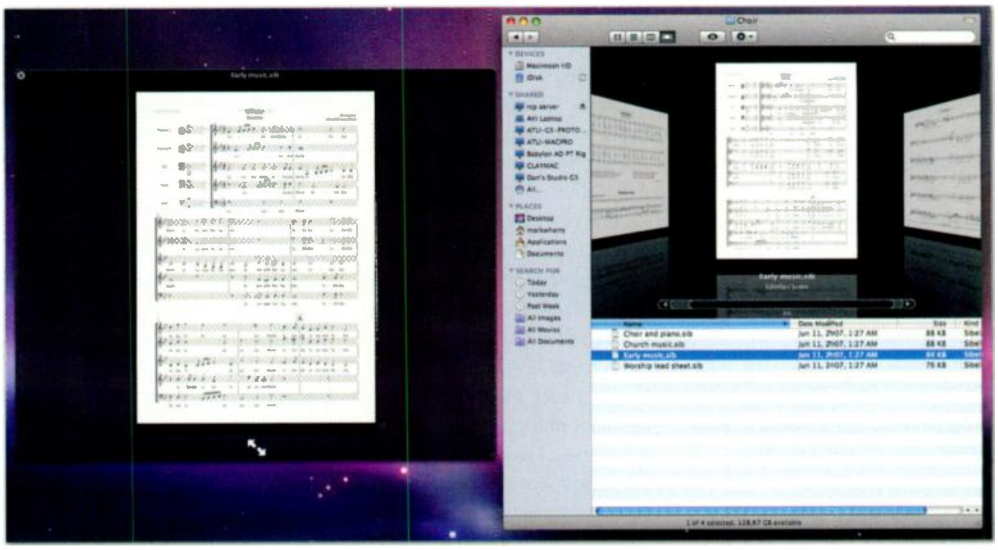
Mac OS 10.5 'Leopard', in all its glory. Notice the new iTunes-like Finder, and the unchanged Audio MIDI Setup utility. In the top-right corner there's a window that shows the new ability of iChat to share the screen of another user, which can be useful for collaboration, and in the background you can see Sibelius 5.1 running. Notice how the new Dock along the bottom reflects the contents from the Sibelius window.

folder would be opened in the Finder, or Control-click it, which would enable you to navigate the folder from a pop-up menu, where the sub-directories appeared as sub-menus. Dragging a folder to the Dock in Leopard now creates a Stack, and the Stack always shows the icon for the first item in the folder, which will vary depending on how you've set the Stack's 'Sort by' option, which you can do by Control-clicking on the Stack.

Although I knew about the behaviour of Stacks, I still felt a bit disconcerted when the shortcut to the Applications folder I keep in my Dock (which is now a Stack, of course) appeared with the Address Book icon instead of the Applications folder icon itself, as before. After a bizarre sense of needing to rebuild my desktop from the days of OS 9, I realised that the Address Book is, of course, alphabetically the first item in the Applications folder, but it seems a shame you can't tell a Stack to use the static icon of the folder it represents. The workaround is to create a dummy file in the Stack with an appropriate icon, but this seems unnecessarily clumsy.

Clicking a Stack displays the contents of the Stack as a Fan or Grid. A Fan splays the items in a manner not dissimilar to the Leaning Tower of Pisa, while a Grid shows the items in, well, a Grid. A Fan shows fewer items than a Grid, but unless you've specified a preference in the Stack's 'View as' option, the Stack will automatically display a Grid once the Fan's number of items threshold has been passed. However, even Grid view won't be able to display all the items in a Stack, so you'll need to use the 'Show in Finder' button when you can't find the item you're looking for in the Stack.

As you can probably tell, I'm not a big fan of Stacks. Perhaps the real killer for me is that when you want to look in the sub-folder of a folder that's a Stack, you have to click the folder and have it open in the Finder. Previously, if I was opening the Audio MIDI Setup Utility, I'd right-click on the Applications folder in the dock, hover the mouse over the Utilities folder and select Audio MIDI Setup from the sub-menu. Now, to do the same thing in Leopard I have to click the Applications Stack, select Utilities, double-click the Audio MIDI Setup icon in the Utilities window that's opened in the Finder, and then close the Finder window once I'm done.



With a suitable Quick Look plug-in, documents can be browsed more visually and previewed without having to open the applications that created the documents. Here you can see scores created with Sibelius being browsed in the Finder, with the Quick Look preview window displayed to the left. Sibelius is one of the first music applications to support Quick Look, and, as you can see, there are some improvement to made. In particular, a higher-resolution preview would be nice.

Not all of the aesthetic changes in Leopard are bad, though. The brushed-metal appearance has now been eradicated and generally everything looks a bit more streamlined and sophisticated. The drop-shadow effect applied to the foremost window has been enhanced to make it stand out even more, and while menus and other panels still have a degree of translucency that shows the contents behind them, this background now becomes subtly blurred to allow the foreground text to be more visible.

A Quick Look In The Finder

A big change in Leopard is the introduction of a new Finder, combining the best ideas of the previous Finder with Apple's iTunes application. Perhaps the most important new feature, which the Finder benefits from, is Quick Look, which can provide a visual preview of a document without having to run the application that actually created the document. For Quick Look to be able to do this, each different type of document needs a corresponding Quick Look plug-in that knows how to interpret that document type and provide the relevant information to the operating system. This is similar to how Spotlight works, and Leopard ships with Quick Look plug-ins for common file formats.

Once a document type is supported by a Quick Look plug-in, you'll notice documents of that type displaying icons that reflect their content, rather than generic icons, as before. You can preview a document by clicking the Finder's Quick Look button or simply pressing the Space Bar. If the document isn't supported by Quick Look, the preview window will instead show some generic information about the file, such as its size and last modified date.

In addition to the three standard views that have always been available in OS X's Finder (icon, columns and list, which now has background shading for alternate entries), Leopard introduces a fourth view — Cover Flow — which enables you to flick through your files in the same way you would flick through media in iTunes. Cover Flow in the Finder also makes use of Quick Look to provide richer visual clues as to a document's content, and although I was a bit sceptical about its usefulness, I have to say that I started to see the potential when flicking through Sibelius scores, thanks to a Quick Look plug-in that Sibelius introduced with the recent 5.1 update. It really is a much nicer way to find a score I'm looking for, although this is admittedly a very specific purpose.

As I mentioned in a previous Apple Notes, an application like Sibelius really lends itself to being previewed in this manner, but the only other music application I came across that supports Quick Look at the time of writing was Apple's own GarageBand '08. In GarageBand, however, you simply get a screenshot of the state of the application when the song was last saved, although this is also surprisingly useful when used with the Finder's Quick Look preview function.

The 64-Bit Question

One of the most significant 'under the bonnet' changes in Leopard is support for full 64-bit applications — assuming you have a Mac with a 64-bit Intel or Power PC processor. One particularly nice aspect of the 64-bit support in Leopard is that, unlike Windows or Linux, you don't have to install a specific 64-bit version of OS X in order to run 64-bit applications. ▶

Another useful thing about 64-bit applications on Leopard is that, in the same way Universal Binaries contain both the Power PC and Intel versions of an application in one package, a single package can also contain 32- and 64-bit versions of the application — for both architectures — which is pretty handy.

Although you could run 64-bit applications in Mac OS X since Panther (10.3), the big catch was that not all of Mac OS X's programming frameworks offered 64-bit support, such as those frameworks that allow programmers to create graphical user interfaces. This meant that if you wanted to write an application that took advantage of 64-bit memory addressing, you actually needed to create two applications: a back-end application that handled the 64-bit addressing, and a front-end one that presented the user interface. And since the Core MIDI and Core Audio frameworks used by most Mac music and audio software weren't 64-bit compatible either, creating a music or audio application that supported 64-bit memory addressing was quite impractical.

In Leopard you can now, finally, create a 64-bit application with a graphical user interface that supports Core Audio and Core MIDI. However, despite this new, enhanced 64-bit support in Leopard, it may yet be some time before we start to see 64-bit music and audio applications running on the Mac.

If you think back to when Apple first released OS X, there was much talk of Carbon and Cocoa, the two different programming APIs (Application Programming Interfaces) that developers could use to create applications for OS X. Carbon was an evolution of the old Mac OS 9 API, while Cocoa was a completely different API based on technology Apple had acquired with the purchase of NeXT, the company Steve Jobs founded after leaving Apple in 1985. While Cocoa makes it really easy for developers to create modern applications for Mac OS X, Carbon makes life slightly easier for developers working on cross-platform applications.

Apple originally announced that Leopard would allow developers to create 64-bit applications that used either Carbon or Cocoa for their user interfaces, but at this year's developer conference they changed tack and stated that only Cocoa would gain 64-bit support in Leopard. Now, although I haven't seen the source code for he major music applications on the market, I would speculate that the majority of cross-platform software currently uses Carbon to some extent for user interfaces, especially if it was released on the Mac prior to OS X. And this means, of course, that it's going to take a bit more work than most developers anticipated to offer 64-bit versions of their products to Mac users.

In the case of Apple, it's possible that Logic 8's user interface was developed using Cocoa, which will make it much easier for a 64-bit version of Logic to be released, but this is highly speculative, as Apple have yet to make any announcement regarding 64-bit support in any of their applications.

Even if we do get some 64-bit applications, the next problem will be plug-ins. A 64-bit application will not be able to use a 32-bit plug-in directly, which means that developers will need to release 64-bit versions of their plug-ins as well. And since many Audio Units and VST plug-ins probably use Carbon for user interfaces, plug-in developers will also need to switch to Cocoa for user interfaces. However, assuming 64-bit music and audio applications do take off on the Mac, it's possible we may see some bridging technologies, such as Steinberg's VST Bridge that allows Power PC plug-ins to run in the latest versions of Cubase and Nuendo on an Intel Mac, making it indeed possible to run 32-bit plug-ins in a 64-bit application. This is the approach that both Steinberg and Cakewalk have taken for 64-bit Windows applications.

A Friendly Leopard

New Leopard features aside, perhaps the most important question for any Mac-based musician and audio engineer will be compatibility. After all, the various bells and whistles of an operating system upgrade are pretty useless if the applications in which you carry out your day-to-day tasks are incompatible. However, the good news is that, for the most part, Leopard is one of the least disruptive upgrades in Mac OS X's relatively brief history, although, as usual, your mileage will vary depending on your chosen products.

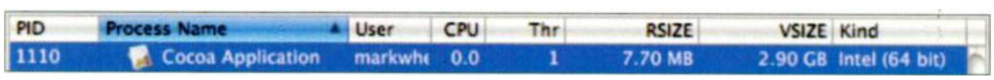
It's usually best with any operating system upgrade to simply wipe your Mac and perform a clean install, and you can always perform a clean install on a separate hard drive if you want to play around with Leopard before committing to it as your main system. However, when I first got Leopard I decided to try upgrading the existing Tiger installation on my MacBook Pro, just to see what happened. To my surprise, it worked pretty well: all of my settings were transferred flawlessly, and most of my music and audio-related applications and hardware devices worked just fine. There were a few problems, such as the Digidesign and Apogee issues mentioned later in this section, but it really was rather painless. That said, I still felt more comfortable when I later wiped the same laptop and started again from scratch; but at least if you absolutely have to upgrade an existing installation, it will work, although I'd still make sure you back up your system before attempting such an operation.

Mac OS X In SOS

For more information about music and audio-related features added to previous versions of Mac OS X, be sure to check out these past articles:
- **Mac OS X For Musicians, April 2003, covers 10.2 (www.soundonsound.com/sos/apr03/articles/osx.asp).**
- **Mac OS X Tiger: A Musician's Guide, July 2005, covers 10.4 (www.soundonsound.com/sos/jul05/articles/tiger.htm).**

Logic 8: For compatibility, Logic 8 seems to work just fine with Leopard, as you would hope, and Apple posted minor compatibility updates for the applications in Final Cut Studio 2, which can be downloaded via Software Update.

Pro Tools: It unfortunately goes without saying that if you're a Pro Tools user you absolutely don't want to even think about upgrading to Leopard until Digidesign officially qualify such a configuration. According to Digidesign's web site, the company are "working closely with Apple to bring support for the entire Pro Tools product line to Apple's latest operating system." Just to be unofficial, I experimentally tried plugging an M Box interface into my Leopard-powered MacBook Pro, but all I succeeded in doing was producing a kernel panic.



There's not much to see, but here's Activity Report showing information about a simple graphical Cocoa application called, er, Cocoa Application, which I created in Xcode 3. Note that under Kind, Activity Reports that it's a 64-bit Intel process. It's a start...

Sibelius 5: Staying with the Avid family, Sibelius 5 claimed to be Leopard compatible when I reviewed it back in September's *SOS*, and although the application does indeed run, I encountered a few issues. Sibelius reported that previous versions of the program can't save when running under Leopard, but I also encountered this problem in Sibelius 5. Installing the free 5.1 update solved the problem, however, and also added Quick Look support, as mentioned earlier, along with a Spotlight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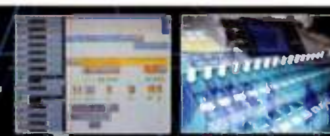
Master of Recording Arts

Department of Contemporary Music Studies, Macquarie University, Sydney



WHAT: The Master of Recording Arts is a one year degree by course work providing students with an understanding of the processes, practices and issues involved in the production of recorded music. The course delivers a detailed program covering recording, production and mastering techniques, supported by a study of the social context in which these occur. A core component of the course is a flexible research project centered on recording projects tailored to individual student needs.

WHO: The course is ideal for students who have recently completed undergraduate studies at Macquarie or other institutions and would like to gain further skills and experience in studio and remote/portable recording environments through the direction provided by coursework teaching. Overseas students intent on pursuing a professional recording/production focussed career in music will also benefit from the program. The course is also aimed at experienced workers in the music and allied industries interested in pursuing further studies to complement their existing knowledge. Students will undertake a focussed, short-term research project catering to their own interests and needs.

CONTACT: Dr Denis Crowdy for more information Denis.Crowdy@mq.edu.au
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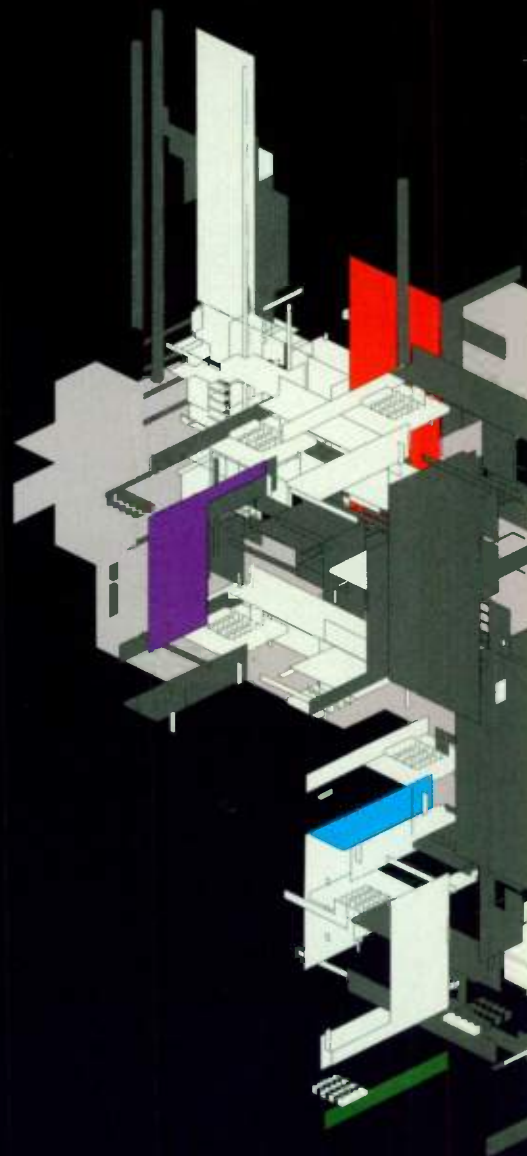

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Make trade fair... and bag a bargain!

Phil Goodall

I love to buy second-hand gear, especially synths. If I'm patient, I can pick up a classic for a fraction of the cost of buying when it was new. Admittedly, it's possible to get caught out and end up with an Emu that makes sounds like, well, an Emu. But most of the time good deals can be found.

A major benefit of buying second-hand is that you can afford to make a mistake! For example, if you've desired a piece of gear, but were not sure you really needed it ("Hmm, I fancy getting a fifth virtual analogue synth, I wonder if it will improve my sound and get me an instant record deal..."), you can buy it and try it, but when you find out that it makes no difference (because, actually, the problem is that your songs are and always have been crap), you can sell it on without much of a loss. You may even make a profit if you buy shrewdly.

The main places to buy your second-hand gear these days are on the Internet. First, there is the wonderful *Sound On Sound* Synth Supermarket, commonly known as the Readers Ads (www.soundonsound.com/readersads). Then, of course, there's eBay. There may be other sites out there, but the ones I've come across are pretty poor compared to these two leviathans.

The irrepressible rise of the web-based auction and classified sites has been a boon for people looking to buy hi-tech gear. I, for instance, have managed to get a Yamaha FS1R and a Kawai K5000R, both for a reasonable price, from the aforementioned Internet rummage sales. In the dark days, before eBay or Readers Ads, I would probably never have found either of those units, let alone bought them!

In general, my experience of the second-hand market has been fairly positive so far. But, let's face it, there's no point writing in to Sounding Off unless you have something to say, so what's my problem with eBay and Readers Ads? Well, put simply, it's you, out there in Readerland, and you know who you are! You're the one who puts buzzwords like 'phatt', 'vintage' and 'analogue' in your advert for a Yamaha DJX! You're the person that puts an asking price on your five-year-old Korg Electribe that is higher than its original retail price; it's not a TB303 yet, you know! You're the eBayer that wants £25 to package a "really nice compressor" — jeepers, what are you packing it with, saffron or something? Another word used in adverts that really ticks me off is 'rare'. For example, "rare 1980s Casio CZ101 for sale, £500". Is this the synth that was so rare that Vince Clarke had 16 of them — one for each MIDI channel?

I suppose it can be quite entertaining, though. I sometimes laugh out loud as I scan the Readers Ads, exclaiming, "I can't believe it! Someone wants three hundred quid for a circuit-bent Speak 'n' Spell." But seriously, some of the rubbish posted on these sites makes me fume. At the end of the day, you look through the Readers Ads to pick up a bargain, not to pay more than you would in a shop.

I still keep seeing TB303s listed for about £600, with TR909s for £800! Are there really people out there who would pay so much for this rickety old gear? OK, about 15 years ago, when alternatives were scarce, I can believe that they may have (sort of) been worth the money to some acid-crazed loon with too much cash. Aficionados of these Roland classics often get really uppity if you claim that modern gear sounds as good. They often weigh in with some silly, generalised defence of their beloved geriatric gear, like "samples just don't capture the constantly changing character of the 909 snare." Or "there's such a groovy instability to the original unit that just can't be created in a software sequencer." Software doesn't usually capture the sound of the scratchy pots, or mains hum from the power supply either!

This brings me on to something else: sometimes us bargain-hunters are our own worst enemies! Words and phrases such as 'classic', 'russian', 'beautiful evolving pads', 'analogue' and 'tight bottom end' do seduce into paying over the odds for gear that is, frankly, not worth it.

So do me a favour: if you are selling gear, be honest and ask a price that is at least a third less than retail. If you are buying, walk away rather than pay over the odds. If we all follow these simple requests, an Aladdin's cave of bargains awaits! SOS

About The Author

Phil Goodall was a member of the Adventure Babies, the last band signed to the legendary Manchester record label, Factory. He has worked with U2 producer Steve Lillywhite and has supported acts including the Pet Shop Boys and Pulp. Phil now has a modest home studio and enters the Eurovision Song Contest for fun (www.myspace.com/swirlerburner).

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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